YAMAHA System Solutions *white paper*

An introduction to networked audio

This white paper's subject is 'Networked Audio'.

In the past decade, audio networking has changed the way audio systems are designed, built and used in the professional audio industry. Compared to the previous generation of point-to-point distributed systems, new powerful networking technologies have become market standard, and with them, new practical and strategic issues have become important to consider when investing in a networked audio system.

In this white paper the basics of audio networking will be covered in a straight forward comprehensive format. We assume the reader has an advanced knowledge of analogue audio systems, a basic knowledge of digital audio systems and no knowledge of computer networking. This white paper is only a basic introduction to the subject; for detailed information we refer to the many documents on the internet made available by the IT equipment manufacturers around the world.

The Yamaha Commercial Audio team.

An introduction to networked audio

1. What is networked audio?

With the introduction of digital technologies the amount of information a single cable can carry has increased from a few thousand bits per second in the sixties to a few billion bits per second in 2014. Regular affordable connections in every day information systems now carry one or more gigabits of information in a single fiber cable over distances spanning many kilometers. This bandwidth is enough to transport hundreds of high quality audio channels, replacing hundreds of kilograms of cabling in conventional analog systems. More importantly, the functional connections in a networked audio system can be designed separately from the physical connections in the network. This functionality opens up a wide array of exciting possibilities for the audio industry: any number of i/o locations can connect to the network anywhere in the system without the limitations of bulky cables, leaving the actual connections to be managed with easy to use software. A networked audio system is digital so audio connections are kept in the digital domain, far away from electromagnetic interferences and cable capacitances that degrade analog audio quality. Control signals can be included in the network without additional cabling. Computers can use the network to control and monitor audio devices such as digital mixers and DSP engines. Video connections can be included using affordable IP cameras; and so forth.

digital audio distribution

Many systems on the market distribute audio between a stage box and mixing console or DSP engine over a single cable using copper or fiber cabling supporting ‘P2P’ (Point To Point) connections such as AES10 (MADI, 64 channels) and AES50 (SuperMac, 48 channels). However, nowadays most systems require more than two locations to be connected, requiring multiple cables if they are built with P2P connections. With the introduction of audio networks a multitude of connections to any number of locations can be supported with more cost effective and easy to implement cabling, including redundancy and the ability to support non-audio connections such as control data and video.

Dante™

Dante™ is an audio network protocol developed by Audinate® that uses a gigabit Ethernet network, providing several hundred audio connections through each cable in a network. Standard Ethernet services such as QoS (Quality of Service) and PTP (Precision Time Protocol) are utilized to achieve a very low latency with highly accurate synchronization. Dante™ uses a star topology, with many products also supporting a daisy chain topology.

EtherSound™and CobraNet™

The legacy EtherSound™ and CobraNet™ audio network protocols developed by Digigram and Peak Audio have the ability to route 64 audio channels in bi-directional mode through an Ethernet cable with very low latency. EtherSound™ systems can be designed using a daisy chain or ring topology, offering buss style routing of audio channels both downstream and upstream. CobraNet™ systems use a star topology with free addressing of bundles of audio channels from any location to any destination.

Open and Closed systems

Dante™, EtherSound™ and CobraNet™ are open systems using standard Ethernet network architecture. This means suitably chosen off the shelf IT equipment can be used to build a network, taking full advantage of IT industry developments in functionality, reliability, availability and of course cost level. All three protocols are licensed to many of the worlds leading professional audio manufacturers, so products from different manufacturers using the same protocol can be combined in a system without problems. Several closed networked audio systems exist on the market, supported only by products of the network manufacturer. Examples are Nexus, Rocknet and Optocore.

Yamaha?

Yamaha adopts an open and inclusive approach, advocating the choice of a network platform appropriate to the system’s requirements. The Yamaha product portfolio includes Dante™ products, and also legacy EtherSound™ and CobraNet™ products. In addition, also closed network protocols and point to point connectivity are supported through interface cards.

example live networked audio system

16 input 8 output stagerack | CAT5E cable | networked mixing console | CAT5E cable | 32 input 24 output stagerack | CAT5E cable | networked mixing console

2. Three good things to know about networked audio

One: cable weight and flexibility

In conventional analog audio systems every single connection uses a copper cable. With high channel counts and cable lengths, cable weight can easily exceed 100 kilograms. With the increasing popularity of digital mixers in the pro audio industry, digital cabling such as AES/EBU has been used often to replace analog cables, reducing cable weight and increasing audio quality as electromagnetic interference and cable capacitance problems are much less of an issue in (properly designed) digital cabling. Point to point audio formats such as AES10 (MADI) and AES50 (SuperMac), and network protocols such as Dante™, CobraNet™, EtherSound™, Rocknet™ and OPTOCORE® have become popular for studio and live applications, replacing individual copper cabling with lightweight STP (Shielded Twisted Pair) or fiber cabling. The weight of STP or fiber cabling is much lower compared to individual analog and digital copper cabling. Additionally fiber cabling gets rid of grounding problems.
An analog multicore cable - or a bundle of individual cables - is bulky and not very flexible. For Touring application this means roll-out of cables requires heavy equipment, dedicated staff and limited layout possibilities.
For installations, bulky cabling requires large conduits to be installed throughout the building which is a problem especially in historic venues. In comparison, STP and fiber cables are thin and flexible, a drum of 150 meter fiber cable weighs just a few kilograms and can be rolled up to the restaurant '58 tour Eiffel' on the Eiffel Tower by just one person. Installation is easy, network cables in an audio system need very little space and can be placed in an existing cable conduit.

Two: physical and functional separation

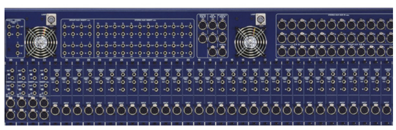
For audio networking protocols such as Dante™ the functional connections are separated from the physical cabling. This means that once network cabling with sufficient bandwidth has been laid out, any connection can be made without having to change the cabling. For touring this allows 'no brainer' connection schemes to be used: just connect i/o equipment to anywhere in the system and press the power button. For installations the inevitable system changes after a project's opening ceremony only require a little programming time to change the network settings, with huge savings on cabling work as a result. Independent from STP and fiber cabling design, signals can reach even the most remote locations in a network. It no longer matters where inputs and outputs are connected to the audio system, any STP or fiber socket will do. In a live touring situation this allows small groups of inputs and outputs to be distributed all over the stage instead of using bulky centralised connection boxes. For installations this means more freedom of choice to use multiple i/o locations in a venue, not limited by physical cabling constraints.

Three: control!

Using network information technology to distribute audio has the advantage of including.... information technology. Control signals can be included on the same STP or fiber cabling, so there's no longer a need to lay out additional GPI, RS232, RS422 or RS485 cables. Examples are IP video connections, software control over Ethernet, machine control using RS422 serial converters, even internet access. Wireless access points can be used to control system components with tablets.

analogue live distribution system

analogue snakes



analogue mixing console rear panel
(PM5000)

networked live distribution system

networked i/o racks

CAT5E cable

networked mixing console rear panel
(CL1)

3. Three things to take into consideration

One: latency

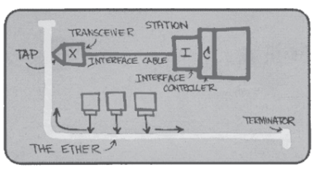
The building blocks of Ethernet networks are cables and switches. To be able to route information over a network, a switch has to receive information, study the addressing bits and then send the information to the most appropriate cable in order to reach the destination. This process takes a little time of several microseconds. As networks grow larger so does the number of switches a signal has to travel through, increasing the delay with every switch. In medium sized live audio systems the network, AD/DA conversion and DSP each cause roughly 1/3rd of the total system's latency. The total system latency must be considered and managed carefully to ensure the best sound. 'In-ear monitor' applications are the most demanding and least tolerant of latency of any kind; a latency between about 5 and 10 milliseconds becomes noticable, above 10 milliseconds the delay becomes too obvious. For PA FOH and monitor speaker systems the problem is relatively small, a one millisecond increase in latency corresponds with placing a speaker just 30 centimeters further away. The latency performance of audio network protocols running on gigabit networks, such as Dante™, can be well below one millisecond, posing no problem even for in-ear monitoring systems.

Two: redundancy

In an analog system the audio signals run through individual cables, so if a cable breaks down typically only one connection is affected. In many cases some spare connections are planned in multicore cables so system functionality is not seriously affected if something happens and a solution is easy to accomplish. In a network however, the failure of a single long distance cable can potentially disable the complete system, giving the engineer a hard job restoring it. This is why networked systems have to be designed with redundancy mechanisms: the system should include redundant connections that take over system functionality automatically if something goes wrong. Some excellent redundancy features have been developed by the IT industry in the past years as banks, nuclear power plants and space agencies also need redundancy in their networked systems just as we do. Cables can be laid out double for all crucial long distance connections; if one cable fails the other takes over. Especially in touring applications it is advisable to use redundant hardware as well, as some IT equipment is primarily designed to be used in air-conditioned computer rooms, and may be more vulnerable when used in harsh on-the-road conditions. For sensitives applications, touring grade switches are available for harsh environments.

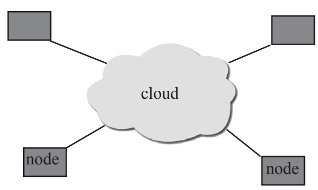
Three: complexity

For every functional connection in an analog system the physical form of the connection is visible, normally as an XLR cable. Anyone looking at the system, or making his way through the spaghetti-style wiring hanging out of the back of a mixing console, can work out what is connected to what. In a network it's quite different as the functional connections are separate from the physical connections. Looking at a networked system a troubleshooter only sees devices connected to other devices with a few STP or fiber cables. One cable can carry maybe two audio signals, or three hundred and sixty eight - there's no way to tell. Where analog systems allow DIY - Do It Yourself - design and assembly by inexperienced users - Networked audio system design requires experienced system engineers who are up to date with networking technology. This drastically changes the role system integrators, system owners and system users play in the process of purchasing, designing, building, maintaining and using audio systems, a new role everybody in the process has to get used to.



Robert Metcalfe's first Ethernet drawing


switch



network architecture

4. What is an Ethernet network ?

Ethernet

Back in the seventies the Palo Alto Research Center in California, USA (www.parc.com) developed some nifty computer technology such as the mouse, the laser printer and computer networks. From the first versions of networks such as Aloha-Net and ARPA-Net the Internet has evolved. Robert Metcalfe, first working at PARC and later founding his own company 3COM, developed a practical networking standard for use in offices called Ethernet. More than 30 years later the whole world is using this standard to build information systems, and all personal computers, smart phones and tablets sold today have some form of Ethernet port built in. The Ethernet protocol is standardised as 802.3 by the IEEE standards organization.

Building blocks

The basic building blocks of Ethernet networks are network interface cards (NIC, built into devices such as computers, digital mixers), cables to connect them to the network, and switches; devices that tie all cables in a network together and take care of the correct routing of all information through the network. The operating speed of these building blocks, determining how much information a network can carry, has evolved from 10 Megabits per second in 1972 to one Gigabit per second and higher in 2014.

Addressing

Ethernet works by dividing information streams into small packets and then sending them over the network to a certain receiver address specified by the sender. Every Network Interface Card (NIC) has an address, and switches keep lists of addresses connected to the network in their memory so they know where to send packets. Every NIC in the world has a unique Media Access Control (MAC) address programmed by the manufacturer. There are 280 trillion different MAC addresses, and there is only one company in the world, the IEEE standards organization, that allocates these addresses to manufacturers. This way all MAC addresses of all NICs in the world are unique: there are no doubles, the system always works. In addition to MAC addresses, a 'user definable' addressing layer is used to make network management easier for local networks. This additional user address is called the Internet Protocol address, shortnamed 'IP' address. The IP address is normally 4 bytes long ('IPv4'), divided in a network number and a host address. This division is determined by a key that is also 4 bytes long called the 'subnet mask'; every bit of the IP address that has a 1 in the subnet mask belongs to the network number, all bits with a corresponding zero belong to the host address. The trick is that only NICs with the same network number can exchange information with each other. In most cases the network number of small office networks is 3 bytes long and the host address is one byte. One byte (8 bits) can have a value between 0 and 255. In network setting displays on personal computers the software fills in the IP and subnet values as four decimal numbers (0-255) corresponding to the four bytes in the address and subnet mask. In small office networks the subnet mask often has the default value of 255.255.255.0 - giving the network administrator 255 host addresses to use as only the last byte can be changed and assigned to devices on the network. The first three bytes do not change and are the network number. For larger networks the subnet mask can be changed to make room for more host addresses. Normally users have to program the NIC's IP address manually to make the network work, but in many cases a centrally located device (switch, router or computer) can be programmed to do this automatically whenever a NIC is connected using the Dynamic Host Configuration Protocol (DHCP). in 2008 a 16 byte IP address ('IPv6') has been implemented because the amount of devices active on the internet had outgrown the 4 byte address range. For industrial networks however, including audio networks, the 4 byte version is still used.

VLAN

The Ethernet 802.1q standard allows for Virtual Local Area Networks (VLANs) to be created within one high speed network. This way multiple logical networks can co-exist using the same hardware to support a system's workflow, for example to create separate logical networks for audio, video and control data. Most managed switches support the VLAN standard.

Networked audio

Every Ethernet compatible networked audio device, such as Dante™, CobraNet™ and EtherSound™ devices, has an NIC built in so it can send and receive information on an Ethernet network. The audio protocols use the MAC addressing layer to send and receive data. As MAC addresses are unique the devices will work with any Ethernet network worldwide.

5. Network topologies

P2P

Strictly speaking a Point to Point (P2P) topology is not a network, although a network can be used to create such a system. A P2P system includes only two locations with a fixed multichannel connection. Examples of digital audio formats for P2P systems are AES3 (AES/EBU, 2 channels), AES10 (MADI, 64 channels) and AES50 (SuperMac, 48 channels). A distribution device such as a splitter or a matrix router can be used to include more locations in the system.

Daisy chain

Daisy chain is a simple topology that connects devices serially. The EtherSound™ protocol allows connections to be made using a daisy chain topology, with devices that read and write audio channels in a bi-directional data-stream at a fixed bandwidth of 64 channels in both directions. An advantage of this topology is that the routing of network information is relatively simple and therefore fast; a daisy chained EtherSound™ device adds only 1.4 microseconds latency to the network. A disadvantage of daisy chain topology is the system behavior in case of a failure of a device in the chain: if one device fails the system is cut in two parts, without any connection between the two. EtherSound™ daisy chains can be split using switches in a star topology, but in that case the audio data can flow through the system's switches in one direction only. Some Dante™ devices have a small switch built in, enabling them to also support a daisy chain topology.

Ring

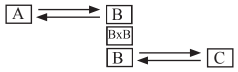
A ring topology is a daisy chain where the last device is connected to the first forming a ring. As all devices connected to the ring can reach other devices in two directions, redundancy is built in: if a device fails only that device is disabled. For additional redundancy a double ring can be used. OPTOCORE® offers a proprietary system using a redundant ring topology with a high bandwidth of up to 500 audio channels, video and serial connections. Rocknet offers a proprietary redundant ring topology with 80 or 160 channels capacity. The EtherSound™ ES-100 standard supports a redundant ring topology offering 64 audio channels.

Star

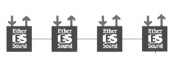
As a star topology makes the most efficient use of a network's bandwidth, most information networks are designed as a star. The center of a star carrying the highest network information traffic can be designed with extra processing power and redundancy, while the far ends of a star network can do with much lower processing power. Variations of a star topology are 'tree' and 'star of stars'. A star topology also offers easy expansion, new locations can be connected anywhere in the network. A downside is the important role of the center star location as all network information to and from connected devices runs through it; if it fails a big portion of the network is affected. A network using a star topology can be made redundant using the Ethernet Spanning Tree Protocol. Dante™ and CobraNet™ use a star topology, supporting full redundancy by offering double links to the network.

Selecting a topology

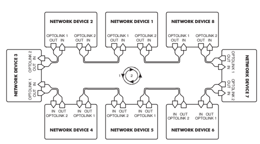
For every individual application one or a combination of these four topologies is most appropriate. Decision parameters include the number of locations, channel count, latency, desirable system costs, reliability, expandability, open or closed, standard Ethernet technology or proprietary systems etc. To make a decision on choosing the topology, a certain degree of expertise on networking technology is required, often found in an external consultant or a qualified system integrator with a track record in designing networked audio systems.



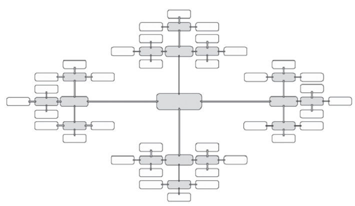
P2P topology (MADI)



Daisy-chain topology (EtherSound™)



Ring topology (OPTOCORE®)



star topology (CobraNet™)

6. Redundancy concepts

Trunking / Link Aggregation

The Ethernet IEEE 802.1.ad link aggregation / trunking standard allows managed switches to be connected with 2 or more cables, distributing information traffic over the cables. If one cable fails, the other cables take over the lost connection automatically. The aggregated link will switch to a lower speed as it misses one cable, so aggregated links should be designed with ample headroom. Trunking only makes the connection redundant, if one of the switches fails then the devices attached to that switch will be disconnected.

Ring

A ring is basically a daisy-chained collection of devices with the last and the first device also connected. Every device is then connected to the network with two cables, so if one cable in the system fails the connection is still intact. A second failure will cut the network in two. A redundant ring topology with packet streaming network protocols such as Ethersound, Optocore and Rocknet offers excellent redundancy requiring less cables compared to star topologies. Although a ring topology can be designed to work for packet switching network protocols such as Dante™ and CobraNet™ as well, it is not recommended because additional switches have to be used.

Spanning tree

In star networks packets of information are sent through the network based on the IP and MAC addresses. It is vital that the network has a logical architecture: for every source-destination combination there can be only one path through switches and cables. If there are more paths loops can occur, with the danger that information packets can flow through the loop forever, congesting the network. So loops are normally not allowed in star networks, unless managed switches are used that support the IEEE 802.1w Spanning Tree Protocol, shortnamed STP. Switches supporting STP can block ports that cause a loop, but unblock it when the active port in the loop fails. Several loops can be created in a network to protect network areas. For full redundancy a network can simply be built double, with double switches at all locations connected to each other. The advantage is that the system can recover from any failure, the downside is that this takes a while: a few seconds for large networks. Most managed switches support some form of STP - possibly as RSTP (Rapid STP) or MSTP (Multiple STP).

Dante™ and CobraNet™ Dual Link

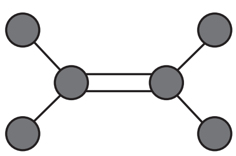
Each Dante™ and CobraNet™ device has two Ethernet ports built in, labelled 'primary' and 'secondary', that work in redundant mode. Normally the primary port does the job, but if that connection fails the secondary port takes over automatically. With Dante™ simply a double network can be used to have full redundancy, with Cobranet™ such a network needs to additionally use the STP protocol.

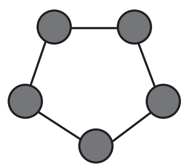
EtherSound™ ES-100 PPM

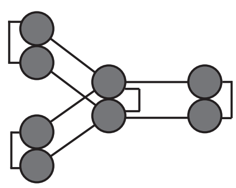
The EtherSound™ ES-100 standard allows devices to be connected using a ring topology, appointing one device as 'Preferred Primary Master'. This PPM device blocks the ring in normal operation, and unblocks it when the ring is broken somewhere; a function similar to Spanning Tree.

Selecting a redundancy concept

For every individual application one or a combination of these redundancy concepts can be selected. One decision parameter is the required redundancy level; in touring applications it would be sensible to use redundant switches, in installed systems single switches might be enough. Normally the minimum is to have the long distance cabling redundant, with the cables separated physically as much as possible. Another decision parameter is the recovery timing - the time the system needs to recover from a cable break or switch failure.
If a closed system such as OPTOCORE®, Rocknet™ is used, the redundancy concept is selected by the manufacturer. If standard Ethernet equipment is used, some advanced knowledge is required to select the redundancy concept and program all switches in a networked audio system.


trunking


ring


spanning tree with double switches

7. Cabling

UTP and STP cables

Most Ethernet networks are built using cables containing eight copper wires twisted by pair. The shielded variety is shortnamed STP for Shielded Twisted Pair, offering protection from electromagnetic interference. The more commonly used unshielded variety is shortnamed UTP for Unshielded Twisted Pair. These cables and their connectors come in different qualities for different applications, standardised by the Telecommunications Industry Association (www.tiaonline.org) in categories 1 to 7. The categories differ by the materials used and the twisting of the wire pairs per meter. CAT3 is a low quality cable used for low speed 10Mb Ethernet networks. For 100Mb Ethernet based networks CAT5 or higher must be used. Caution: CAT3 cable looks the same as CAT5, so always study the category indication on the cable’s sleeve. For use with Gigabit systems an improved version of CAT5 is available: CAT5E. The recently introduced CAT6 and CAT7 have even better performance characteristics. The TIA categories are backwards compatible. Within cable categories different qualities are available: solid core for installation, flexible core for patching, protection jackets and Shielded Foiled (S/FTP) for road proof touring.

UTP and STP Connectors

Copper Ethernet network cables use RJ45 form factor connectors. The industry mostly sells cables and connectors separately, system integrators and installers can assemble cables using simple cable tools. Installation cables (solid core) and flexible cables (stranded core) in their UTP and STP form require different versions of RJ45 connectors. Switch manufacturers mostly refer to a CAT5 copper network connector as ‘TX’, i.e. ‘1000BASE TX’. In the audio industry Neutrik’s EtherCon® is often used for RJ45 road proof connector systems.

Fiber cables

Optic fiber cables can handle much higher frequencies compared to STP cabling while cable runs of over 10 kilometer can be used. The industry offers two kinds of fiber system: multimode and singlemode. Multimode fibers can handle gigabit connections of up to 2 kilometer. Single mode fibers require a more expensive laser diode, but can handle connections of up to 80 kilometers. Both varieties are commonly available in IT shops as installation fibers; some companies such as Fiberfox® offer military spec fiber cables for road proof touring applications.

Fiber connectors

Fiber cable connectors come in many varieties named SC, ST, LC etc. As it’s very difficult to assemble fiber connectors, cables are mostly sold including connectors. Switches often use modular systems to offer fiber connectivity; industry standards for these modules are GigaBit Interface Converter (GBIC) and its mini version called Small Formfactor Pluggable (SFP). Switch manufacturers mostly refer to fiber network connections as ‘FX’, ‘LX’ or ‘SX’, e.g. ‘1000BASE FX’. For road proof connectivity Neutrik developed the OpticalCon® connection system offering extra protection of the vulnerable fiber connectors. Connex offers Fiberfox®, a connection system that uses lenses to disperse the fiber signal to make it less sensitive to scratches and dirt.

Media converters

A switch without a fiber module can be used to work with a fiber connection using a media converter and vice versa. Media converters are widely available for high bandwidth connections. However, it is recommended to use switches with internal fiber modules to keep network latency to a minimum.


RJ45 connector


Neutrik EtherCon®


SC fibre connector


Fiberfox® EBC52


Neutrik OpticalCon


GBIC

8. More about Dante™

The Australian company Audinate® invented Dante™ in 2006, using gigabit ethernet networks as a more powerful alternative to the 100Mb based CobraNet™ and Ethersound. Dante™ is a licensed protocol, and has been implemented by more than 100 manufacturers in 2014. Dante™ works on networks using suitable ‘off the shelf’ switches that support the required QoS and PTP protocols, and allow for EEE (Energy Efficient Ethernet) mode to be disabled. For larger networks, switches also should support the IGMP Snooping protocol.

Concept

Dante™ manages audio data by grouping channels that travel from the same transmitter to the same receiver into “flows”. Each flow will consist of up to 8 channels, and will normally be created without intervention from the user. Dante™ devices also have an automatic discovery mechanism that allows audio routing to be based simply on device names and channel names. This uses Ethernet OSI layer 3 (IP addressing). Most Dante™ equipment require the use of gigabit Ethernet switches, meaning the store-forward and queue delays are much lower than for 100Mb networks. In fact, all Dante™ slave devices are regularly communicating with the Dante™ master device to determine the delay timing. And they will adjust their own timing for the audio accordingly: they use the standard Ethernet Precision Time Protocol (PTP) for this function, ensuring a synchronization accuracy within one microsecond. Furthermore, Dante™ uses a standard Quality of Service (QoS) Ethernet function to ensure that Dante™ sync and audio data are processed more quickly than any other data in the switches. This allows Dante™ to share networks with regular office and other IT equipment.

Routing

Dante™ provides ‘Dante™ Controller’ software, offering a visual matrix type user interface for channel routing, supporting all Dante™ devices available on the market. Some manufacturers provide alternative means of Dante™ routing control, an example is the Dante™ routing user interface implemented in Yamaha CL and QL mixing consoles. The software also controls latency and synchronisation settings. Audinate® also markets a Dante™ Virtual Soundcard (DVS), capable of sending and receiving up to 64 channels from and to the Dante™ network using a personal computer’s ethernet port. This functionality finally includes personal computers as i/o device in networks at no extra hardware cost.

Redundancy

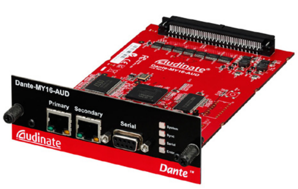
Similar to CobraNet™, Dante™ devices offer primary and secondary ports to connect to the network. Both ports can be connected to a star topology gigabit network to provide seamless redundancy. If required, for example to also make the control and video parts of the network redundant, additional redundancy functionality can be used, such as trunking and spanning tree.

Daisy chain

Some Dante™ products have a small managed switch built in to connect the primary and secondary ports to a network. This switch can be programmed to replace the secondary port for a second primary port, allowing simple daisy chains to be used for easy set up of live systems. This functionality is built into the Yamaha QL, CL, Ri/o, MTX5D and XMV-D products. Note that redundant ring topology is not supported with this method. Trunking using additional switches can be applied to create cable redundancy.

Dante™ Controller

DVS Dante™ Virtual Soundcard



MY16-AUD Dante™ interface

9. More about Cobranet™

CobraNet™ was invented by the US based company Peak Audio in the year 1996 using the 100Mb ethernet standard common at that time. The protocol has evolved since then into a well known and very reliable world standard, still used in many installation projects even in 2014.

Concept

Audio data is chopped up into ethernet packets, with chunks of audio signals combined in a ‘bundle’ containing a string of audio samples of one, two, four or eight channels. The bundles are addressable using ethernet layer 2 (MAC addressing), making CobraNet™ a true network: within the available bandwidth the routing is completely independent of the physical cabling. The big challenge in ethernet audio transfer is to achieve a synchronous timing - audio must arrive at all locations in the network at the same time. To overcome the delays caused by store and forward latency in the network switches, Peak Audio came up with a brilliant clocking scheme. One CobraNet™ device in the network is appointed - automatically or by the user - as a master clock and named the ‘conductor’, sending out a few hundred very small ‘beat packets’ per second. At the time a beat packet is sent, the network is idle, but right after all other devices received the beat packet, they all start transmitting all audio packets (bundles), congesting the network. This takes some time to resolve, depending on the amount of packets and the queues that form at the output ports of the network switches. The trick is that all CobraNet™ devices receiving audio packets wait for a certain amount of time before outputting the audio data to their outputs, with a choice of 5.3, 2.6, and 1.3 milliseconds - depending on the network size. This time delay is enough for the network switches to sort out all the congestion and deliver all audio packets to the receiving devices. Because the bundles contain enough samples to exactly fill the waiting period, the output devices never run out of data so they can can output continuous audio signals. Because the beat packet is sent when the network is idle (no audio data going on), it takes a very short time for the beat packet to travel to the other devices, causing all outputs to be synchronized with an accuracy of only a few microseconds. After the waiting period, the network is idle again and the conductor sends a new beat packet, starting the process from the beginning. CobraNet™ allows for other ethernet functionality to be implemented on the network, such as video and device control, as long as the combined bandwidth doesn’t exceed 100Mb.

Routing

Routing is done by assigning numbers to bundles. A bundle can be multicast - transmitted to all CobraNet™ devices in the network, or unicast, transmitted to a single device. As a multicast bundle is delivered to all other devices, it takes up bandwidth on all cables. With a maximum number of 64 channels per 100Mb connection, this is the limitation of a CobraNet™ multicast network. When an audio signal only needs to be sent to one single destination, unicast can be used, using up only bandwidth on the ports and cables on the route from transmitter to receiver. This way more channels can be used in the network. The setting of bundle numbers, size, latency mode, conductor priority etc. can be done by software. Cirrus Logic, the company that currently holds the CobraNet™ licences, provides the ‘CobraNet Discovery’ software package, while individual CobraNet™ device manufacturers also provide their own software - eg. Yamaha ‘CobraNet Manager lite’. Although CobraNet™ is fully ethernet compliant, Cirrus Logic never released a computer driver to allow direct transmission and reception by personal computers.

Redundancy

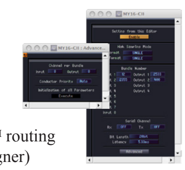
To provide redundancy, CobraNet™ devices have two ports, primary and secondary, allowing the secondary port to be activated when the primary port connection is lost. This ‘dual link’ method can be used together with spanning tree and trunking to make redundant systems.

Phase out

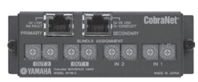
At the start of CobraNet™, switches had a long store/forward delay, requiring the 5.3 milliseconds latency mode to be selected even for small networks. Because this latency is problematic for live systems, CobraNet™ is mainly applied in fixed installations. When fast and powerful gigabit switches became available around the year 2000, 1.3 millisecond mode could be used for medium sized live systems. Since CobraNet™ is no longer maintained by its current owners Cirrus Logic, and Dante™ has become available as a more powerful alternative, CobraNet™ is slowly phasing out.

CobraNet®

CobraNet™ Integrated Circuit



CobraNet™ routing
(DME designer)



CobraNet™ interface card

10. More about Ethersound™

The French company Digigram invented EtherSound™in the year 2001 as a more simple and faster alternative to CobraNet™for live systems, arguing that a simple daisy chain is enough to achieve live sound functionality. By using a daisy chain topology, addressing is no longer required because every device has only one other device to send to, and one device to receive from. This means that switches are not required, so there are no store/forward delays in the network. Also, because there is always only a single stream of audio packets through the cables, waiting queues can never occur. This simplified use of ethernet allows an EtherSound™device to have a latency of only 1.4 microseconds. Because of this low latency, the system uses the audio packets to synchronize, there is no need for a clever clocking method. Although EtherSound™doesn't rely on ethernet packet addressing, it is still a true network: within the available bandwidth of 64 channels, audio connections can be made independent from the physical cabling.

Concept

An EtherSound™device doesn't have primary and secondary ports like CobraNet™and Dante, instead it has IN and OUT ports. The ports support 100Mb ethernet, fully occupied with a stream of 48,000 packets per second, each containing 64 channels of single audio samples at 48kHz. There is no room left for anything else, so although EtherSound™is fully ethernet compliant, an EtherSound™network can not be used for other ethernet functions such as video and control - unless it is routed in isolation through a VLAN using a gigabit network.

Routing

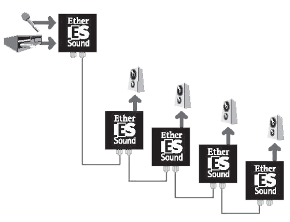
The data stream coming from the device connected to the IN port and sent to the device connected to the OUT port is called 'downstream', allowing channels to be sent from the first device to the last device - and all devices in between. The data stream received from the device connected to the OUT port and sent back to the device connected to the IN port is called upstream, allowing channels to be sent back from the last device to the first device, and all devices in between. Routing of channels is done using AVS-Monitor software from Auvitran, selecting channels to be taken from the up/downstreams and output to the device outputs, and channels to be taken from the device inputs to be inserted into the up/downsteams. An ASIO driver is also available from Auvitran, allowing up to 64 channels to be sent and received by a personal computer through the standard ethernet port.

Redundancy

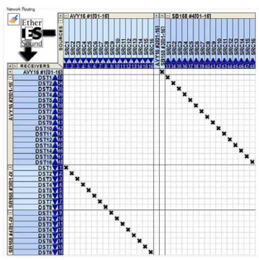
The latest version of EtherSound™is called ES100, allowing the daisy chain to be closed with all devices performing a redundancy protocol similar to spanning tree, forming a redundant ring. The take-over time is very fast - just a few samples. In addition, trunking and spanning tree can be used for larger systems using gigabit switches.

Phase out

Although EtherSound™has a limited bandwidth, it is very simple to set up, and it can be used without any switches with a latency that is low enough for use in live systems. like CobraNet™, EtherSound™is also slowly phasing out, being replaced by Dante™ as a more powerful alternative.



Daisy chain topology



Auvitran AVS-Monitor

MY16-ES64 interface card

11. Other audio network protocols

Other Ethernet compliant audio network protocols or standards are being introduced to the professional audio market at this time. These, like Dante™ are based on OSI layer 3, but have not gained such wide acceptance. They all have a slightly different set of benefits.

AVB

Audio Video Bridging (AVB) is the name given to a network that implements a set of protocols outlined by the IEEE 802.1 standards committee. These include features to guarantee precise synchronization between all devices in the network, to guarantee the available network bandwidth required by the audio and video data, and to guarantee a steady stream of data rather than sudden bursts and pauses within the network. A group of AV and IT equipment manufacturers formed the AVnu Alliance to draw up guidelines to ensure compatibility between all the AVB enabled devices they produce. In 2014 the first AVnu certified AVB equipment has become available. However, AVB networks require the use of special switch hardware which is currently rather costly and not easily available. Also there is no agreed mechanism for redundancy at the time of writing. Both Yamaha and Audinate®have been participating in the AVnu Alliance.

AES67

In September 2013 the AES67 standard was announced for high-performance audio in IP networks. Operating at network layer 3, it provides recommendations in several areas including synchronization, media clock identification and network transport. Audinate® stated in February 2014 that Dante™ will incorporate the AES67 standard, allowing Dante™ equipment to share audio with other AES67 equipment.

Ravenna

Ravenna is the name of a town in Italy where the “Divine Comedy” author Dante™ Alighieri is buried. Thus the Ravenna network was cheekily introduced in the year 2010 by a group of pro audio manufacturers with a focus on broadcasting. Like Dante, Ravenna is an IP based layer 3 solution, and uses IEEE standards such as PTP for clock sync and QoS for data traffic management. Ravenna is already compatible with the AES67 standard, and has been adopted by pro audio manufacturers such as Lawo, Genelec and Merging Technologies.

OCA

Open Control Architecture (OCA) is not an audio network protocol, but a set of specifications agreed by the OCA Alliance (several professional audio equipment manufacturers including Yamaha). The aim is to create and release a free-to-use network control and monitoring communication standard that future audio network devices will adhere to. This promises to enhance compatibility between software and devices of different manufacturers.

AVB logo

RAVENNA logo

12. System Engineering

System users

From the user's point of view a properly designed networked audio system is hassle-free; offering easy connectivity and flexible logistics, supporting the most complex and demanding applications such as installed systems in theaters, concert halls, leisure centers, community centers, schools, etc. Also live touring applications such as touring theater productions, pop concerts, musicals, operas etc. using their own systems or hiring systems from a rental company can benefit from networked audio systems.

System engineering

In the engineering stage normally a part of the system engineering process is handled by the system owner's technical staff, and the other part is taken care of by a consultant or a system integrator. As network engineering requires in-depth expertise on networking technology normally not found amongst audio engineers the role of qualified consultants and system integrators will increase to cover the network specification, design and programming of networked audio systems, as well as the design of easy operating and set-up procedures for the system's users.

System specification

First of all an audio system's specification must be set up. Audio networking opens up a wide array of new possibilities, but the sky is still not the limit; without a deep understanding of networking technology it's very difficult to assess if specifications are feasible or not. A system specification includes the number of audio channels, the number of locations, the distances between the locations, the required audio quality settings, redundancy level, control services etc. If an installed system uses an existing IT infrastructure, the IT system administrator should be included in the specification process as well. For touring applications special handling specifications should be included such as cable and connector quality and connectivity standardization. Based on the system specifications a network format, networked audio format, network topology, redundancy and connectivity can be selected that suits the specifications as much as possible.

Audio components

Closed systems offer a choice of audio components set by the manufacturer. For open systems any brand of audio component that is compatible with the audio network standard can be included. Current examples of open networked audio systems are Dante™, CobraNet™ and EtherSound™.

Network components

For closed systems the manufacturer supplies the network hardware. For open systems the choice of network components is overwhelming; the mature IT market offers many brands of different quality and functional levels of network equipment. For Dante™ systems, switches need to fulfill a set of basic functionality requirements; manufacturers often recommend switches that have proven to work.

Future expansion

Closed systems offer expansion using a limited choice of the manufacturer's hardware expansion options. Open systems using standard networking technology offer user definable scalability: after the purchase of a system both
network and audio components can be added, not restricted to the component brands used in the original system.

Qualified system integrator

All networked audio system designs require an amount of system engineering to be taken care of by a qualified consultant or a qualified system integrator. There are no standards to these qualifications other than having in-depth knowledge and experience in networked audio engineering and a track record of producing system designs being used in the market.

13. Investing in a networked audio system

System costs

The total cost of a system is the sum of component costs and the labour costs needed to design, build and support the system. Basically a networked audio system increases component costs and decreases labour costs. The investment in a networked audio system also has influence on the costs of usage and maintenance of a system after it has been delivered. Using networked audio systems in the touring industry allow for significant cost savings in logistics and set-up time. Installed systems benefit from the low cost of ad-hoc system changes.

Component costs

With P2P and audio networks having replaced the analog long distance cabling, the corresponding component costs moved from the cabling to the interfaces. Where P2P based systems (eg. AES10 - MADI, AES50 - SuperMAC) need proprietary hardware routers at every connection point, networked systems can utilize less expensive standard network switches. If only two devices - eg. a single mixer and a single stagebox - need to be connected the difference is not so high, but as soon as systems become more complex, audio networks often are more cost efficient.

Labour costs

With P2P based systems the cabling requirements are directly connected to the functional requirements of the system, and has to be carried out by experienced staff. For networked audio systems, within bandwidth limits, the cabling is completely separated from the functional requirements, allowing less experienced staff to be involved. For installed systems, any functional change after the commissioning can be implemented without affecting the cabling, further reducing the involved labour costs.

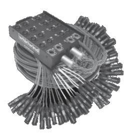
Competitive benefits

A networked audio system has a much higher quality and functionality level compared to a P2P system. As projects grow to be more and more complex every year, an increasing number of jobs simply can not be done anymore without the use of networked audio systems, giving the investor in such systems a clear competitive advantage over P2P solutions. This competitive advantage should also be included in cost calculations.

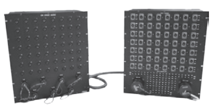
The bottom line

Every system has its own economics, there are too many variables to propose sensible basic rules on cost comparison. In general when replacing analog as well as P2P designs with a digital networked audio, the component costs might be less or equal, the labour costs will decrease and the competitive benefits will increase. The larger or more complex the system, the higher the cost savings.

component cost savings
(compared to analogue cabling)



no analogue multicore needed



no break out, stage snake & splitter needed

networked audio system costs

investment in network equipment & cabling

investment in I/O equipment

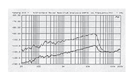
labour cost savings

touring: save on transport, roll-out

Install: save on cabling work

Competitive benefits

increased functionality - flexibility



increased audio quality

14. Networked audio glossary

AES/EBU
A digital audio format standardised by the Audio Engineering Society and European Broadcasting Union as AES3. Uses balanced copper cabling with 2 channels per connection.

AES67
A list of recommendations from the Audio Engineering Society for transporting high-performance audio in IP networks. It is not an audio transport protocol of itself.

AVB Audio Video Bridging
A type of network using specialist switches and other equipment that have implemented the AVB set of protocols. This guarantees synchronization, bandwidth and consistency.

AVnu Alliance
A forum dedicated to advancing the use of AVB and the interoperability of equipment manufactured by its members.

Bridge
A network device used to connect networks together. Bridges work with MAC addresses, they ignore IP addressing. To connect networks on IP addressing level a router must be used.

Broadcast
The 802.3 Ethernet standard allows information to be sent to all devices on a network as broadcast packets. EtherSound™ uses this method to send audio channels on a daisy chain.

Bundle
A CobraNet™ information packet containing up to eight audio channels at 48kHz. A bundle can be 20-bit or 24-bit, and can have 1.33, 2.66 or 5.33 ms latency.

CAT5
Category 5 cable capable of carrying 100Mb worth of network signals over a maximum distance of 100 meters.

CAT5E
Extended specification of CAT5 cable for higher frequencies. CAT5E cables can handle gigabit ethernet connections.

CobraNet™
A network protocol that uses Ethernet to transport audio as well as control and monitoring data over a 100Mb network with a maximum channel capacity of 64 channels per link.

Daisy chain
Method of connecting devices. In case a device fails the system is cut in two.

Dante
A multi-channel digital media networking technology based on IT industry standards, with very low latency and highly accurate synchronization. Dante™ supports a channel capacity of more than 500 channels per link.

Dual link
CobraNet™ redundancy method by connecting a device to a network with two links; if one link fails the other takes over

EEE Energy Efficient Ethernet
Also known as 'Green Ethernet', standardized as IEEE 802.3az. It is intended to reduce the power consumption of switches by around 50%. However, it is not always compatible with audio network equipment such as Dante, so is best avoided or disabled.

End of Loop device
EtherSound™ version 2.09 and up, including ES-100, allows the creation of multiple bi-directional segments in a daisy chain. In addition to the Primary Master, any devices can be set to End Of Loop mode, blocking the upstream data.

ES-100
An advanced version of EtherSound™ offering increased functionality. ES-100 allows a redundant ring topology to be used.

EtherCon®
An RJ45 connector combined with road proof XLR housing, manufactured by Neutrik.

Ethernet
Most commonly used network protocol in the world, standardised by the Institute of Electrical and Electronics Engineers as the IEEE802.3 standard.

EtherSound™
a
A network protocol that uses Ethernet to transport audio as well as control and monitoring data over

100Mb network. EtherSound™ uses a daisy chain topology with a 64 channel fixed bandwidth data flow and deterministic (variable to the network topology) low latency. An advanced version of EtherSound™ with increased functionality has been introduced in 2006 as ES-100.

Fiberfox®
A road-proof system to connect fiber cables, dispersing the light signal with a lens to increase the contact surface of the connector. The large surface is less sensitive to scratches and dirt.

Fiber
A medium used to transport information using light. There are single mode and multi mode fibers. Fibers can handle high bandwidth information streams and can be several kilometers long.

GBIC Giga Bit Interface Converter
A hot swappable module to add gigabit copper or optical connectivity to a switch.

Gigabit
One billion bits (1,000,000,000 bits; Gb). A Gigabit link can carry one Gigabit per second worth of information; 10 times more data compared to 100Mb links (100 Megabits per second, a.k.a. fast Ethernet).

Global Address
An IP address allowed to connect to the Internet. Global addresses are allocated by InterNIC (www.internic.org) in order to keep every global address unique.

Hub
(Repeater hub). A simple network device that resends incoming packets to all ports without checking addresses. Repeater hubs can be used to connect network segments together forming one big network. Repeater hub technology is obsolete and should never be used in new systems.

IGMP Snooping
A switch function, allowing the device to listen to Internet Group Management Protocol messages. This means the switch can block multicast traffic from going to places it is not needed, thus freeing up network bandwidth.

IP address
Internet Protocol address, a user definable address to manage information streams on a network. The IP address includes a network number and a host number. It allows information to be routed on a local area network (office network, normally using IPv4 with 4 byte adressing) as well as a wide area network (the Internet, normally using IPv6 16 byte addressing).

Latency	(Network latency, forwarding delay). The time it takes for an information packet to travel from the sending device to a destination device.
Loop Back device	The EtherSound™ Loop Back device sends it's data not only downstream to the next device as broadcast packets, but also as upstream to the Primary Master device (or the End Of Loop device, V2.09 or higher) as Unicast packet , creating a bi-directional daisy-chain segment between the two.
MAC address	Media Access Control, an addressing system using a 48 bit (6 byte) address, allocated by the IEEE standards organization. 48 bits equals 280 trillion unique addresses, there are no doubles.
MADI	Multichannel Audio Digital Interface**,** an audio protocol standardised by the AES as AES10. MADI uses a single connection to transfer 64 channels of 24-bit audio
Managed switch	A switch with extra capabilities such as handling VLAN's, Trunking, Spanning Tree, Quality of Service, statistics gathering, error reporting.
Media converter	A device to convert a fiber connection to a copper RJ45 connection and back. Media converters are available for most fiber connectors and speeds.
Megabit	One million bits (1,000,000 bits; Mb). A fast Ethernet link can carry 100Mb per second worth of information, a gigabit link carries 1000Mb. In this document a connection speed or bandwidth of 100 Megabits per second is abbreviated as '100Mb'.
Multicast	The 802.3 Ethernet standard allows information to be sent to multiple devices on a network as multicast packets. The information can be picked up anywhere in the network.
Multi Unicast	Some audio devices support sending information to a limited numbet of destinations as 'multi unicast'. For sending bundles to many destinations, multicast must be used.
Multi mode fiber	Connections capable of handling large datastreams over a distance of up to 2 kilometers depending on the network standard. Multimode connections use an inexpensive laser type.
Network class	Categorisation of a network's subnet mask; determining what portion of the IP address is the network number and what portion is the host address. Class A: 1 byte (8 bits) network number, 3 bytes (24 bits) host address. Class B: 2 bytes (16 bits) network number, 2 bytes (16 bits) host address. Class C: 3 bytes (24 bits) network number, 1 byte (8 bits) host address. Small office networks mostly use class C.
OCA Open Control Architecture	A system control and monitoring interoperability architecture, designed to simplify the design and integration of professional media networks.
OpticalCon®	Neutrik XLR connector housing for LC type fiber connectors, protecting the vulnerable fiber ends from scratches and dirt.
OPTOCORE®	A ring topology audio network standard capable of handling more than 500 channels, video and serial connections with low latency.
OSI model	A standardised model for network protocols published by the International Organization for Standardization ISO (www.iso.org). The OSI model defines seven layers, defining the physical form of electrical data (layer 1) up to the network service application that uses the network (layer 7). MAC addressing is defined in layer 2; IP addressing in layer 3.
Preferred Primary Master	EtherSound™ ES-100 devices can be used in a redundant ring topology, setting one device as Preferred Primary Master. This device then blocks the ring (so it's a daisy chain), but unblocks it when a connection is lost.
Primary Master	The first device in an EtherSound™ daisy chain is called the Primary Master, starting the 64 channel data stream sent downstream through the daisy chain. In bi-directional mode the Primary Master is the last device to receive the upstream data. A computer running ES Monitor software can be connected to the Primary Master's IN port to monitor and control all EtherSound™ devices in the network.
Private Address	IP address to be used for private networks without getting approval from InterNIC. Class A: 10.0.0.0-10.255.255.255 Class B: 172.16.0.0-172.31.255.255 Class C: 192.168.0.0-192.168.255.255. These are non-routable addresses and are restricted for use only within a local subnet.
PTP Precision Time Protocol	Clock synchronization standard IEEE 1588. It uses a master-slave architecture and has sub-microsecond accuracy, making it suitable for use by audio network equipment.
QoS Quality Of Service	An Ethernet functionality allowing switches to provide certain types of data with a higher priority, supporting a faster transport through a switch. Dante™ uses this to assure low latency.
Ravenna	An unlicensed, IP based media network using standard IT protocols and equipment. It is primarily used in the professional broadcast industry, and manufacturers are able to participate in its ongoing development.
Redundancy	Designing networks with extra functionality to automatically recover from failures in the system.
Ring	A daisy chained network with both ends connected, forming a ring. Unlike a daisy chain, a ring that can transmit data in both directions has built in redundancy: in case of a failure all devices are still connected.
RJ11	Connector used for copper cabling in phone applications.
RJ45	Connector used for copper cabling in network applications (e.g. CAT5E)
Router	Network device used to connect networks together. A router works with IP addresses and is capable of routing data between connected networks with different network numbers. Routers are rarely used in networked audio systems.

RS232	Serial connection standardised by the Electronics Industry Alliance (EIA) defining electrical and mechanical characteristics, supporting low bitrate P2P connections. In 1991 an upgraded standard RS232C was introduced.
RS422	Serial connection standardised by the Electronics Industry Alliance (EIA) defining electrical and mechanical characteristics.
RSTP	IEEE802.1w Rapid Spanning Tree Protocol, a faster version of the IEEE802.1d Spanning Tree Protocol.
Serial bridge	Serial connection within a CobraNet™ network allowing the use of the network to communicate with RS232 devices.
Serial server	Device to convert RS232 or RS422 into Ethernet and back, so serial signals can be used through a network.
SFP	Small Formfactor Pluggable, a mini version of GBIC's.
Single mode fiber	Connections capable of handling large datastreams over a distance of up to 80 kilometers depending on the network standard. Single mode connections use an expensive high power laser type.
SNMP	Simple Network Management Protocol, a standards based method of controlling and monitoring devices in a network.
Spanning Tree Protocol	IEEE802.1d Ethernet standard. A protocol for Ethernet switches to block loops in networks and reserve them for use if an active link fails.
Star	Most commonly used network topology. The center of the star can be designed with high processing power switches, while the ends of a star network can be designed with less processing power. 'Star of stars' or 'Tree' structures are also common variants of this topology.
STP	Short for Spanning Tree Protocol or Shielded Twisted Pair.
Subnet mask	A number that specifies what part of an IP address represents the network number and what part the host address.
SuperMAC	A point to point audio distribution standard from Oxford Technologies, standardised by the AES as AES50. Transfers 48 channels of 24-bit 48 kHz audio through a CAT5 cable.
Switch	A network device that connects network components together. Switches are intelligent hubs, forwarding incoming packets only to ports connected to the packet's target address.
Topology	The way network devices are connected in a network. Basic structures are a ring, daisy chain, star, tree.
Trunking	Using two or more cables to connect switches supporting the IEEE802.3ad Link Aggregation functionality; allowing the use of two or more connections to act as a single higher capacity or redundant connection.
Unicast	The 802.3 Ethernet standard, allowing information to be sent to only one specific device on a network. Since the transmitted packets only use up bandwidth on the ports and cables on the route from transmitter to sender, the network can support more connections compared to Multicast.
UTP	Unshielded Twisted Pair. Most commonly used is category 5; CAT5.
VLAN	Virtual Local Area Network. A managed switch can separate network traffic into two or more 'virtual' networks using the same hardware.
Wi-Fi	Wireless networking standard IEEE802.11. Most used varieties are 802.11.b (11Mb/s), 802.11.g (54Mb/s) and 802.11.N (up to 150Mb/s)

Useful websites

www.aes.org	Audio Engineering Society, AES3, MADI/AES10, AES50, AES67
www.audinate.com	Dante
www.aviom.com	A-net™
www.avnu.org	AVnu alliance
www.cisco.com	Cisco
www.cobranet.info	CobraNet™
www.dlink.com	Dlink
www.ethersound.com	EtherSound™
www.hp.com	Hewlett Packard
www.ieee.org	Institute of Electrical and Electronics Engineers
www.iso.org	International Organization for Standardization
www.internic.org	ICANN Internet Corporation for Assigning Names and Numbers
www.lightviper.com	Lightviper™
www.ravenna.alcnetworx.com	Ravenna
www.riedel.com	RockNet™
www.oca-alliance.com	OCA alliance
www.optocore.com	OPTOCORE®
www.parc.com	Palo Alto Research Center
www.sonyoxford.com	SuperMAC/AES50
www.tiaonline.org	Telecommunications Industry Association
www.yamahaproaudio.com	Yamaha