'Systems can be made as large and as complex as we want'

Audionetworks

The distribution of audio nowadays almost always takes place over a network. In this article an overview of protocols, commonly used terms and the latest developments.

What we dreamed of as sound engineers thirty years ago has now become an ordinary thing: it has become. An almost unlimited number of channels, high quality, flexibility, seamless redundancy, all over a 5 mm thick pipe. Welcome to the world of digital audio networks.

Definition

A network can be defined as a structure of connections in which the logical layer is decoupled from the physical layer. In practice this means that the data connections (in our case audio, clock, control and the like) can be freely programmed within the available bandwidth, independent of the way in which the network is physically structured and wired. We are already used to that with computers. The computer no longer has a direct connection to the printer, but the document to be printed finds its way through the network cables or through the air via wili. If a printer is connected via an old-fashioned serial port, then that physical connection is there and speaking we of a point-to-point connection.

Ethernet history

It was precisely the inefficiencies of point-to-point connections that prompted Richard Metcalfe of the Palo Alto Research Center (PARC) in 1973 to work with David Boggs to lay the foundation for what would later become Ethernet. Ethernet got its first commercial applications in 1980 and was established as a standard in IEEE802 in 1983. 3. This Ethernet has since gone through a huge evolution from 10Mbps to 100Gbps or even higher. The internet, Wi-Fi and fiber optics are all developments in line with the original Ethernet.

Audio over ethernet (2)

When the data rates were finally high enough, a number of

manufacturers came up with the idea of sending digital audio over Ethernet. The first very successful protocol was the CobraNet developed by Peak Audio in 1996: a bi-directional connection of up to 64 channels over every 100 Mbps cable in the network. CobraNet was picked up very widely by industry and used in particular in installation projects. A primary and a secondary connection to the CobraNet device are standard to provide independent network connections for redundant applications.

build works. In 2001 the protocol was taken over by Cirrus Logic and the technology has not been further developed. However, products with CobraNet are still being made and thus ensure compatibility with the large amount of installations that are based on CobraNet worldwide.

Audio over ethernet (2)

Although EtherSound works a little differently technically, this protocol released in 2001 is also an audio network that uses standard Ethernet components. Where with CobraNet the bandwidth used depends on the amount of data traffic over the network, EtherSound is a so-called streaming network. A bidirectional stream is established from 100Mbps that precisely reserves space for 64 channels in each direction. In a later variant of Ethersound (V3 or E1100) it is possible to configure a ring of devices to guarantee redundancy.

Audio over ethernet (3)

In 2006 Audinate introduced the Dante protocol. an Audio over IP network based on gigabit connections with up to 500 channels per network cable in the network. The total maximum number of channels in the network is much larger, only limited by the available bandwidth in the system. Higher sampling rates (up to l 92kHz) and bit rates (up to 32 bits) are also supported. Dante has the option of running systems redundantly through a redundant network. Dante is currently used by 150 manufacturers in all areas of application, including touring, fixed installations and broadcast. At the moment, the acceptance in the market of the Dante protocol is the greatest.

Audio over ethernet (4)

Ravenna, Q-LAN and Live Wire are Audio over IP network protocols that are very similar to Dante and are used by different manufacturers. They are expected to offer some degree of compatibility over the AES 7 6 protocol in the future. This protocol offers the lowest functionality of the affiliated protocols, thus sacrificing functionality in mils for compatibility. For the implementation, the AES Media networking Alliance was founded in 2014 by Bosch, Telios Alliance, QSC, LAWO and Yamaha. Another development is AVB, which is expected to be a layer 2 standard that cannot be used with the other protocols. Both AES6 7 and AVB are not yet ready, implementation will become reality in 2015 or later.

Propriëtair protocols

Some manufacturers have opted for their own, so-called proprietary solution instead of Ethernet. Optocore uses its own protocol to transport up to 500 channels over a fiber optic connection. Within this format there is room for audio, low-res video and also low-bandwidth data connections. With RockNet, Riedel has an up to 160-channel ring system. CATS or fiber optics are used for the connections. Optocore and RockNet both use a ring system for redundancy. A' 10Mbps Ethernet connection can also be made between two machines via Rocknet. Because Optocore and RockNet are not supported by several manufacturers, the choice of components is limited, but as long as the manufacturer has the desired product

'The challenge for the coming decade is to stay on top of the extremely increased possibilities and scale'

offer, that is no problem at all. Both have a wide range of converters, stage boxes and mixer expansion cards available.

Own protocol as system backbone

Stagetec (Nexus) and Yamaha (Rivage/TwinLane) use a proprietary network as a 'system backbone' with a higher bandwidth and audio resolution than current gigabit protocols can provide. In addition to the internal backbone, Dante is particularly supported as an open network protocol.

Point-to-point coonections

Two commonly used point-to-point protocols are MADI (AES10) and SuperMac (AES50). These are not networks - they are solutions that connect two devices together, for example a mixer with a stagebox where the functional connection is determined by the physical cabling. This is fine for simple systems, but larger systems require hardware audio routers and splitters to connect each additional device. A' network is often more efficient, both in terms of costs and flexibilty

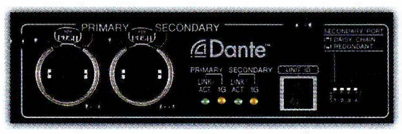
Cobranet

- Ethernet compliant protocol
- Introduced 1996 by Peak Audio (VS)
- Capacity 24bit@48kHz: 64 channels per cable, >100 per netwerk
- Ued by Peavey, QSC, Yamaha and more than 50 others
- www.cobranet.org

EtherSound

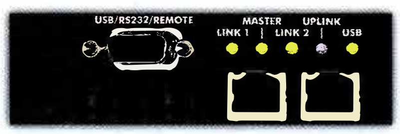
- Ethernet compliant protocol
- Introducion 2001 door Digigram (Frankrijk)
- Capacity 24bit@48kHz: 64 channels per cable / network
- Used by Auvitran, Allen & Heath, Yamaha en more than30 others
- www.ethersound.com

Dante



- Ethernet compliant protocol
- Introduction 2006 by Audinate (Australië)
- Capacity 24bit@48kHz: 500 channels per cable, >1000 ... per netwerk
- Used by Bosch, QSC, Yamaha and more than 150 others
- www.audinate.com

Optocore



- Propriëtair ring-netwerk protocol
- Introduction 1996 door OptoCore (Duitsland)
- Capaciteit 24bit@48kHz: 512 channels per cable/ network
- Used by Optocore only
- www.optocore.com

Latency

A much discussed aspect of digital networks is latency. Latency can be defined as the time it takes for an applied signal to reach the output of the system. So from converter to mixer, or from matrix to amplifier. An entire system will almost always run through the network several times and the final latency of the signal will be a multiple of the single latency. It is therefore important to keep the latency of the network as small as possible. In the past, systems had high latency due to limited network speeds and switching capacity, but today that speed is no longer a limitation. A latency of 250µs is possible without any problems and that is relatively small compared to the latency that AID and DIA converters have (-2ms at 48kHz). An efficient approach here is to convert as little as possible to analog and leave everything in the digital domain where possible.

Clocking

Clocking is inherent to a digital system. The devices that send and receive digital information all need to have an idea of relative time so that the large amounts of data passing through are correctly interpreted. A device in the network determines that time (the wordclock master) and all other devices (the slaves) synchronize to that clock signal. In conventional digital setups this is accomplished by mastering one of the audio devices (or a separate wordclock generator), and synchronizing all slaves using the audio signal itself or separate BNC 'wordclock' cables. In networked systems, the network can take over this function. No additional cabling is then required. With Ethernet mechanism and Precision Time Protocol (IEEEl 5 8 8) the stability of the synchronization is in many cases also higher than can be achieved with external clock distribution.

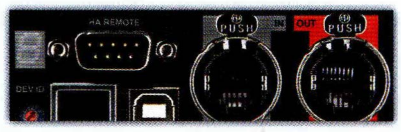
Redundancy

Redundancy means that the system automatically switches over in the event of a failure, for example if an audio connection is lost. The system as a whole will then continue to work. This is not only desirable for larger events or broadcast applications, but also an extremely important function for safety reasons. Proprietary protocols such as Optocore and RockNet have built in a mechanism for this. Both systems can be connected as a ring, so that the system always chooses an alternative route in the event of a connection failure. In the Ethernet compliant protocols it is often possible to make both the network layer and the audio layer redundant. In the first case this takes place within the switches, in the second case via the audio devices themselves.

How do audionetworks sound?

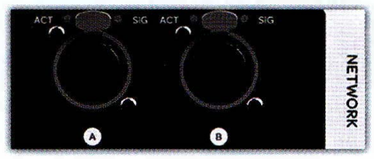
Let's dispel a myth. Behind the RJ45 port of devices in an audio network is a Network Interface Card (NIC) that sends or receives the audio samples as Ethernet packets in the network. The bit rate and sample rate at which this happens are determined by the host components where these NICs are built in, for example A/D converters and D/A- ▸

Rocknet



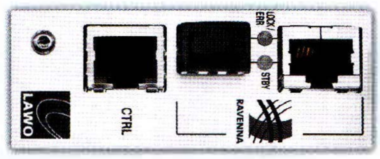
- Propriëtair ring-network protocol
- Introduction 2008 by Media Numerics (Duitsland)
- Capacity 24bit@48kHz: 80 of 160 channels per cable/ network
- Used by Riedel (only
- www.riedel.net

AVB



- Ethernet compliant protocol
- Introductie: 2014 d
- Capacity 24bit@48kHz: 500 channels per cable, >1000 per network
- Used by AVID, Meyer Sound (own implementions), more than 60 members (AVNU alliance)
- www.avnu.org

Ravenna



- Ethernet compliant protocol
- Introduction 2010 byr ALC Networkx (Duitsland)
- Capacity 24bit@48kHz: 500 channels per cable, >1000 per network
- Used by: Genelec, Lawo, Neumann and more than 30 others
- ravenna.alcnetworx.com

AES10 /MADI

- Point-to-point proto-col (no network)
- Introduction 1991 as AES-standaard
- Capaciteit 24bit@48kHz: 64 channels per cablel
- Used by DigiCo, Soundcraft, Stagetec and more than 35 others
- http://en.wikipedia.org/wiki/MADI

AES50, Supermac

- Point-to-point protocol (no netweok)
- Introduction 2006 Sony Oxford (UK) as AES-standaard
- Capaciteit 24bit@48kHz: 48 channels per cable
- Used by Auvitran, Midas/Behringer/Klarkteknik, Lynx
- www.supermac-hypermac.com

converters. So as long as the bit rate and sample rate of the hosts are supported by the network, the associated audio quality is not limited in any way. The only thing that an audio network changes from the audio signal is the extra latency that the network adds to the already existing latency of the converters and DSPs in the system. Conclusion: apart from the latency, a network does not affect the audio signal in any way. What goes in comes out exactly the same. Networks therefore have no 'sound'. The statement 'network X sounds better than network Y' always has to do with the sound quality of the hosts (A/D, D/ A, DSP), not with the network. Keep in mind that hosts sometimes support multiple network protocols, often through plug-in cards.

Praktical aspects

An audio network is made up of hosts, the actual audio components. Each host is connected to the network. This can be done in a daisy-chain or ring topology, without using network equipment. Complex systems were often built as a 'star', with all hosts connected to a central switch with Ethernet cables. Switches are available in all sizes, from 8 to over 100 ports. The connection is usually via copper cable (CATS, CAT6) up to 100 meters in length, for longer distances via optical connections. The most commonly used road proof connectors on the market are Neutrik EtherCon (for CATS/CAT6) and OpticalCon (for optical cables).

Switches are built for IT infrastructure and have many software functions to optimize data traffic. But in between audio data and normal data traffic (downloading, printing, internet) there is an important difference: the audio must arrive at the other end super fast, preferably within 1 millisecond. Fortunately, there are many features in the Ethernet protocol that make this possible, but there are others that hinder it. When choosing a switch, it is important to choose a type that supports all functions that are important in audio, and where inhibiting functions can be turned off. Most audio manufacturers have a list of suitable switches, with corresponding manuals to configure the switches for audio use.

Scalability

The number of 'networked' devices on the market is growing exponentially. The advantage is that more and more audio connections can remain digital, finally getting rid of the quality limitations of multiple A/D and D/ A conversions in the signal path. It also saves on hardware for the conversion from one protocol to another. And the wide range of audio products makes systems scalable. Two smaller systems can be temporarily combined into a larger system without the need for additional hardware. Computers have an Ethernet connection as standard and can therefore also be connected directly to the network, in any amount. More and more audio-software solutions applied - think of plug-ins, multichannel live recording and Digital Audio Workstations.

> **'Networks don't have sound. What goes in, comes exactly the same again**

Different knowledge from audioengineer

To a certain extent, audio networks are easy to manage yourself. It becomes complicated when audio data has to be sent over an existing Ethernet infrastructure. If that is only cabling (CATS patches or fiber optic) then there is no problem. But if it's an IT-run infrastructure, first sit down and agree on how one should be implemented. Managing the IT side of audio networks isn't the most significant shift, though. "The biggest challenge for sound designers and mixing engineers is to manage the enormous increase in functionality. With an analog mixing console, the number of functions is limited by the physical buttons and the available channels on the rear panel. There is no such limitation with digital systems. DSP -Functionality is in principle available without limitation because multiple DSP sources can be combined without any problems. Inputs and outputs can be added to the network in a modular way - as much as you need. Recently we also see the decoupling of user interfaces. There are more and more alternative ways to control DSP algorithms - for example with apps on a smartphone or tablet. Finally, the bandwidth limitations of the previous decade, both in terms of inputs and outputs and the amount of DSP. Thousands of channels and many gigaflops of DSP are available at a low cost. So systems are so big and so complex if we want to - and that usually means it will. The major challenge for the operators of the system is to manage it, to maintain control and overview of the routing, DSP functionality and user interfaces.

Conclusion

Ten years ago, we closed the decade of digitization in the audio industry. Now we are in the process of completing the introduction of network technology - few products have been released without an Ethernet connector. Managing the IT side of the story is new and complex, but doable. The challenge we will face in the coming decade is to stay on top of the extremely increased possibilities and scale due to network technology. Ultimately, any audio system, networked or not, is about the audience's listening experience. The challenge of audio networking is not so much the IT technology itself, but learning to use the abundance of functionality to create even more fantastic audio productions.. ◂

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Onlangs heeft Yamaha zijn white paper over audionetwerken geupdate, zie onder Training & support op www.yamahaproaudio.com/ global.