Technology Snapshot: Digital Network & Transmission Protocols

Getting sound from one place to another is a lot more complicated than it sounds. Luckily, quite a few clever people have done the hard work for us, and there are now a good choice of systems and protocols for the digital journey. Stephen Bennett picks out the gems…

Protocols – don't you love 'em? Just as you've got used to one, along comes another younger, better, faster, and sexier standard to knock the socket on the back of your desk into touch. With the increased popularity of digital consoles in the live and broadcast worlds, faster and more capable interfacing has proliferated. However, it's common for many manufacturers to have different ideas of what constitutes the perfect connection for their digital audio gear.

Protocol Genealogy

The great granddaddy of connection protocols was, of course, MIDI (Musical Instrument Digital Interface). It's hard to imagine in these days of digital cooperation what a revelation the ability to connect and control the equipment from many different manufacturers was. It's proved amazingly resilient to the developments in audio technology, and is still used to control DAWs and other software and hardware directly from control surfaces. Of course, MIDI is an 8-bit data-only protocol, so when you want to distribute digital audio along with your data, things need to be speeded up a bit. S/PDIF (Sony/Philips Digital Interconnect Format) and the AES/EBU (Audio Engineering Society/European Broadcasting Union) protocols were created to distribute stereo audio data at varying sample and bit rates, while Alesis's ADAT connection allows for the transfer of eight tracks at 48kHz down a single fibre optic cable.

All are well established – but today's multi-channel, high data rate world requires new improved protocols to get the audio from A to B via C, and possibly D.

The advantages of using digital transmission for multi-channel audio over the long distances used in live and broadcast applications are that the signals are immune to radio frequency and mains-borne interference because the systems don't need to use thick, expensive, audio grade multi-core cables – usually just a simple CAT 5 cable will do. Lower cost is also a factor, with some companies implementing the less expensive solutions (or their own propriety protocols) in their cheaper consoles and other hardware. Most of these technologies are based on bog-standard Ethernet hardware and audio transmission, and can often work together with existing distributed networks. However, the need for asynchronous multi-channel data transfer requires software and hardware that needs to be able to cope with these transmissions without drop-outs and at useable low latencies.

Physically, most of the available systems are similar, but differ in their software implementations, with some using standard networking protocols, some proprietary, and some 'standard' transmission protocols. They are usually available directly via consoles, on computer-based cards or as stand-alone rack mount hardware. Data is sent via standard CAT 5 Ethernet cables or fibre optics. On either end of the system lurks a mixing console and/or stage or distribution boxes, or hardware interfaces for direct connection of line, microphone, and digital signals.

RSS by Roland's REAC-based snake system.

MADI

MADI (Multi-channel Audio Digital Interface) is an obvious successor to the basic stereo interfaces described above. The AES document AES10-2003 describes the protocol that has features in common with AES/EBU, and is capable of coping with up to 64 channels of 24-bit audio at sample rates of up to 96kHz over various cable types and over long distances. Use of optical fibres makes transmission of large numbers of channels a doddle, and several companies feature MADI in their consoles including Studer, AMS, Neve, Fairlight, and Lawo.

REAC

Roland's S-4000 series of digital snakes use the REAC (Roland Ethernet Audio Communication) protocol (www.rolandsystemsgroup.net/en/0111d.htm). The REAC specification provides a high quality, redundant digital audio transfer system that can be easily installed or integrated for any audio snake application. REAC is a 'plug and play' system that is easy to configure and requires no complex setup via computer operation. Running over Ethernet, REAC has extremely low latency and is capable of transferring up to 40 channels of 24-bit linear audio at 96kHz while generating a latency of only 0.375ms.

CobraNet

Cirrus Logic's CobraNet (www.cobranet.info) was the first successful implementation of multi-channel audio transmission over Ethernet. Sixty four channels of uncompressed audio can be transmitted over a single CAT 5 cable, and Cobranet is particularly useful in networked or distributed systems. However latency in CobraNet is relatively high (1.33 to 5.33ms) which may make it unsuitable in live situations. CobraNet is a licensed technology and OEM implementations are available for use in third-party products such as Yamaha's MY-16 compatible digital consoles, D&R broadcast digital consoles, and SoundCraft's Vi Series.

Ethersound

Digigram's Ethersound (www.ethersound.com) is another protocol designed to carry multiple channels of audio over standard networking cables. Up to 64 channels of 24-bit/48 kHz PCM audio plus various embedded control and monitoring data are transported via a single cable, and the protocol is designed to provide low-value, predictable latency figures. Companies implementing the Ethersound protocol include Allen and Heath, Focusrite, DiGiCo and, of course, Digigram itself.

Optocore

Optocore (www.optocore.com) is a standalone modular optical fibre network technology that features a synchronous, redundant, optical ring network capable of transporting audio, video, data, and word clock over long distances. All the various Optocore modules can be combined to offer the maximum flexibility in terms of layout, number of channels, and types of signals. SANE (www.optocore.com) is a low cost Ethernet based implementation of Opticore's core technologies consisting of hardware interfaces with AES/EBU, microphone, line level, and Ethernet connections. Further hardware interfaces sport Opticore's fibre optic transmission connections for creating larger networks. One of Opticore's suggested layouts uses the fibre network for building-to-building connections, while CAT 5 Ethernet cables are used for breakouts for distribution within each location. SANE and Opticore can be used together in all their synchronised word clocked glory for maximum flexibility. Optocore has been incorporated into Yamaha's product range, including its YG2 and YS2 digital audio distribution cards.

AES 50

As always, as soon as a protocol becomes a 'standard' some upstart steps up to the front and tries to improve it. The AES 50 – 2005 protocol, an Ethernet based system and the associated SuperMAC and HyperMAC technologies (www.supermac-hypermac.com) were initially developed by Sony. However, both Klark Teknik and Midas have been closely involved in its recent development, while Lynx Studio Technology's AES16e-50 PCI card features AES50, enabling it to offer up to 32 channels via a standard network cable.

Optocore system components.

The implementation of the protocol in Midas's XL8 is called MidasNET, and uses a CAT 5 or 6 cable for connectivity. All digital audio is transmitted using the AES50 protocol (implemented as SuperMac) and the HyperMac high-capacity system. These protocols benefit from extremely low latency, robust feed-forward error connection, and advanced system clocking. The use of this AES standard means simple, straightforward interfacing with any third party hardware that features this connection.

Dante

Audinate's statement on its website that its technology is 'the future of audio and video technology….' Is quite a claim! (www.audinate.com) DANTE does its business using bog standard Ethernet hardware and TCP/IP protocols (which is exactly what the Internet uses to swish data packets around), and the company states that 'digital audio networking is easy, intuitive, cost-effective, and error-free with no need for any specialised knowledge.'

Blakes Seven fans will be please to know that Zen, the Liberator's shipboard computer, seems to have been repurposed as a utility for detecting DANTE-aware hardware on the network. The company aims to remove the proprietary nature of digital distribution solutions and provide a low latency, easy to use universal protocol. Using TCP/IP means that DANTE is routable through standard data networks (or even the Internet) and allows for the direct connection of personal computers. Having just spent a week soldering an analogue multi-core cable to a batch of XLRs, the thought of sending the lot through the CAT 5 sockets in both recording and control rooms is mouthwatering. It's early days, but apparently Yamaha is already involved in the DANTE project with the Dante-MY16-AUD card, and it can't be long before others come on board. It'll be interesting to see how Roland respond as its REAC technology is similar in many ways – and also allows for direct computer connection. Audinate claims that DANTE has 'Sample-accurate synchronisation and industry-leading low latency', which must have been a challenge when using open source industry standard data communication protocols. There are no stand-alone interfaces announced as yet, but a couple of those and a CAT5 cable may be all you need in the studio of the future – and short network cables at that!

Nexus

Stagetec's (www.stagetec.com) Nexus is a (or 'the' according to the company!) versatile audio network and routing system for controlling studio or mixing-console resources, for routing arenas, broadcasting-complex networking, outside broadcast trucks, sound reinforcement, and, indeed, for all other professional audio applications. Distances of up to 100km can be spanned using optical fibres and its 19-inch rack mount interfaces allow for any audio signal to be sent over these distances. The system is capable of transmitting data other than audio, and is controlled by a PC-based software program. Systems can be configured with a high level of redundancy, and every device includes a dedicated controller board. 24-bit audio can also be processed using DSP, can be routed to wherever signal processing is needed, and provides all necessary controls for EQ, delay, dynamics, faders, and so on, all controlled from the NEXUS user interface.

Conclusion

As with any interfacing technology its wider acceptance depends on its ability to deliver the required results with the performance required in real-life situations. All of the current protocols achieve these goals at their respective price and performance levels. If devices from different manufacturers are going to be connected together, it's obviously important that they all speak the same communication language. But in this area, a plethora of protocols shouldn't be too much of a bind and improvements in technology, bringing higher channel counts and sample rates with lower latency should mean that new protocols are a welcome sign that the industry is moving forward.