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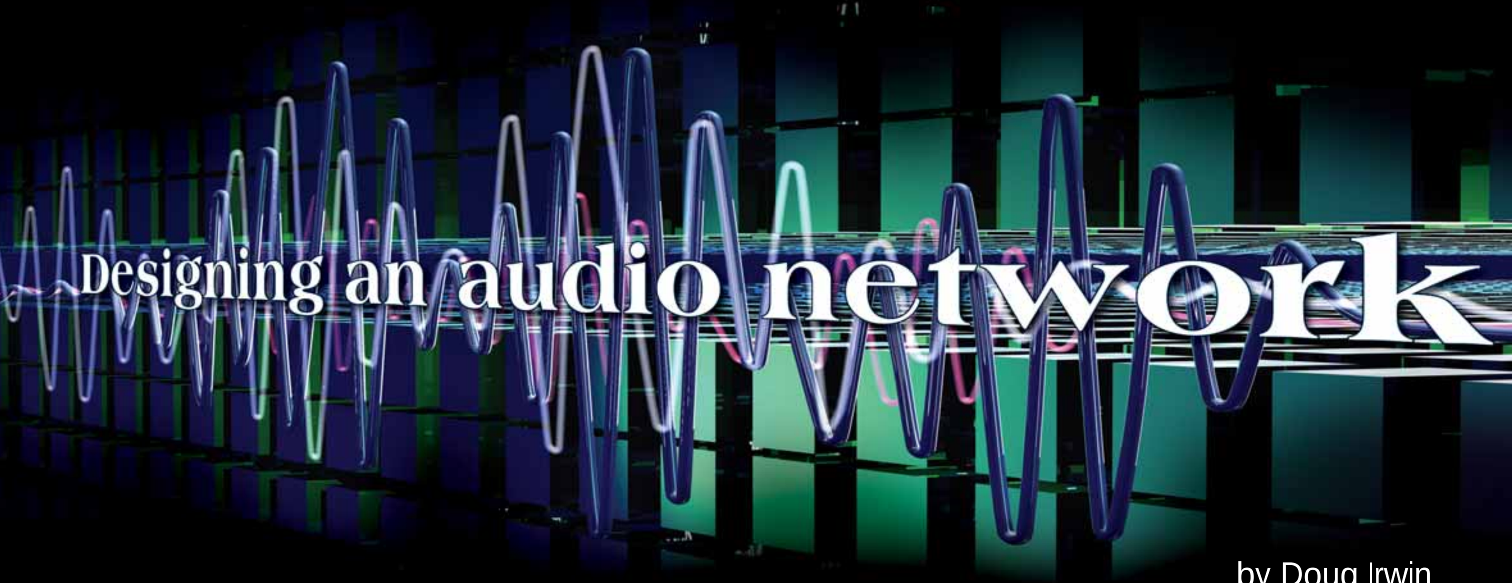

Designing an audio network

by Doug Irwin

It's time to design a facility. You need to route and control audio and its related control functions. Today you're faced with a choice: stand-alone consoles or an audio network. It isn't the point of this article to review how radio studios were designed and organized in the old days, but it will be instructional to go over those techniques briefly so that the benefits of newer technology are more clear and evident.

It used to be that you would first decide how many studios were needed on a per-station basis. Typically the answer would be one on-air studio, one production studio, and perhaps another combination studio that could be used as a backup for the on-air room and for less complex commercial production features (such as dubbing agency spots and adding tags). If you had "n" radio stations, you usually just multiplied the number of studios by "n" to get the final studio count.

Based on the studio requirements, you picked a console of the appropriate size.

Typically a production console was more suited to that function. Often they were larger, with more inputs, more output buses and so on. All the equipment that was needed for the particular studio was located in that particular studio—everything from tape machines to CD players to DAT machines to DAWs.

Invariably there was rack space somewhere in the facility that was effectively a terminal for all the studio programs, and likewise a jumping off point to the transmitter site.

After the Telecommunications Act of 1996, more and more radio stations were crammed in to the same space; and for this reason, more and more equipment was shared between stations under the same roof. In the early 1990s ISDN codecs came in to their own, and often landed in the terminal area racks so that they could be shared. Because those codecs had to receive and send audio, sharing them between stations required complex audio switching. Their inherent delay necessitated a mix-minus feed, which complicated things even further. More and more stations started taking satellite feeds for chunks of the day—and those satellite receivers were typically located in the terminal room racks as well.

As facilities grew larger with more stations, the old way of sharing audio resources—such as using DAs and audio switches that were either push buttons or relays—became more cumbersome. New facility builds often included audio routing switchers that, while expensive at the outset, simplified the construction and later provided new levels of convenience and performance that had not been obtained before.

But even with the "older" type of audio routers, radio stations were pretty much built the same way: using a spoke and hub topology. Any studio would be a mini hub with most of the equipment needed on a daily basis located in it. The terminal rack room would be the major hub with cable runs (spokes) going out to all the various studios. The spokes would consist of fat multi-pair cable that was expensive to buy. Each end of the spokes typically terminated on blocks—many times the ubiquitous 66 blocks or something more modern such as ADC Icons or Krone blocks. A large part of the labor going in to any facility build was the installation of these cables and punching down both ends. The design of a studio was typically to overbuild, because the last thing you wanted to have happen was for all the spoke (trunk) pairs to get used up. That was bad.

One of the final parts of the facility construction was the addition of cross-connects to connect the various pieces of equipment to the trunks themselves, making everything talk. Making changes later during the life of the facility involved literally moving wire pairs or adding new ones so that equipment was physically connected to where it needed to be.

Fast forward

The future is here and instead of building individual radio stations we now consider what is known as an audio network. This new term reflects not only the function, but the methodology as well.

Designing an audio network

With the explosion in computer networking over the last 10 years, audio equipment manufacturers have taken notice of the new ways in which information (whatever that information consists of) is moved from one point to another. The most basic change is that one pair of wires is no longer just assigned to one static function. In the old days, one pair of wires might be used to carry audio from a DA output that was assigned to a remote broadcast line to the on-air studio. With each pair of wires performing only one static function, hundreds of pairs needed to be purchased, installed and punched down.

Which would you rather connect? Both can be used to carry multiple signals.

In an audio network, one twisted pair can literally carry hundreds of signals, whether they consist of digitally encoded audio, or control or other ancillary data.

Because communication between nodes of an audio network is done in a digital format, CAT-5, CAT-5E, CAT-6 or even fiber is used for the actual communication. This type of cable is produced on a massive scale, unlike the fat audio cables that are still available. Consequently, the cost of this type of cable is drastically less than that of fat audio cables. Another thing to consider is that it is easy to find plenum-rated data cables, and this simplifies the construction process as well. And if you need to run AES3 audio signals, CAT-5 cable works well.

The addition of routers to a radio facility made more effective use of many of the wire pairs that were originally installed—and the coming of digital audio routers doubled that efficiency yet again (after all, an AES data stream includes left and right channels).

Equipment manufacturers then saw the inherent efficiency of computer networking, and began looking at moving audio around the radio station facility in a similar fashion via fast data streams (in synchronous or non-synchronous modes). What is known as the physical layer in the language of computer networking—the cable types, the connectors and patch bays—could also be used in the transmission of these high-speed data streams.

At least one manufacturer actually makes use of Ethernet to move audio and control signals around. And the era of the audio console might well be coming to a close because many manufacturers now offer control surfaces that, while they look like consoles, are simply human user interfaces for a remotely located routing switcher.

Spoke and hub topology

The spoke and hub remains the logical topology for an audio network. It is one of the most basic methods of communication between multiple locations. This method is used by airlines, the post office, the phone company and many others.

Radio stations still need separate studios to carry out specific functions, such as on-air and commercial production. Most stations have a rack room or some other location that is the heart of the facility, as shown in Figure 1. This is where the audio network hub (in the spoke and hub context) is located. Several manufacturers make use of a peer-to-peer variant on the spoke and hub idea, by use of a device commonly known as an audio engine. This is a device that handles the cross-point switching and other functions that would heretofore have been accomplished in a router. A unit such as this would be at the hub (in the spoke and hub context) but also, in some cases, in a studio. Most units such as this are mainframes, and as such have I/O cards to receive and send audio signals, in the analog or digital formats; logic I/O cards; DSP cards for audio processing (including mixing); and finally specific cards for communication with other audio engines (peers) or peripherals (spokes) in the audio network such as control surfaces. Shared audio devices (such as a satellite receiver) located in a rack room would connect to an audio engine located in the same room.

The studios themselves still have specific items that are needed locally (such as a CD player, for example). To avoid a wire run all the way back to the master control room audio (because after all, that was part of the reason for the network, right?) an audio I/O device still needs to play the role of interface between the audio network and the local device itself. Some manufacturers make special I/O devices and some simply specify that a stripped-down version of an audio engine be located in the studio. In some cases, an audio engine located in a studio has specific plug-in cards that allow it to work in conjunction with a control surface located in-studio, thus providing all the necessary and familiar console functions. Each audio engine, whether located in a studio, a rack room or some other place, is a peer. Essentially they are all functionally equivalent, and they communicate with one another. Their specific inputs and outputs become available throughout the entire network.

The audio engines

AEQ's offering is the BC 2000D, a mainframe (and hub of the system) that is built to house the various plug-in modules associated with the typical audio network: I/O cards and DSP boards. The control surface for the AEQ system is made up of combinations of the Arena DM (five input modules plus a monitor module) and an Arena D10, which has 10 input modules.

Axia's approach is perhaps the one that is most like a computer network. All signaling is routed via Ethernet. The heart of the audio network is an Ethernet switch that ties all the spokes together. Elements that communicate via Ethernet through this switch include the Axia Studio Engine, which is a 2RU device that uses DSP to accomplish all the mixing functions given it via the Smartsurface. The Smartsurface is the user-interface that looks and functions like an audio console.

Harris offers the Vistamax system. It starts with a mainframe that is loaded with the appropriate analog and digital I/O cards and communications cards, and will typically be installed in the rack room. Rack room sources are integrated into the audio network at this location.

This particular frame would then make peer-to-peer connections with other frames throughout the facility.

Klotz Digital has recently introduced the Vadis 212, a fanless mainframe with 10 freely assignable slots for interface cards accommodating analog or digital inputs and outputs. With the appropriate DSP cards installed, the unit can perform mixing functions along with real-time audio processing functions such as EQ, compression and limiting. Like the other audio engines mentioned, the Vadis 212 operates in conjunction with a control surface, for example the Vadis DCII. (It should be noted in this case that the communications between the DCII and the Vadis 212 is via a proprietary digital interface.)

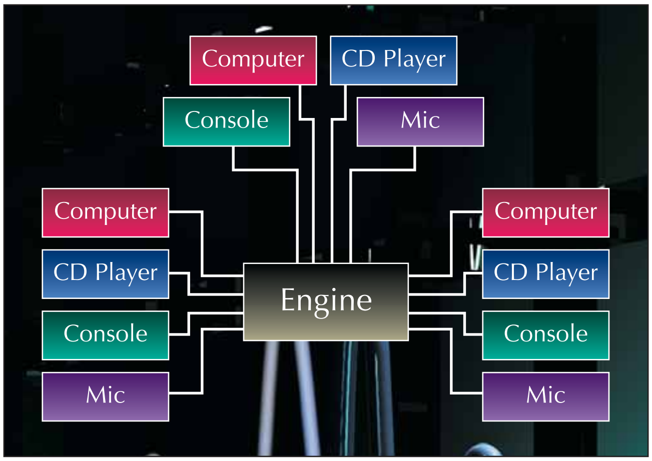

Figure 1. The basic spoke and hub network places the audio engines in a central location, such as a rack room.

Logitek is a long-term player in the audio networking game. Its Audio Engine is a rack-mount main frame that offers the capability of direct connection to multiple control surfaces, and other Audio Engines as well, via fiber. In addition, the Audio Engine is a full X-Y router, and acts as the heart of the audio network. Under commands from the control surface, it performs all the normal console functions, such as mixing, channel on / off and cue. A fully configured Audio Engine can handle as many as 128 mono (64 stereo) inputs and outputs by way of plug-in analog or digital I / O cards. Control, programming and analysis can all be done via a TCP / IP connection.

Sierra Automated Systems offers the 32KD—its digital audio router that performs the requisite routing functions, and others as well, such as mixing, level control, intercom, IFB and mix minus. A single 32KD frame can accommodate 512 inputs and outputs, and multiple frames can be connected together via a fiber-optic link. Separate plug-in modules handle digital inputs, digital outputs, analog inputs, analog outputs and serial interfaces.

Studer offers the Route 5000 digital router system. The core of the 5000 is the mainframe into which the multichannel audio digital interface (MADI) cards are installed. Each MADI input or output card accommodates 28 AES data streams. In addition to the normal routing functions the 5000 offers DSP capability. It communicates with the PC that is the control server via a fiber-optic link. Other control workstations or XY controllers communicate with the control server via Ethernet. The 5000 can communicate with a Studer digital console via a MADI link.

Wheatstone offers the Bridge, an audio engine and routing system that works in conjunction with control surfaces remotely located in studios. The Bridge is a mainframe into which all the necessary cards are installed: analog and digital I / O cards, DSP cards, serial data cards and even one that supports 16 audio streams. Multiple Bridges can be connected via fiber or CAT-5 cables.

To the other end

Turning attention to the opposite end of the spoke, Wheatstone offers, in addition to the Bridge, a satellite router frame that is a scaled-down version of the Bridge. It lives in a studio. This unit holds the I / O cards that are studio-specific; audio peripherals such as CD players, DAWs and monitor amps get their audio inputs and outputs from this device. It will also work in conjunction with a control surface located in the studio. Connections from the satellite to other Bridges are formed via fiber or CAT-5 cable. Once that connection is made, all the peripherals become part of the network.

In the Studer 5000 router system, studio-specific peripherals are first connected to an I / O frame such as the D19M series. Individual I / O cards (such as four-input 24-bit A / D converters, four-output 24-bit

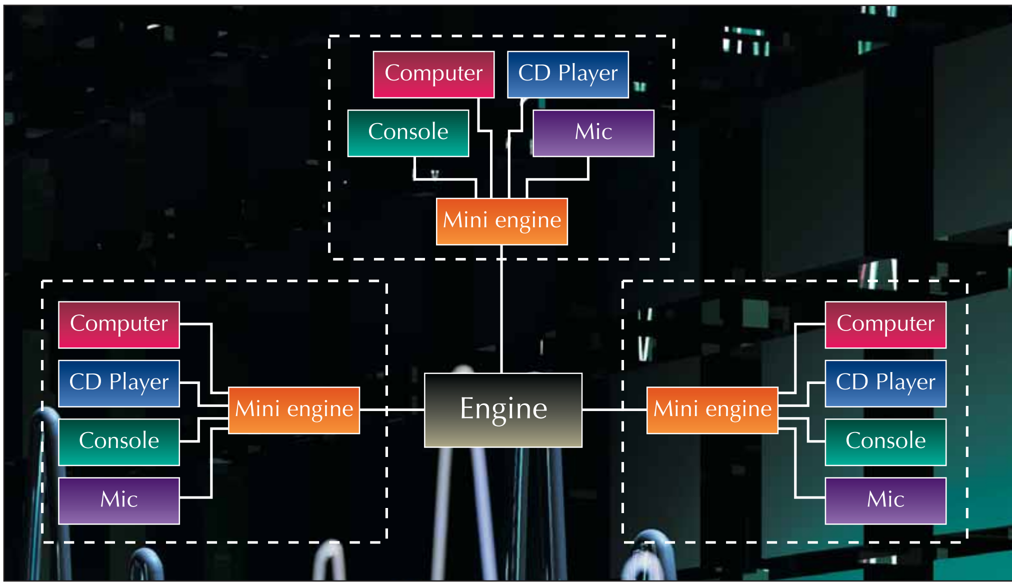

Figure 2. A mini-engine or other I/O device can be used in a studio to further reduce the cabling needed to attach sources to the audio network.

D/A converters, dual-input AES or dual output AES) are plugged into this satellite frame, and are then added to the audio network by way of an MADI connection back to the 5000 router core.

SAS offers the Riolink—a device it calls an extension cord for the 32KD. This device is installed in a studio and serves as the audio and control I/O for the room. The Riolink is actually connected between the SAS control surface (known as the Rubicon) and the 32KD itself. The Riogrande, a new offering from SAS, works in conjunction with the Riolink and can change the Riolink in to a small 32 x 32 mixer in the event of a loss of the 32KD. Connectivity between the Riolink and the 32KD is formed with CAT-5 cable and RJ-45 connectors.

Logitek's approach to the studio-specific I/O is to use another audio engine; all of its power and features are thus available locally. One audio engine in a studio, plus the corresponding control surface, provides the user with the normal console functions plus complete access to the audio network.

Klotz uses a similar approach for the studio I/O. For example, a Vadis 212 frame would be placed in a studio, then connected to a control surface, making up (as far as the user was concerned) a console and at the same time adding all that studio's equipment to the audio network.

The Harris Vistamax system has a unique approach to the studio I/O aspect of the audio network. A Vistamax frame can be located in a studio, providing the appropriate I/O and functionality therein. Alternatively, one of Harris' digital consoles can also serve as a node, or peer, in the network. This provides scalability and allows the system to be integrated with legacy equipment.

Axia's approach to the audio network is to have all the spokes communicate with one another via Ethernet by way of a network switch that is effectively the hub of the spoke and hub topology. Axia offers function-specific 1RU devices it calls nodes that perform functions such as microphone amplification; analog I/O; AES digital I/O; router X-Y control; and finally control I/O. These nodes live in studios or other locations as needed. These nodes, in conjunction with the Studio Engine, the Smartsurface and the Ethernet switch, make up the audio network Axia-style.

AEQ's system uses satellite BC2000 frames linked to other frames in the system for local studio I/O. In conjunction with the modular DM/D10 control surface, the console functions are completely handled, while all sources/destinations become integrated into the audio network.

While multiple cable runs may still be necessary with an audio network, the number of cross-connects and distribution points are reduced.

The explosion of computer networking over the last 10 years has had an effect on the methodology and technology of radio station construction. The appropriate concepts, physical layer materials and devices have been borrowed from the computer network and modified and used completely in others to make up what is now known as the audio network. It's time to take what you have learned about computer networking and apply it to audio, especially if there is a studio build or upgrade on your horizon.

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