Packets Everywhere: How IP-Audio and Ethernet Are Transforming Modern Radio Facilities

Marty Sacks
Vice President, Axia Audio
Cleveland, Ohio
marty@AxiaAudio.com

ABSTRACT

For years, Ethernet has been on the periphery of radio station infrastructure. We've used it for traffic management, file sharing, allowing the Internet into our studios, and some audio playback — and there's where the story ended.

Now, however, broadcasters are finding interesting new uses for Ethernet, thanks to the emergent audio standard called IP-Audio. IP-Audio opens up new capabilities and possibilities for broadcasters because it enables real-time transmission of uncompressed, linear audio alongside direct machine control of audio devices and playout computers, transmission of PAD / metadata streams, direct playout of digital PC content sans audio cards, wide-area connectivity (separate floors or separate buildings), high-spectrum STL data links and more — all using the dramatically reduced wiring infrastructure that the Ethernet environment provides.

This paper will examine some of the innovative ways radio broadcasters are employing IP-Audio and Ethernet for applications such as alternative STL, large-scale studio connectivity, networked audio monitoring, and remote system administration.

HOW IP-AUDIO WORKS

The mechanics of IP-Audio are fairly well known, so I'll give just a brief overview.

To begin, audio sources are connected to "audio nodes" located in studios, server areas or anywhere audio sources exist. The nodes convert analog or AES/EBU signals to uncompressed 48 kHz, 24-bit digital audio, which is then packetized for transport via Ethernet. Audio streams from the nodes are assigned unique IP addresses for identification and routing.

After sources are connected, logic ports on audio devices are connected to "GPIO nodes" which convert start/stop/status commands to packet data also.

Each node connects to a QoS-compliant Ethernet switch, which connect to other switches around the facility, and each node's audio and control data are "advertised" to the network, for consumption by anyone who needs them. Gigabit Ethernet or fiber-optic links between switches provide a fat pipe that can handle tens of thousands of stereo signals per system.

IP-Audio is extremely automation-friendly. Once audio and machine-control commands are converted to data, automated software control of routing paths and switching configurations is not only possible but desirable, allowing routes or entire networks to be remapped easily using a simple software interface.

Of course, all this business concerning nodes is simply the way in which IP-Audio must be implemented *today*. In the near future, broadcast equipment will have IP-Audio interfaces built in, eliminating the audio/data conversion altogether.

Picture how a built-in IP-Audio interface would simplify setup of new audio devices. Traditional broadcast phone systems, for instance, typically need wiring for two audio inputs and outputs, program-on-hold input, logic connections to the console, a serial control connection for PC screening software — somewhere near 20 separately soldered connections.

Now picture that same phone system with an IP-Audio interface. First, you plug an Ethernet cable into the jack on the hybrid; you plug the other end into a network switch. That's it! All that I/O and logic travel the same cable. Just think of how much installation time would be saved during new studio builds!

A side note: IP-Audio technology not only delivers a higher performance / cost ratio ("bang for the buck") than traditional methods of studio building, it saves money outright. Short- and long-term cost benefits can be realized in quite a few different areas, like materials (cabling and mainframe router/switcher hardware), installation (reduced labor), maintenance and troubleshooting (simpler infrastructure) and even in time spent assembling system documentation. Users of IP-

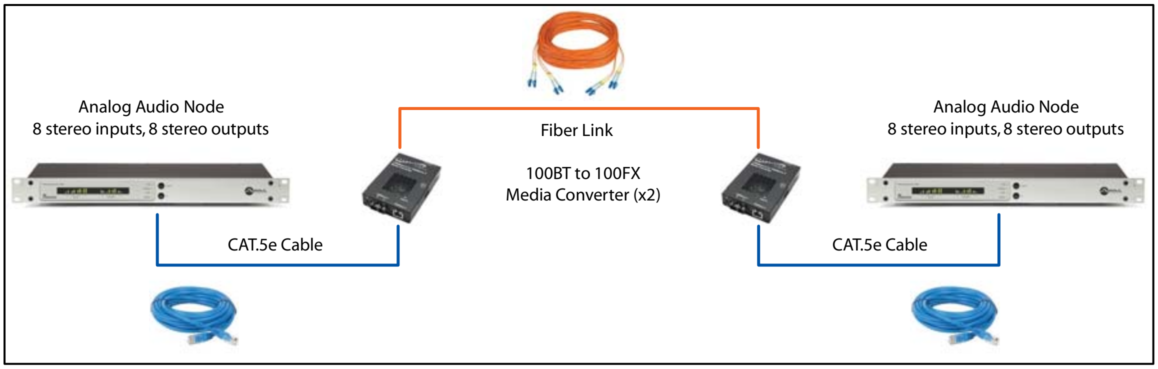

Figure 1

Audio networks are finding their installed costs on new studios or remodels to be between 20% and 35% less traditional hardwired studios.

NOT JUST CONSOLES

When most broadcasters think of IP-Audio, they tend to relate it to studio replacement. And why not? With over 500 console / routing systems on the air at this writing, IP-Audio has certainly proven itself in this area.

But broadcast engineers are nothing if not innovative. They take new technology and invent ingenious new ways of using it that even its designers might not have conceived of. And sharp broadcasters have figured out that once you get audio into the Ethernet domain, you can do many different things with it, thanks to IP-Audio's amazingly easy, flexible, scalable transport system that can use CAT-5 or CAT-6 cable, fiber or Ethernet radios – or whatever comes down the pike next – to economically transport audio and associated data wherever it's needed. From the most basic broadcast plant to the most sophisticated, IP-Audio can be a great problem solver.

SLIMMER SNAKES

The most basic use for IP-Audio is for constructing audio snakes. Snakes are useful at remotes and at venues of all kinds (including in-house performance areas) because, like their reptilian namesakes, their relatively slim profiles enable them to go almost anywhere and squeeze audio into some pretty tight areas.

In today's multiple-studio installation, however, the snake is more likely to resemble an Anaconda than the garden variety. Multi-pair cables comprised of 50, 75 or 100 discrete copper audio pairs are not only bulky, they're expensive. And yet, we've got to move audio around.

IP-Audio offers a handy alternative to the traditional audio snake. Copper bundles are not so easily scalable! And when you run out of capacity, you've got to pull another. But Ethernet capacity is easily scalable; a Gigabit or Fiber Ethernet link can carry *multiple hundreds* of audio channels at a time.

In fact, one of the very first installations of IP-Audio was for exactly this application, at Clear Channel's WREO. They needed a way to move signals between adjacent buildings, and found that using IP-Audio nodes in each building, connected by a fiber link, worked perfectly — and avoided the cost of having to trench in a conduit for multi-pair cable.

Figure 1 shows a typical IP-Audio snake, similar to what WREO deployed. IP-Audio nodes placed in each building are connected to media converters, and then fiber. Each audio node has 8 inputs and 8 outputs, well under the capacity of a 100 Mbit link, which can transport up to 32 stereo signals. If more than 8 streams are needed in each direction, another audio node and an Ethernet switch are added to each end. More audio nodes can be added until link capacity is reached. Gigabit Ethernet can carry up to 250 stereo audio channels.

Of course, building connections aren't the only reason for audio snakes. Live performance venues, worship sound and other staged events are natural applications. Nor do all IP-Audio snakes have to be permanent: remote broadcast setups at sporting venues or other outdoor events can certainly benefit from IP-Audio's single-cable setup.

FLOOR AND BUILDING INTERCONNECT

An IP-Audio application that's a natural extension of the snake is the interconnection of studios and technical operation centers in multi-floors or multi-building facilities. Minnesota Public Radio's

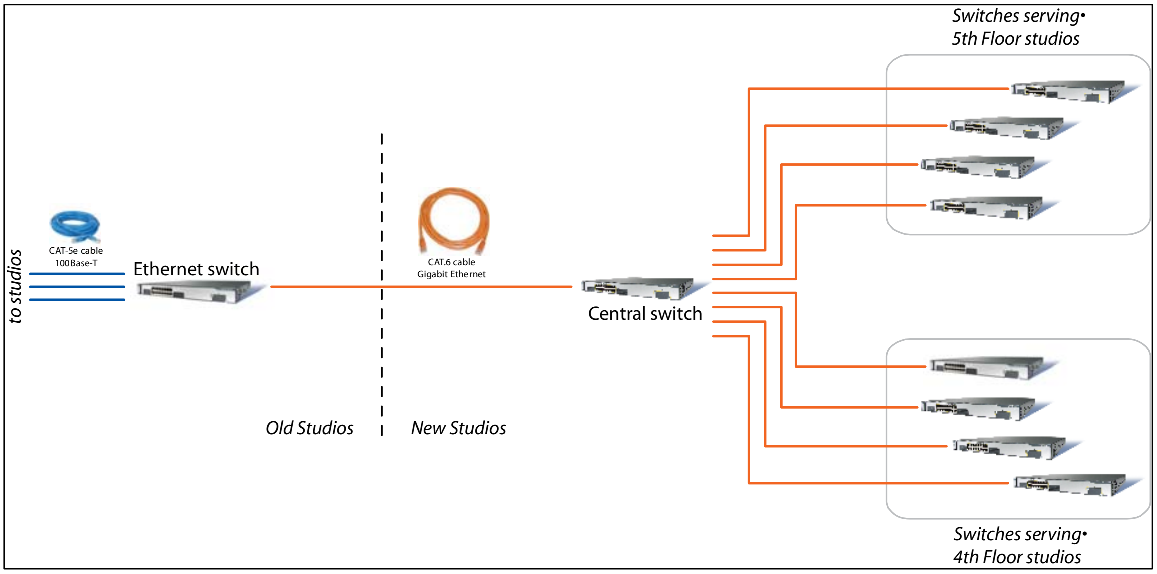

Figure 2

recent studio expansion is a perfect example: their studios span two floors of a new building, plus two floors of an adjacent building.

Users have discovered that IP-Audio is a natural for getting audio to and from transmitter shacks or where satellite or RPU receivers are located. Some have told me that fiber is an excellent connection method for this application, because it greatly reduces the potential for transmitted lightning damage — fiber doesn't conduct!

Let's look at the Minnesota Public Radio example first. MPR put a central switch in their new building, a load-balancing bladeserver from Cisco. The old studios that needed connection with the new ones are located in an adjacent building with a shared wall, so a Gigabit Ethernet link was run from the central switch through the wall to an edge switch that served the audio nodes in the old studios. These nodes use CAT-5e to connect to their edge switch, which connects to the central switch in turn with CAT-6 (for Gigabit).

The central switch also connects via Gigabit to the edge switches on both floors of the new studio complex, as shown in the (much simplified) diagram above.

Now let's look at the transmitter-to-doghouse mentioned previously as a solution for solving lightning isolation problems. Cumulus Media's Youngstown, Ohio facility is made up of on-air and production studios for 6 stations scattered across two buildings. Also on site are two transmitter buildings containing transmitters for two of those stations along with the STL and RPU gear for the rest, all attached to a 700-foot tower. The distance between the buildings is between 100 and 200 feet, and all are interconnected by copper in a triangle arrangement. Each building has a separate power feed as well. Lightning has been a recurring issue, not to mention simple surge and ground differentials.

The solution was to build a processing / data rack room and run fiber between that facility and the other studio and transmitter building, eliminating ground loop problems and helping attenuate

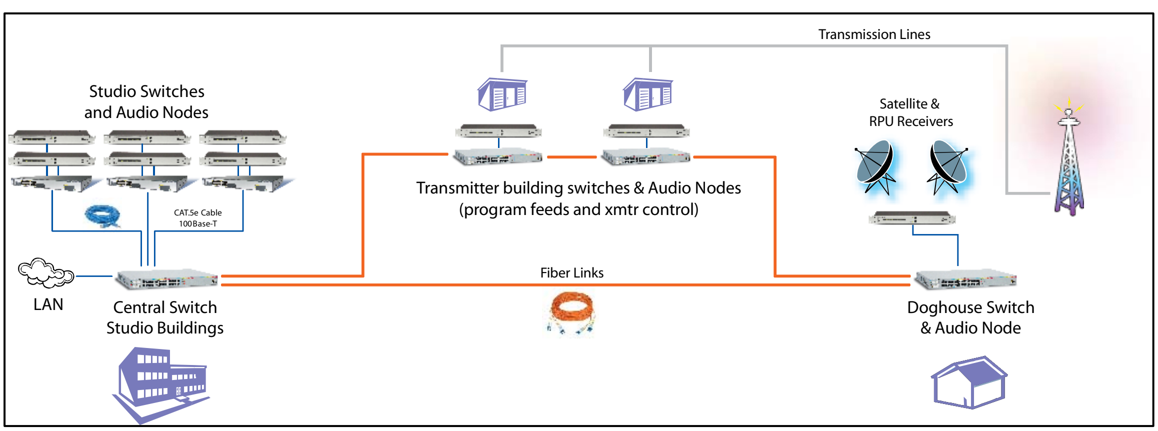

Figure 3

lightning attraction, as well as breaking the conductive link between buildings.

A combination of bidirectional AES, analog, and Ethernet has to be carried over this fiber, and Cumulus engineers found, after extensive research of fiber-only systems, that Ethernet combined with fiber provided more flexibility.

As shown in Figure 3, IP-Audio nodes and Ethernet switches equipped with GBIC (Gigabit Interface Converters) are placed in each location. Connecting all of the switches with fiber creates an IP-Audio network across all of the buildings, allowing sharing and routing of audio, device control and LAN traffic, eliminating ground loop problems and effectively limiting potential lightning damage to very localized areas.

SOUND CARD ALTERNATIVES

The phrase "paradigm shift" is overused, so I apologize in advance for using it here! Overuse aside, however, it's an accurate description for an IP-Audio application that has the potential to completely revolutionize one segment of broadcast studio architecture.

Since IP-Audio converts audio and control into routable Ethernet data, and since Ethernet was designed to facilitate computer users' exchange of PC data, it stands to reason that PC audio workstations should be able to exchange audio with the rest of the network directly, right? And they can. PC audio workstations or audio delivery systems can be equipped with an IP-Audio Driver that emulates a soundcard. The driver can then route multiple channels of audio playback, record and data/logic command streams via the PC's Network Interface Card (NIC), which connects with the local Ethernet switch.

Many IP-Audio users have discovered that they can save substantial amounts of money by using this approach. Not only is there money saved on sound card purchases, but also on associated wiring. Not only that, the cost of the router switcher port (or console input module) that a traditional hardwired router system would require for these sound card inputs is eliminated as well. Once PC audio sources are part of the Ethernet data flow, they are universally available as IP streams and can be switched or mixed as needed.

Users have been quick to realize the benefits of this approach, and many well-known delivery system providers have announced that they have IP-Audio drivers available for popular playout systems. These providers include the likes of Broadcast Electronics, BSI, D.A.V.I.D. Systems, ENCO Systems, Google/dMarc, iMediaTouch, Netia, Pristine Systems and Prophet Systems, to name a few. Media giants such as Univision, Clear Channel and France's Lagardere Group have already built facilities using this tight PC/router model.

Of course, this approach isn't mandated. IP-Audio systems coexist quite nicely with traditional sound cards (by simply plugging their inputs and outputs into Audio Nodes).

Even sound card providers themselves realize the benefits of IP-Audio compatibility. Respected audio-card maker AudioScience recently announced their intention to manufacture a broadcast-quality card that works directly with IP-Audio networks. Instead of the usual "pigtail" and tangle of I/O and logic connectors, multiple channels of audio and control data will travel from the card directly to the network over a single thin Ethernet cable.

ROUTING SWITCHERS, REMIXED

While the routing switcher is a pretty familiar sight, especially in larger markets, IP-Audio users have found some new ways to build them.

Traditional routers are usually fairly large and complicated, with topologies of 64x64 crosspoints or larger. IP-Audio users have found that, thanks to Ethernet's inherent scalability, it's not only possible but practical to assemble an audio routing switcher as small as 8x8 stereo inputs and outputs (analog and/or digital, with or without program associated data) that could scale to a very large system without huge jumps of costs or complexity.

Conventional routing systems, while serving their purpose well, nonetheless force a choice during initial planning and building: the "small frame" and the "big frame." And this choice must be made before even one cable is run. Studio projects are constantly in flux during construction, and requirements often change mid-stream. The result: if you purchase more capacity than you thought you'd need, money is wasted. If you purchase too little capacity, costs escalate rapidly when it comes time to correct that mistake: most routing switcher frames are custom-built, so if you wind up with more inputs than your router frame is capable of taking, the only solution is a second, expensive frame.

Another consideration is the cost of the multiple plug-in I/O cards that hardwired routers require. And it's not like you can re-use them: cards from one system – even from the same manufacturer – often cannot be used in a different card cage.

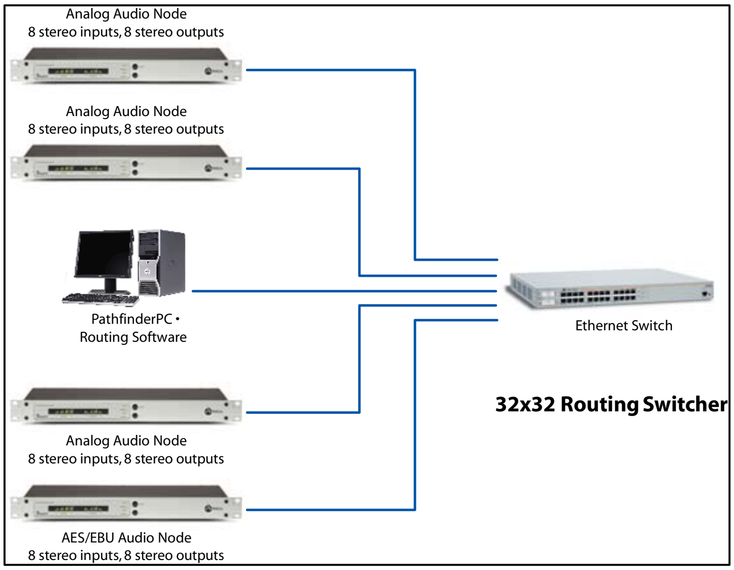

Figure 4

IP-Audio could completely change the way routing applications are constructed. Thanks to the scalability of their Ethernet backbone, small systems can be built with relatively little initial cash outlay. Then, when more capacity is required, the network can be expanded, again and again if need be, without ever discarding the original parts of the system. An IP-Audio router can thus grow from very small to very large at a very predictable and linear cost.

Figure 4 underscores this point, demonstrating how, with very little equipment, you can easily construct a 32x32 routing switcher using IP-Audio components.

The diagram shows four Audio Nodes, each of which have 8 stereo inputs and 8 stereo outputs. The nodes are connected to a 100/1000Base-T Ethernet switch using inexpensive CAT-5e cable. Only one Ethernet cable is needed to connect each node to the switch, because each 100Base-T link has enough capacity to carry 32 stereo signals — all running at once! Automated routing control and scene changes can be handled by software running on a connected PC.

32x32 is a pretty respectable routing system, easily capable of serving two or three studios, but times change and needs grow. IP-Audio users have found that it's especially easy to add capacity to their routing systems because all they have to do is connect more nodes to the existing setup, again using standard CAT-5e.

As you can see in Figure 5, the original 32x32 system is still intact — but with more Audio nodes added to accommodate increased routing demand, doubling the size of the original routing switcher system to 66x66.

Should the need for more capacity arise, the same procedure is repeated, adding more nodes where they're needed. Figure 6 shows how the original system has expanded yet again to a 132x132 router.

As it's somewhat related, this is a good place to mention another neat benefit of IP-Audio systems: portability.

With traditional router systems, it's nearly impossible to re-use hardware in a new location. A

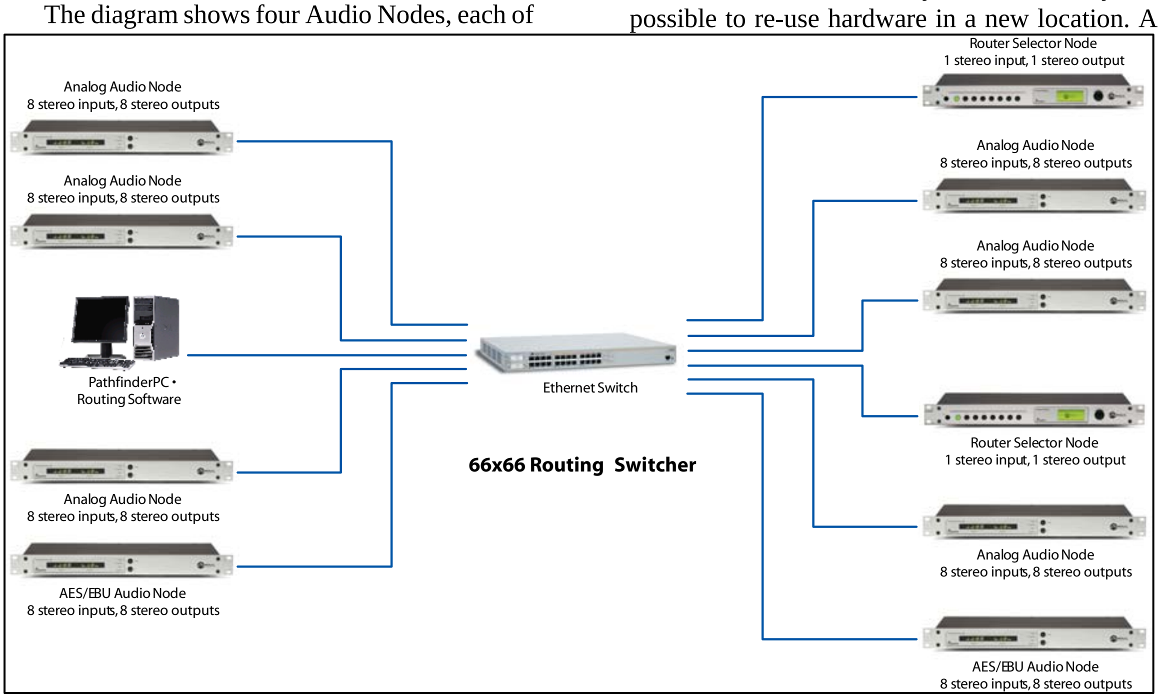

Figure 5

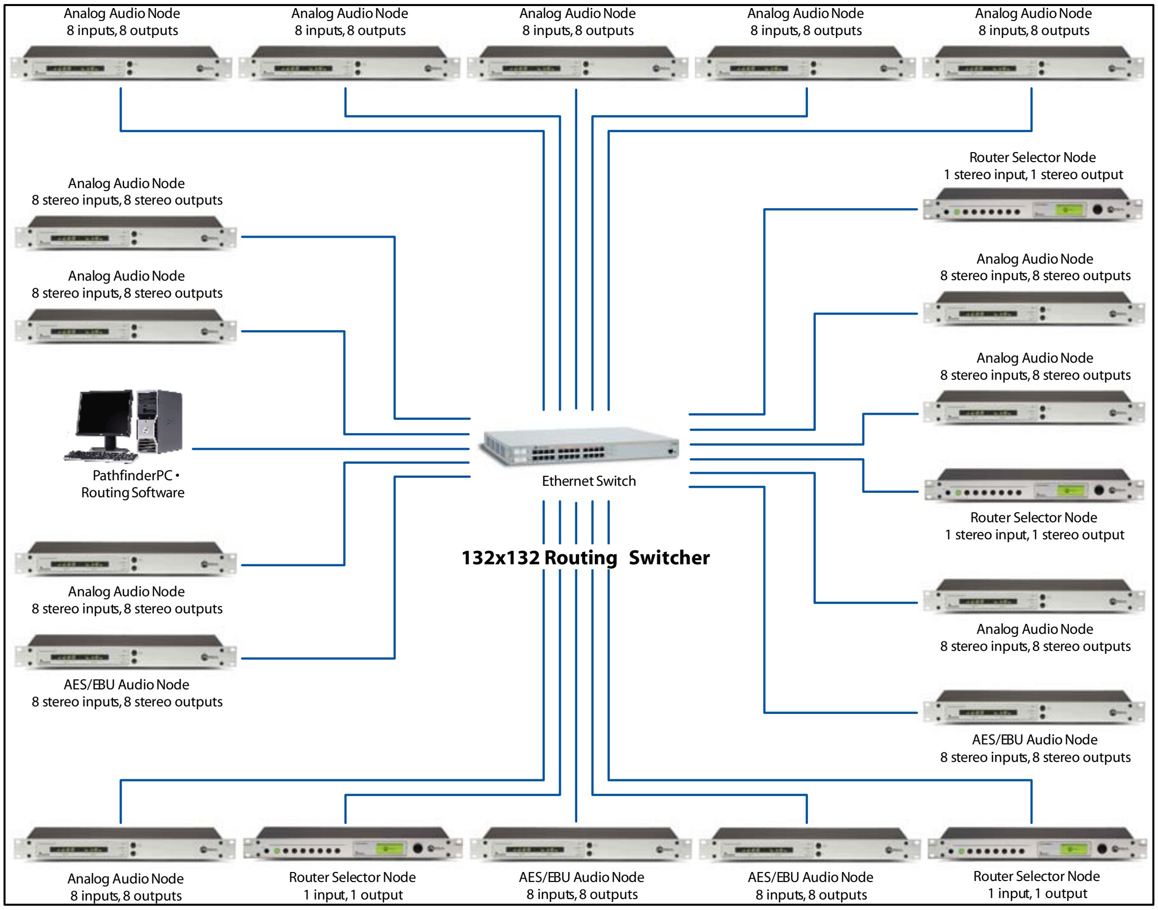

Figure 6

different building means different card cages, different locations for inputs and outputs — not to mention all the expensive multi-pair cable that is thrown away.

In sharp contrast, IP-Audio routing systems can literally be picked up and moved to a new location. Since audio nodes are rackmount devices, they can simply be racked in the new facilities and connected with Ethernet. A little bit of software reconfiguration, and the entire system is online again.

UNCOMPRESSED STL

Everybody knows that the 950 MHz STL band is terribly crowded in all but the smallest markets. Hardly a month goes by where there isn't a story about a market where somebody was knocked off the air by another station turning on an STL, or where STL bandwidth is reduced by the frequency coordinator.

An innovative use of IP-Audio is helping to solve these problems in several markets. Broadcasters are using IP-audio nodes, combined with Ethernet radios from providers such as Broadcast Electronics, Dragonwave and Orthogon to replace 950 MHz transmissions with license-free data channels in higher-frequency bands.

These combinations of IP-Audio gear and Ethernet radios can provide multiple channels of bi-directional analog or AES audio (as well as GPIO commands for remote control of transmitter rack equipment such as audio processors satellite receivers). And, unlike traditional STL, these audio channels are uncompressed, so transmission-related coding artifacts are eliminated.

(A side benefit to this transmission method is that the link can be easily reconfigured to add and subtract audio channels — an excellent fit for HD Radio applications with multiple concurrent program channels.)

An example of such an installation using Ethernet radios is found in use at the cluster of radio stations operated by Clear Channel in Birmingham, Alabama.

Chief Engineer Bob Newberry found that the portion of the local spectrum he'd used for years for STL was unsuitable for future HD Radio implementation. Also, his stations had actually been taken off the air due to interference when another station accidentally fired up transmissions on the same frequency.

Eyes to the future, Bob considered IP-Audio as a way to solve not only frequency crowding, but also to consolidate multiple audio and data channels into a single transmission path.

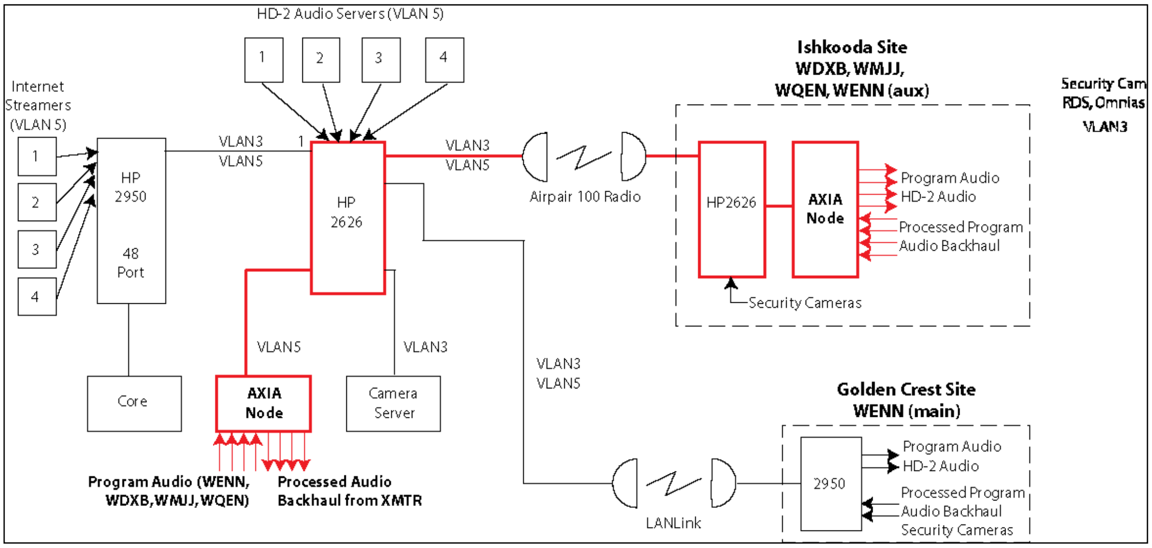

Figure 7

The Birmingham cluster was already feeding four transmitter sites with two 4-channel STL systems, and employed a LAN extender to get RDS data and transmitter remote control to the transmitter site. With three HD-2 channels planned for broadcast, the question arose: how would they get the additional audio (plus associated RDS and remote control) to the transmitter site without overloading the existing gear?

Investigation showed that an Ethernet radio with a 100 MB/s bidirectional link would have enough capacity for audio, PAD and control data, plus room for more audio/data channels should future plans so necessitate.

To implement the system, audio nodes were placed on either end of an 18 GHz link; the simplified diagram shown in Figure 7 illustrates how program content, audio backhaul and even streaming images from security cameras coexist on the same "STL" link.

At the time of this writing, the system will be celebrating its first anniversary in service; performance has been flawless.

A second example comes from Radio Skonto in Riga, Latvia. They'd received a license for transmitters in two other cities outside their main service area, and management was quite pleased. But for engineer Ivo Bankovs, the problem was one of distance: how would he get the signal from the studio in Riga to the cities of Rezekne and Liepaja, each about 250km away?

Traditional Telco audio service was very expensive. Installing his own STL radio system was impractical due to the need for intermediate repeaters and towers — if he could get a license. So he asked a local Internet Service Provider, Latnet, if it would be possible to use IP links to the two sites.

Fortunately, the ISP was able to offer a guaranteed bandwidth service to the two sites at a cost much lower than the alternatives. To avoid paying Telco costs, Latnet installed a 26GHz IP radio link with equipment made by Netro to Radio Skonto's studios. Radio Skonto contracted for 384kbps from the Riga studio and 256kbps at each of the remote sites, providing plenty of margin for packet overhead.

Radio Skonto decided to use Telos Zephyrs to provide MPEG compression and IP conversion. The station is a fully IP-Audio networked facility, so they wanted to input audio to the Zephyr at the main site via IP. This was accomplished by simply plugging it in to a spare port on the Ethernet switch and configuring the main IP-Audio program channel as its input source. A low-cost ($50) consumer IP router was used to connect the Latnet IP radio to another port on the Ethernet switch. This router was "locked-down" to pass only the required audio signals, and thus to isolate the IP-Audio network from any other traffic on the external network. The Zephyr provides two streams, one for each site.

At each remote site is a small studio set-up for local programming. The Zephyrs deliver their audio outputs via analog connections to a console fader input; DSL lines provide the IP connection. As before, these are firewalled with a low-cost router inserted between the DSL line and the Zephyr; see Figure 8 for a simplified diagram. The routers have additional LAN ports that are connected to some other PCs used for non-real-time audio transfer;

The station was naturally concerned about delay, wanting the lowest possible value so that people listening to live telephone calls would not be

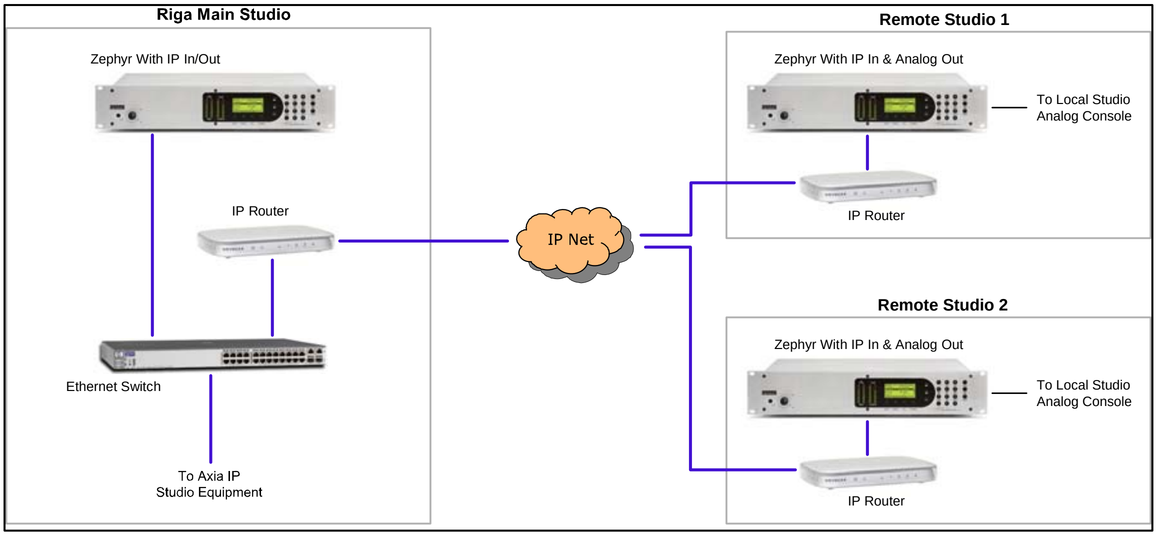

Figure 8

confused (the station runs without a profanity delay). To accomplish this, they set their Zephyrs to the lowest buffer setting: 250ms. They were not sure if a setting that low would work, but decided that the best strategy would be to start with a minimum value and increase it in small steps until any audio drop-outs stopped.

As it turned out, the minimum setting worked without problem — fortunately, the IP network had very low jitter.

Ivo and his assistant engineer, Karlis Malkavs, wondered if 128kbps would provide sufficient quality and were ready to increase to a higher rate if compression artifacts were evident. But after a month of on-air operation, they decided to stay with this lower rate. MPEG AAC has been officially designated as “indistinguishable from the source” at 128kbps by the European Broadcasting Union; this real-world application certainly proved it to be so!

The system has been in operation for a number of months and is working well.

There was an outage at one of the sites caused by a power failure affecting the ISP’s equipment center. A UPS was supposed to provide back-up, but a switch in the path was mistakenly not connected to the UPS. Other than that, Ivo reports no problems. The ISP’s promise of guaranteed bandwidth has been kept, and the IP option has proven to be a satisfactory studio-to-transmitter link.

AUDIO MONITORING FOR ALL

Listening stations are another area being transformed by IP-Audio. Until very recently, making listening stations available for GMs, PDs, Sales Directors – even the DOE! – meant filling a TOC rack with distribution amplifiers, tuners, line selectors and other gear… and then running more cable to individual offices throughout the facility, outfitting them with speakers, volume controls, et cetera.

Needless to say, this can be an expensive and time-consuming endeavor. As a result, these projects end up being very costly or providing less monitoring points than desired.

IP-Audio users have found that they can use PCs connected to their audio network to provide monitoring for practically everyone in a networked facility by using software applications that translate networked audio streams to PC audio playable on any local computer.

In this way, anyone connected to that network can monitor air – or any other available audio stream – using the speakers already attached to their computer. GMs, Program Directors, salespeople and programming staff can all listen to their choice of air monitors instantly, from any location, with no additional equipment to purchase and install or extra cable to run.

Quite a change from the old Altec corner speaker hung in the GM’s office.

FACILITY AUTOMATION MATURES

The more activities (routine and emergency) we can simplify or take out of the hands of operators, the more reliable our plants are likely to be. IP-Audio, thanks to its computer-tech base and subsequent natural capability for integrating and interacting with PCs, is working changes in this area as well.

Certainly, we’ve had some ability to automate aspects of the broadcast environment before: audio processors with dayparting capabilities are a good example; PC-based audio delivery systems are now so ubiquitous that the service they provide (with smoothness and accuracy unthought-of in the Schaeffer days) is routinely taken for granted.

But now, with computers tied even more closely to hardware, software applications can take control of nearly every audio device throughout the plant. They can administer pre-scheduled routing scene changes, react in predetermined ways to trigger events, even take emergency action in the absence of station personnel.

PathfinderPC is such an application. It can work with several different kinds of routing switchers, and is optimized for control of IP-Audio networks. Pathfinder contains powerful scripting capabilities that use simple and familiar Boolean logic to enable users to construct logic trees, building applications that react either to direct user interaction such as a button press or contact closure activation, or to pre-defined fault states such as dead air, loss of program feed or diminished audio levels. Scheduled routing changes can also be programmed.

Of all the ways in the world to be awakened, a phone call at 3AM is one of the least enjoyable. Pathfinder can help take the urgency out of unexpected events by sensing and reacting to loss of program audio.

Let's say that (at 3AM, of course) a satellite channel supplying programming suddenly goes dark. Pathfinder's silence-sense function can be programmed ahead of time to deal with such a problem; when audio drops, it takes an alternate air feed. There's e-mail notification capability too – conditional routing scenes can be defined so that if, for instance, a primary ISDN feed gets knocked out, the system switches to backup and sends an e-mail to your Blackberry® telling you what happened.

Another aspect of PathfinderPC is its support for touch screen monitors. On-screen "virtual button panels" can be constructed to run on local PCs so that board ops can execute simple (changing satellite receivers) or complex (transferring program feed origination to another studio) switching functions with just one tap.

An informal poll reveals that roughly 90% of broadcasters who adopt IP-Audio studio architecture opt to include this sort of advanced automated control; such systems are in use at stations belonging to Univision, Clear Channel, Cumulus Media, MPR and many others.

INTERESTING CONSOLE APPLICATIONS

There are some unique benefits associated with IP-Audio mixing consoles that can help simplify life in the studio. First, a console in an IP-Audio system makes the distribution of I/O, logic and backfeeds easier than ever.

Sharing a standard audio connection in a traditional studio means two pairs for the stereo audio, another two pairs (at least) for machine control logic, and another two pairs for a backfeed or mix-minus. That's at least 6 pairs of wire for each audio connection, multiplied by the number of places you need to send it.

By contrast, an IP-Audio studio setup reduces the amount of wiring needed by an order of magnitude, because audio I/O, machine logic and even mix-minuses are all converted to digital streams, which are routed together in a single "bundle" of packets using Ethernet cable which can be shared with hundreds of other like signals. But elimination of wiring isn't the best part — there's also the fact that since audio, logic and backfeed are now part of the same digital package, you will *always receive the correct mix-minus* when routing remote or telephone audio. In other words, you can bring up a codec or hybrid on any console and it always works right.

Speaking of phones, IP-Audio also provides much tighter phone/console interoperability than has ever been possible before. With traditional consoles, phone operations have required outboard controllers – switch panels or phone-like devices – to control the hybrid. This interrupts the workflow in the control room, since the jock has to take his hands and eyes off the console to work another device. Drop-in control panels for consoles mitigate this somewhat, but these controls are often not adjacent to the audio faders themselves, and in any case require discrete wiring for audio, hybrid control and mix-minuses.

As noted previously, IP-Audio networks route I/O, control and backfeeds together, so the phone controller can live in the console, right next to the audio faders, and the console can receive audio, generate hybrid control logic and send mix-minus, all over the same Ethernet cable. Another benefit is that the hybrid itself doesn't need to be located in the studio anymore — it can be placed in the TOC or some other central rack room, safe from curious hands.

Finally, something I consider to be a very unique console application: the ability for two consoles to work simultaneously with shared sources.

Imagine a scenario in which talent in the talk studio wants their own level control of some audio devices. But the control room operator needs to keep control of the levels in order to "assist" the talent should they run their audio at the wrong level.

The solution: link motorized faders on the control room and talent consoles, designating the CR as the "master" console, and the studio as the "slave." This allows talent to set levels as desired, but also allow the board op to override those levels as needed – something that can't be done with a traditional console setup.

CONCLUSION

Historically, IP-based systems have worked a sea change in every industry into which they have been introduced.

The World Wide Web would not exist if not for IP. Defense computers connected with serial data transmissions formed rudimentary networks in the 1960s, and those networks expanded with ARPANET in the 1970s. But the development and implementation of TCP/IP on ARPANET in 1983 led directly to the growth of what is now the Internet.

Banking networks using ATM (Asynchronous Transfer Mode) revolutionized that industry in the 1970s by allowing computers to take on the tedious accountancy handwork that was the foundation of banking. But the implementation of TCP/IP over ATM made possible instant money transfers between far-flung locations. Remember the days of hurrying to the bank on Fridays (between 10 and 4, of course) to get cash for the weekend? Now, instant banking and automated tellers are available 24/7 thanks to IP.

Telephone systems have undergone tremendous change thanks to IP. Traditional PBX systems and Centrex lines considered advanced only a few short years ago are now being abandoned in favor of VoIP (Voice over Internet Protocol) systems that are more flexible and more cost effective than the old services could ever be. The migration to VoIP has been so universal that in early 2006, Cisco announced that it had sold its 7.5 millionth IP telephone. And the adoption by consumers of mass-market VoIP services over broadband Internet access which began in 2004 continues unabated. According to research firm Cahners In-Stat, more than 9 million US households now have at least one active VoIP user; it was estimated at the end of Q4 2006 that nearly 8% of U.S. households now use a VoIP telephone service —up from 6.5% just six months prior.[1]

Finally, the television industry has embraced Video-over-IP with a fervor. Gone are the days of sending dubbed U-Matic cartridges to other stations by courier, or even satellite uplinks: TV production facilities now share content via video gateways that link production facilities and television stations with each other directly via IP connection. Writing in Broadcast Engineering, Brad Gilmer asserts that "Video over IP is destined to become the predominant technology for the transport of professional video over WANs."[2] And, as with IP telephony, this service is spilling over into the consumer arena with IPTV reaching millions of homes via digital cable services.

With this data in mind, it's certainly no stretch to predict that IP-Audio is on a course to revolutionize the broadcasting industry. More and more applications for this technology are being discovered as the numbers of IP-Audio installations rise. The question for radio broadcasters is not whether to transition to IP-Audio, but when.

[1] "Report Sees Rise in Residential VoIP Service", Cabling Installation & Maintenance Online, December 22, 2006 (http://tinyurl.com/3cmvcw)

[2] "Video Over IP", Broadcast Engineering, August 1, 2006 (http://tinyurl.com/23mwky)