NEXUS

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The audio router NEXUS by Stage Tec, Germany, is considered to be the originator of distributed audio systems. From the very first, NEXUS Base Devices formed a complete network that enabled inputs and outputs to be interconnected irrespective of their position in the network. The fibre-optic cables between the individual Base Devices enabled these networks to cover long distances, making links between rooms, buildings and later even cities. As an answer to the ever increasing number of audio channels in networks, the capacity of NEXUS systems has grown over the years. The most recent step towards huge, flexible and dynamic audio networking was the introduction of the new NEXUS XRT router board which can route up to the immense number of 8000 channels on one board.

Since some years, there is also the strong trend towards Audio IP networking, with the Dante protocol by the Australian company Audinate as market leader. Dante is basing on a completely different concept than NEXUS: The one is using IP packet transfer on a standard Gbit Ethernet network (Dante) while the other one routes the audio samples by the help of a time division multiplex bus system (NEXUS).

Which system is good for which purpose? Which one is the best choice for your application?

Extensions through Dante

Dante networks are charming simple. Just connect your device to the standard Ethernet network, configure the Dante channels – and that's it. Now you can transmit uncompressed audio through the network. The advantage of this concept is that you don't need dedicated lines, not even dedicated hard-

ware for routing the signals. This way of transmitting audio is ideal when you need to send some signals to distant locations which are reachable through an existing network. For example, if your OB truck connects to the network of a stadium, you can use this network for distributing the microphone signals from the sports ground into the OB truck's audio system. That way you avoid laying an own temporary cabling for the time of the sports production.

Audio timing

The IP transport on Ethernet cuts the audio signals into packets. Each packet is then transmitted in asynchronous mode through the network and assembled in the right order again at the receiver's side. The drawback of this procedure is that the routing of the packet through the network need time.

In contrast, the TDM based NEXUS is a fully synchronous system where audio is transferred with the highest priority, guaranteed timing and with the lowest possible latencies. One of the main advantages of TDM technology is and will always be that these systems are absolutely superior in guaranteeing the lowest possible latencies. NEXUS route audio within a single audio sample or – in large networks – within about four samples. At a sample rate of 48 kHz, this is only 20 (or up to 80) microseconds; at 96 kHz, this reduces to just 10 (or up to 40) microseconds.

With IP switching such values are inherently unattainable. When routing audio over IP using Dante, the lowest possible latency is 250 microseconds, which is a whole magnitude greater than the latency values achievable using TDM. Furthermore, this performance is highly dependent on the network topology and good network design. The more switches there are in the audio transmission routing path, the longer the network delay. With up to ten switches the delay of a Dante transmission can be set to one millisecond. In larger networks with more switches the delay time will be adjusted to five milliseconds.

Another timing problem arises in IP audio networks when several channels are to be switched at the same time. By principle, the simultaneity of this switching cannot precise because the IP packets of each channel are routed through the network individually. In case you want to switch all audio lines from one studio to the other, you better use TDM rather than IP based technologies.

Not only the latency but also all questions around synchronisation, digital clock distribution throughout the system and audio jitter are answered much more professional in NEXUS than in any IP audio network. If the audio signal is relevant for broadcast or for any other high-quality application, the low jitter ratio of NEXUS convinces, as do the reliable synchronisation which avoids digital clicks and other disruptions.

Reliable audio distribution

In highly professional environments, the monitoring of the audio systems is an important factor. NEXUS offers a sophisticated internal system monitoring. A NEXUS network can even be included in an SNMP monitoring (Simple Network Management Protocol), allowing to centralise the management of the audio routing system. This approach is often used in large broadcast stations and is very common for example in the German home market of NEXUS.

Highly reliable audio routing can be guaranteed by adding redundancy. An IP system can offer cable redundancy. If you use dedicated hardware such as the NEXUS STAR, you can also add CPU redundancy, you always have redundant power supplies, you can hot-swap all components at any time and you can even set up fully redundant mirror systems.

Because of the decentralized architecture, global NEXUS system breakdowns are virtually impossible. Thanks to the high-redundancy structure, even connection board failures cannot cause the system to grind to a halt! The reliability of a NEXUS system is such that it can even be used as the heart of an emergency alert and evacuation system that conform to European law standards.

Additional functions

NEXUS also offers a wide range of additional functions. By the help of the internal NEXUS Logic Control, a NEXUS audio network can control the entire switching operation of a broadcast station. You can switch all relevant lines from one production studio to a control room by the press of a key. One of the most elaborate examples is the installation at Radio Bremen, where hundreds of logical functions were programmed and now control the entire studio to production delegation every day. Many broadcasters use the NEXUS logic functions also for checking the on-air signal output: If the signal is too low or is cut off completely, the NEXUS displays an error. NEXUS can also switch to an alternate signal and give out an alarm trigger. These kind of monitoring and watch-dog functions are not possible with any IP based audio routing system.

Stage Tec, the producer of NEXUS, is also offering several digital mixing consoles which can be integrated into a NEXUS audio network. All of these consoles have in common that they use a NEXUS network for providing the I/O interfaces of the consoles. NEXUS also serves as host to the mixing-console processing boards. Thanks to this proven design, the mixing consoles are ideally integrated into the audio network. This is true even for large installations with plenty of NEXUS I/Os and plenty of Stage Tec audio consoles.

Combination of both

Both methods – distribution of audio over IP on the one hand and TDM as used in dedicated professional routing systems such as NEXUS on the other, have very different pros and cons. The technologies complement each other rather than the newer one being on course to replace the older completely in the foreseeable future. The consequence is to combine both ways of switching to gain the maximum advantages. For doing this, NEXUS offers an interface board for Audinate's Dante system. The so called XDIP board connects NEXUS to networks running Dante, which enables powerful audio and media networks to be created on existing standard Ethernet infrastructure. The sample rate converters on the NEXUS XDIP board allow the asynchronous Dante audio signals to be used in sync with all NEXUS audio signals.

Coming back to the example from the beginning, the NEXUS XDIP board connects the OB truck to the Ethernet infrastructure of a sports stadium, allowing all signals within the NEXUS network to be used on the sports grounds for monitoring purpose and vice versa feeding all signals from the location into the audio system of the OB truck and its mixing console. And the user has got the best of both audio worlds!

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