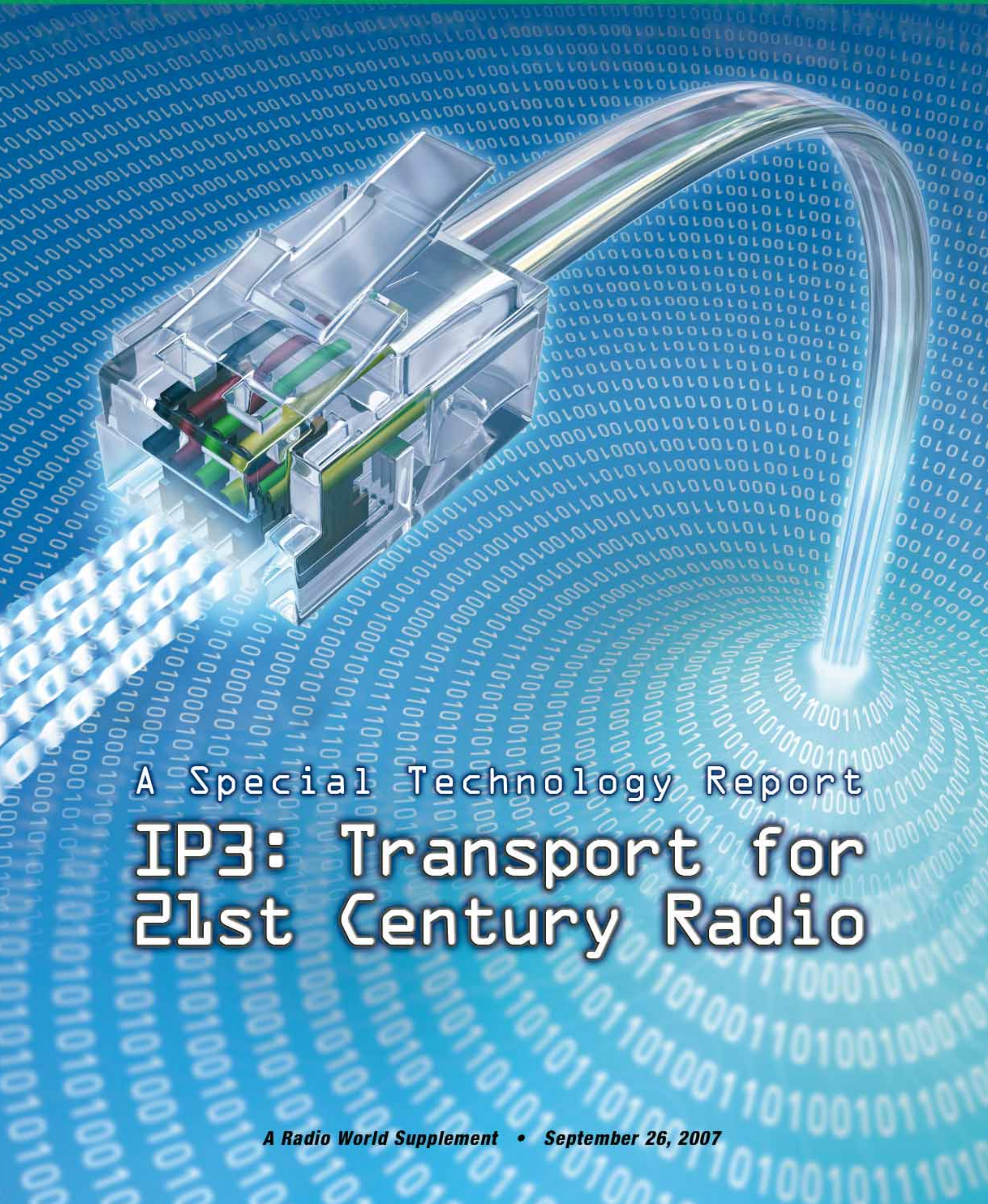
Real-World Tools & Tips for Broadcasters



A Special Technology Report

IP3: Transport for 21st Century Radio

A Radio World Supplement • September 26, 2007

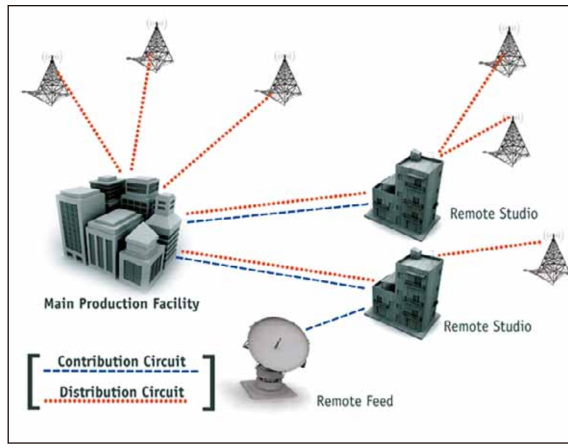
IP Audio Transport Permeates The Radio Environment

The Technology Continues Its March Into Mainstream Broadcast Applications

By James G. Withers

The author is chief technology officer for Koplar Communications in St. Louis.

Not so many years ago, if you needed an audio feed routed into a studio, or worse, across a field to the STL tower out back, you got out the shielded audio cable, a dirty pair of jeans and started pulling wire. Of course, that was only the first step. Ground loops, matching networks, level changes, terminations, bridges … all added to the general angst of moving audio from place to place in the typical radio station. Even the most meticulous attention paid to "Good Engineering Practices" was not always enough to prevent audio degradation. This all changed — and still is changing — with the advent and general implementation of IP audio transport.



Simplified depiction of a radio broadcast network.

HELLO IP AUDIO!

IP audio transport is a general term used to describe the ingesting, routing, transportation and delivery of digitally-based audio streams, using the TCP/IP and RTP/IP data transmission standards, over the Internet or packet-switched networks. TCP stands for Transmission Control Protocol, while RTP stands for Real-Time Protocol. IP is the generic Internet Protocol standard and applies to both subsets. IP-based Ethernet networks were created for nonreal-time data transfer to and from computer terminals and peripheral devices. Latency, or the delay exhibited as data packets were created, labeled, forward error corrected, transferred, and finally, reassembled at the receiving end of a network, was a minor consideration, since it was (and still is) usually measured in milliseconds. However, even a few milliseconds' delay in an audio packet will interrupt an audio playback and cannot be tolerated in a live broadcast environment. Therefore, before TCP/IP could be used to reliably transport live audio, careful consideration had to be paid to packet sizes, so-called "jitter" compensation, buffering calculations and priority assignments.

Initially, TCP/IP, as well as Ethernet speeds, were simply too slow and fragile for the absolute reliability required, but with the advent of RTP/IP (Real-Time Protocol), and as Ethernet bandwidths have gotten greater and data buffering schemes have improved, IP-based audio is gaining favor among broadcasters in all size markets.

As with most new technologies, initial steps into IP-based audio have been baby ones; relegated to the periphery, rather than at the center of a station's operation. Now, though, stations are increasingly using IP audio as the core technology to move audio around the station and beyond: to and from remote studios, transmitter sites and throughout regional origination and distribution centers. Additionally, the point-to-multipoint capabilities of IP systems are making IP-delivered audio a reasonable alternative to dedicated occasional-use satellite feeds in many cases.

Johannes Rietschel CEO of Barix Technology, a Swiss-based IP equipment systems provider says, "IP technology is reliable, in wide use, and for many applications available at a much lower cost basis than traditional systems." As an example, he elaborates, "One of our clients, who operates radio stations in Guam, feeds them from the U.S. using Barix terminal equipment and the stan-

APT Audio Codecs

Designed to deliver optimum audio performance and reliability over IP networks, APT's audio codecs are the professional broadcaster's choice for STLs and studio networking.

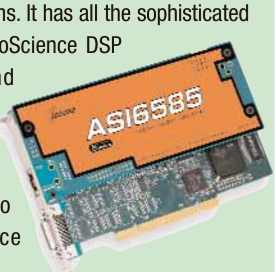
Ranging from the entry-level WorldCast Horizon offering Enhanced apt-X coding over IP Links to the richly featured, multi-algorithm WorldCast Eclipse offering IP, X.21/V.35 and ISDN transport options, the WorldCast line offers a wide variety of options to fit the needs of a wide variety of broadcast applications.

For those requiring multiple channels of audio, the WorldNet Oslo Audio Multiplexer offers a flexible, modular solution capable of delivering up to 28 channels over either IP or a T1 line. Features such as DSP-based architecture, automatic back-up to synchronous lines, silence detection, alarms and contact closures, embedded auxiliary data and a highly sophisticated GUI ensure that APT codecs are truly professional solutions.

Contact APT at (781) 810-2260 or visit www.aptx.com.

AudioScience AS16585

The ASI6585 Livewire sound card from AudioScience brings a new level of flexibility to IP Audio routing systems. It has all the sophisticated features of an AudioScience DSP accelerated sound card combined with the ability to connect directly to a Livewire network to dramatically reduce system costs.



Using a powerful Texas Instruments onboard floating point DSP with Axia Livewire networked audio protocol allows the ASI6585 to simultaneously play up to eight stereo streams of audio which can be mixed to eight stereo outputs, and record up to eight audio streams fed from eight stereo inputs, over switched Ethernet. The feature set of the ASI6585 extends to MRX multi-rate mixing, MPEG Layer 2 and 3 encoding and decoding, TSX time scaling and SSX2 multichannel record and playback.

Contact AudioScience at (302) 324-5333 or visit www.audioscience.com.

Product information is provided by suppliers dard public IP infrastructure. How else could you possibly do this at a reasonable cost in quasi-real time?”

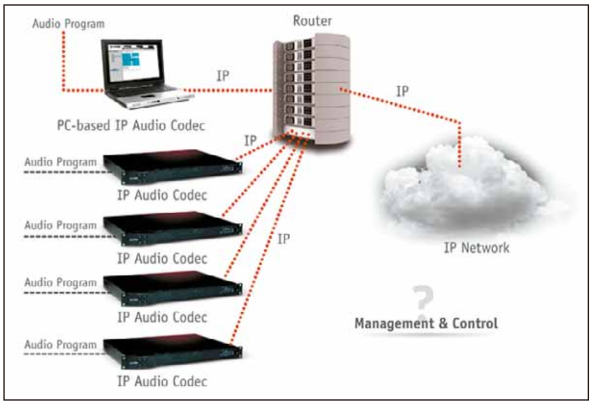
Ireland-based APT is a company built entirely around the idea of using IP transport as an audio routing technology. Jonny McClintock, commercial director of APT, says the idea has almost universal appeal among broadcasters.

“Every broadcaster, service provider and system integrator we deal with has either put in an IP network for networking between locations for distribution and contribution networks or are considering putting in a network.” The reasons, he says, have as much to do with improved processing, as with cost.

“The first generation of IP codecs was based on a PC architecture using an ‘off-the-shelf’ OS [operating system]. The hardware was consumer quality. The OS could cause instability and the system simply was not designed for 24/7/365 mission critical applications.” He goes on to explain that APT recognized those issues and modified its offerings accordingly. “We use application-specific DSP architecture, based around professional-grade electronics. This supplies the long-term reliability that broadcasters require. The ‘PC-in-a-box’ was fine to prove a concept, but DSP is a fundamental requirement.”

Axia Audio of Cleveland offers a broad suite of IP solutions. Long a player in the IP audio-for-broadcasters field, Axia markets their proprietary “Livewire” transport standard. According to its specifications, Livewire is a total networking system of hardware and software that can take a station from traditional analog audio routing to IP audio in incremental steps, or all at once. Using RTP/IP protocols, Livewire assigns packets priorities to all incoming data; small, high-priority packets for live audio are assigned the “Livestream” designation, while lower priority, larger packets—for offline audio file transfer, for example—are assigned the “Standardstream” designation. Since all Axia routers and source/destination equipment accept and can switch seamlessly between both standards, delays in routing mission-critical, live audio are minimized or even eliminated. Using standard Ethernet network hardware simplifies things even more. IP audio easily coexists with regular Ethernet data, so in many cases, stations can simply tap into their existing Ethernet networks and begin the conversion process immediately.

One company jumping onto the Livewire bandwagon is AudioScience, the sound card/computer interface maker. Stephen Turner, vice president and co-founder, says: “AudioScience has, for many years, been interested in supplying an audio over IP solution to our automation customers in radio. However, we had been waiting for a standard to emerge. In the radio space at least, that standard now seems to be Livewire. We also had numerous requests from our customers to supply a Livewire solution that was compatible with their automation applications and contained all the functionality that our regular sound cards provide.”



Traditional approach to transporting audio over IP networks.

And IP is not at all limited to “wired” applications. Chris Crump, director of sales at Comrex, points out: “Major advances in wireless IP technology have significantly changed the options broadcasters have for establishing point-to-point audio links between two or more locations. In areas where traditional ‘wired line’ telco or IP services are not available due to phone company limitations, terrain or budget, services such as satellite, 3G Wireless and unlicensed 5.8 GHz IP products all provide robust connectivity solutions for both remote broadcast as well as STL applications. In fact, we are seeing increasing numbers of broadcasters that are deploying IP-based STL solutions using VSAT satellite technology and 5.8 GHz radios for transmitter sites in relatively inaccessible locations. In some cases, it’s the only practical solution for their specific application.”

MISSION-CRITICAL RELIABILITY

Rietschel of Barix Technology adds, “The whole world relies on IP in many applications,

Axia Zephyr iPort

Broadcasters in different locations can now easily and transparently connect and share audio, thanks to the Zephyr iPort MPEG Gateway, a new multichannel IP codec from Telos and Axia. Facilities with Axia IP-Audio networks can use the Zephyr iPort to send and receive eight audio channels plus program data and machine logic over T1 or T3 lines, or other types of QoS network services. Broadcasters without IP Audio networks can use Zephyr iPort too, with the simple addition of an Axia Audio Node.

Contact Axia at (216) 241-7225 or visit www.AxiaAudio.com.

Barix Exstreamer 1000

Barix introduces the Exstreamer 1000, the first Barix device to combine the company’s low-cost Instreamer IP encoding and Exstreamer IP decoding technology into a single, one-half 19-inch rack-mountable device.

Barix audio over IP technology is often used in STL, RPU and Internet radio applications. The Exstreamer 1000 adds several professional features to the existing technology, including balanced inputs and outputs, a high quality A/D-D/A signal converter to reduce noise and improve audio quality, eight relay contact closures to trigger local announcements, and a professional AES/EBU interface to capture a digital signal at the source and maintain it throughout the chain.

Contact Barix at (866) 815-0866 or visit www.barix.com.

secure and mission-critical applications included. There is no problem relying on IP audio transfer and control." He says IP systems often have higher reliability and better quality than traditional technologies. But, he cautions, "Don't purchase the cheapest connectivity — service and quality has its price."

When considering making changes or additions to a system as critical as an in-house network, though, most of the vendors and engineering managers contacted for this article recommend having a competent IT manager on staff. Phil Owens of Wheatstone Corporation agrees, saying, "We do advise our customers that want to interface our systems to IP networks to use a staff or contracted IT professional. Programming the network switch to partition audio from the regular business network data is not complicated, but if it is not done correctly, the first thing to go will be the audio." He adds, "Stations also need to be careful with priority assignments, so the GM's printer queue doesn't cause the air feed to hiccup!"

Harris/Intraplex is another vendor hoping to satisfy customer demand for IP equipment and systems. Harris first introduced Intraplex NetXpress in late 2005 as a managed platform for the transport of audio over IP. The platform can accommodate multiple services, including audio, data and PBX telephone communications, over a single IP connection. With the company's experience in providing solutions using dedicated T-1/E-1 paths, Bob Band, business development manager, Intraplex Products at Harris, says it was a natural next step to venture into IP-based audio transfer.

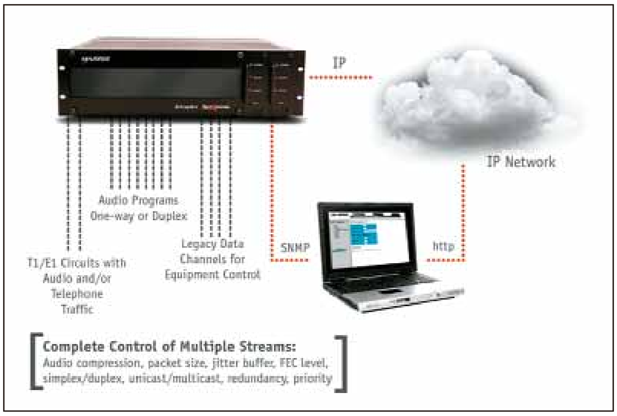
"NetXpress is our next-generation IP multiplexer. Stations ingest audio from any in-house or external IP network and NetXpress takes it from there, dynamically controlling packet size and jitter resequencing to ensure interruption-free audio." Jitter, Band said, is the term used to describe packets that arrive so far out of sequence that they cannot be successfully reinserted into the stream in the correct order. The result is lost audio packets; a disaster for live broadcast audio streams.

Harris is now on its second generation of IP-based audio equipment and this year, NetXpress is being expanded to include transmitter surveillance services, in which on-demand security video can be return-fed from a remote transmitter site to a monitoring point. That is exactly the kind of data-carrying flexibility that makes IP transport so attractive to broadcasters, according to Band. "Its ability to transport multiple audio programs in multiple formats — including HD Radio, along with data, and PBX telephone communications — over a single IP connection offers a more cost- and bandwidth-efficient transport network than separate wired T-1 circuits, microwave or fiber T-1/E-1 circuits."

With regard to cautionary notes, Band refers to a white paper he wrote that was published in Radio World Engineering Extra ("A Packetized Look at Audio Over IP," February 22, 2006) discussing the ins and outs of audio packets and IP audio. Implementing a successful IP audio network, he writes, gets down to a few basic issues: packet size, jitter buffering and resequencing, FEC, or Forward Error Correction, and correct packet labeling for priority assignments. If the system is designed to address those potential pitfalls, it will deliver pristine audio with complete reliability.

SIZE DOES NOT MATTER

Broadcasters large and small have adopted the technology. Sterling Davis, vice president of engineering for Cox Broadcasting, is a big fan.



Managed platform approach to transporting audio over IP.

"We are rolling it out in our stations that need a full makeover, particularly when routers and consoles are being replaced," he says. In addition to the improved quality of IP audio, Davis likes the flexibility of using IP audio as an intra-plant distribution method. "It's also about the workflow benefits and rewiring ease," he says, "It makes equipment configurations and reconfigurations easy and the audio is available anywhere we might eventually need it. Flexibility really is the key for us." He adds that Cox has begun using Motorola Orthogon spread-spectrum IP transport STLs, particularly in markets where traditional STL frequencies are congested. "The Orthogon radios do not require a license, and being IP-based, they are the final link in keeping our audio in digital form all the way through the process, from origination to final transmission."

The benefits of IP extend to smaller markets. Bill Doerner, operations manager at KSIX Radio in Corpus Christi, Texas, says his station uses IP transfer all the time. "We get a lot of production done at the Radio Lounge in Houston, and everything we get from them comes in as an MP3 to our FTP site." From there, he says, the spots are ingested into the station's automation system, which also lives on the network, and played back to air.

KSIX has other "islands" of IP audio, as well, Doerner explains. "We originate sporting events from all over the state using the IP feature on our Tieline Commander Codecs," he says. "We used to use the POTS connection ... but finally decided to give the IP feature a try, because our long-distance landline charges were getting out of hand." Doerner approves. "Once we changed, we just never went back. They've been rock-solid." As for "gotchas," Doerner says he cannot think of any.

"There is nothing that is different from the way we would set up a remote using a phone coupler or the codecs in POTS mode. We obviously depend on our DSL connection back at the studio, but if a station has a reasonably wide pipe at that end, the remote side is a piece of cake."

For a station considering converting to IP-based audio, the benefits are readily apparent: IP audio transport is easy to implement, maintains absolute audio quality, even over long distances and multiple generations, allows unparalleled routing and control flexibility, and saves money in the process. However, certain cautions must be understood and dealt with. For example, Owens of Wheatstone says, "Mixing AES audio with nonaudio-related data packets on an Ethernet network places special burdens on the network, since unlike regular data, audio streams cannot tolerate any latency whatsoever." He adds that Wheatstone has taken a slightly different approach to IP audio transport to address those issues. "Although we use an IP-type 100baseT infrastructure, the transport is actually done via nonpacketized TDM audio stream between our router cages. That way, we do not have to use the data headers associated with packetized audio and our interconnections display very low latency."

However, he says, "Wheatstone does offer audio streaming in RTP format over Ethernet. This is mainly being used by engineers looking for an easy way to move audio into and out of their automation server PCs." In that case, he notes, "the only connection needed to the PC is a Cat5 cable running to a switch which in turn is connected to our system. That configuration delivers up to 32 RTP streams [16 in/16 out]."

Cat5 cable is the pipeline of preference at Radio Systems. Dan Braverman, president, says their customers are intrigued by IP audio because of its efficiency and flexibility. "IP audio delivery is very attractive to stations already invested in business data systems — which is basically all of them," he says. "It's extremely efficient in terms of equipment costs, and the learning curve is very quick." Braverman really doesn't see a downside, saying, "A station can take one wire — a Cat5 cable — route it through a system like our StudioHub and mix and match analog, digital and IP over the same infrastructure."

Alan Maltagliati, IT manager at Koplar Communications in St. Louis has installed networks at both radio and TV stations across the Midwest. He says station managers need to be aware of a few fundamentals about computer networks in general. "First," he says, "an assessment should be made about the amount of traffic the network will be handling. A live-to-air stereo audio feed is 192 kilobits/second, which cannot be interrupted, so a traffic analysis is a 'must.'" Maltagliati adds, though, that bottlenecks are quickly becoming "yesterday's" problem. "Gigabit Ethernet is rapidly becoming the new standard, and the telcos are rolling out MPLS (Multi-Protocol Label Switching), which allows IP networks to accept and sort any format data packet more efficiently, so network bottlenecks will just go away."

IP-based audio is not yet universal and implementing it properly does require knowledge of a whole new set of "Good Engineering Practices," but there is no question it has become another tool in the station engineer's kit for solving complex audio issues in an elegant and cost-effective manner. ■

How One Cluster Went About It

Profile of a Project: Univision Radio Moves Its Houston Stations Into the IP Future

by Marty Scruggs

The author is chief engineer of Univision Radio in Houston.

Several years ago we began plans to move radio and TV from separate leased facilities and combine them into one company-owned facility. As we researched location and design of the building, we started looking at what our technical plant should be.

With HD Radio coming it was important to implement a platform that would help us move forward and also allow us to grow and expand our capabilities while seamlessly integrating new equipment.

At the time we had six FMs and two AMs. Although most of the formats are music-intensive, we have special requirements in that one is a news/talk and one does live mixing. We also do commercial production and have network shows come to town so an additional requirement was that we needed the ability to reconfigure or move studios quickly.

Home sweet home.

DREAM

Every engineer hopes that sometime during his career he gets to build his dream facility. I have always been one to adapt to new technology; here was the opportunity to be leading the pack implementing it.

I brought together my engineering team for discussion.

Being able to move signals around without having to rewire was something we wanted so we decided our plant would have a router-based audio system. This would allow us more flexibility; we would also be able to share sources and eliminate DAs.

We had discussions with four vendors and even had them come demonstrate their systems to the engineering and air staffs. We sought to involve the folks who will be working on and maintaining this equipment; if a system overwhelms the staff, the technology is useless.

After careful consideration we decided that Axia would best suit our needs. Although several of us were familiar with IP schemes, having worked with computers, putting together an audio network on IP was new territory. But we could see the direction of the industry.

PLANNING

With 10 air studios and 11 production rooms, this would be the largest project I had ever managed; and with a budget of more than $1 million it was the most expensive. I knew that planning was important, and I was fortunate that Mark Stennett, Univision Radio's vice president of special projects, was able to come in with advice and help. He showed me how to take software programs I already knew and adapt them.

One of his tips helped us figure out how much floor and rack space we were going to need: We set up rack templates in Microsoft Excel.

Using a line and a column in the spreadsheet to represent individual racks, we began with populating the spreadsheet racks with the existing equipment we planned to use. Knowing that heat would cause problems for new computer-based systems, we added additional rack spaces for cooling.

We also looked at areas we wished to upgrade or improve. We were able to fine-tune the estimates we had given to the architects and finalize floor plans via our spreadsheets.

Meanwhile we were making decisions as to equipment.

We had worked with Houston area-based Giesler Broadcasting Supply (GBS) during upgrades and changes to our STL systems, and we decided to use them to help us build

Comrex ACCESS

Comrex ACCESS Stereo BRIC/IP/POTS codec is a broadcaster's dream, especially for those who have long dreamed of complete flexibility and ultimate mobility for remote broadcasts without having to lug around unwieldy racks of gear or deal with clumsy setups too difficult to configure in the field. ACCESS PORTABLE delivers in a sleek, compact, Lithium Ion battery powered handheld unit capable of sending mono, stereo or dual mono audio over POTS, DSL, cable, Wi-Fi, 3G cellular and satellite.

ACCESS Rackmount is the perfect studio compliment with stereo analog and digital inputs and outputs as well as a very easy-to-use Web browser interface for clear and simple control of ACCESS connections and settings. Your remotes will never be the same.

Contact Comrex at (978) 784-1776 or visit www.comrex.com.

Harris Intraplex NetXpress

The Intraplex® NetXpress™ IP multiplexer from Harris Corporation takes IP audio transport to a new level of performance and reliability. As the industry's most advanced platform for professional audio over IP, NetXpress offers system level resiliency, sophisticated network monitoring, excellent bandwidth management and up to 32 simultaneous streams. In addition to real-time audio, NetXpress supports the transport of voice, data and surveillance video, in applications such as Studio-to-Studio, STL/TSL links, remote pickup, program and spot delivery, remote site confidence monitoring and emergency backup of program feeds.

Contact Harris at (513) 459-3400 or visit www.netxpress.harris.com.

Product information is provided by suppliers

new transmission paths. Dan and Tim Giesler provided new Harris Intraplex STL HD systems that were needed to facilitate the transition. GBS had them configured so that all we needed to do was to plug them into the T-1s, power and connect them to the other pieces of equipment.

We also had GBS provide us new STL dishes and coax to reestablish our shots. We wanted to upgrade our STLs to extend the IP network of the Axia system but we were not able to satisfy ourselves that the technology was robust enough to work in Houston's radio wave-saturated atmosphere. We also had budget restraints to consider.

The studio buildout crew confer: AD Rigmaiden, Fred Morton, Sandy Johnson, Bill Hartman, Jim Hibbard and Orlando Valdivia.

We were, however, able to upgrade several STL systems to the Moseley SL9003Q. We took advantage of the fact that we could send two stereo signals across the new STLs, which allowed redundancy as we feed multiple signals to most of our transmitter sites.

For our Axia system we used Broadcasters General Store. Gary Tibbot, working with Kirk Harnack of Axia, helped us put together the pieces; they made sure we had the right number of analog and digital nodes.

Every engineer hopes that sometime during his career he gets to build his dream facility. I could not pass this one up.

The rest of the studio equipment was acquired through SCMS, where we worked with Tyler Callis and Mary Schnelle. Getting the equipment at a price that would fit into our budget turned into a major challenge. We were not willing to settle for lower quality or sacrifice programming requirements just to get a better price. That is where SCMS shone. They also arranged demos for unfamiliar equipment.

Morton at work.

CAPACITY

We wanted the ability to move a station to a different studio without major reconfigurations; and this was one of the major reasons we chose Axia.

We had seven on-air studios at the old facility. We decided we needed more to accommodate the network and talk shows we originated, and decided to build 10 on-air studios. This would also give us the ability to move a station from one studio to another in case of a major failure or problem. This has ensured almost no down time due to power supply or console failure. And yes, we have had some of each, but the Axia support crew was there to help by phone and in person.

Another challenge was having enough production time during business hours. We built nine full-fledged production rooms and made the production manager's offices functional too. This effectively gave us a total of 11 production rooms, one of which doubles as a mix studio for our hip-hop station.

MIXING IT UP

Because of the size of our installation, it soon became apparent that we needed to keep the Axia audio network separate from the building's other data networks. We would also need a separate network for the Telos Series 2101 phone system, and we would need to connect the two with a virtual local area network (VLAN).

This had to be done through a VLAN because we had to limit IP traffic to certain ports to keep the Axia system (which multicasts the IP information) from overloading the 2101 network.

This involved deciding upon an IP scheme, setting up routers and various VLANs, or virtual local area networks. We relied upon Mark's expertise here. We pulled up information on the Internet about setting up a network and how to allocate addresses and set up subnets. At times I felt like I was way over my head. Fortunately one of my engineers, AD Rigmaiden, was more familiar with IP networks and able to grasp the concept fairly easily. We also found a Web site that was useful in setting up our networks, *www.subnetmask.info*.

We developed our own separate network to handle the Axia. This involved deciding upon an IP scheme, setting up routers and various VLANs, or virtual local area networks.

Once the network configurations were determined, setup of the Axia system went quickly. Every connection to the Axia system is made using a "node." There are several versions of these nodes: analog, AES, microphone preamplifier or GPIO.

The standard analog or AES nodes come with eight inputs and eight outputs. This allows you to connect audio sources and destinations in one location that can be shared across the network. There are also router nodes that have one analog and one AES input and output. These have eight user-programmable buttons that can be changed easily according to your needs.

The router nodes display a menu of everything available on the network; you can scroll through and select the source you want to feed to the device connected to the output of the node. We use these in production and for our audio codecs. This lets us share codecs between rooms and route the correct audio down the line. Everything is programmed via a built-in Web page within each node, mix engine, control surface and IP switch or router.

A mix engine is a single-purpose computer that runs proprietary software allowing you to mix different sources of audio that come to it over a Cat5 cable in an IP stream. Sounds complicated? If I had to build it, I am sure it would be; but Axia makes it easy; all you have to do is plug in a few cables connecting it to the network and power, and give it an IP address.

TEAM

You might have noticed that I mentioned Cat5. I am not using that to run audio over in the traditional sense. Since everything on the Axia system uses IP packets, you can connect everything using Cat5 and Cat6 cables. This makes wiring much easier. By using Radio Systems StudioHub adapters we were able to build the systems using readily-available store-bought cable, plugging it into the adapters and right into the node. Most connections to the audio nodes are made using a standard RJ-45 connector.

This does not mean we did not have to run any regular audio cables; but it sure cut down the amount.

For example, the audio from our BE AudioVault comes out of the servers. We purchased a StudioHub breakout box that allowed us to use a short prebuilt Cat5 jumper to go from the breakout box into the hub. No punch blocks, no soldering and no terminal strips.

While this seems almost like connecting a home stereo system, believe me it is not. Due to the size of our plant, it took quite a bit of time to plan and configure everything. I spent many hours using Visio to create drawings showing how everything was going to connect together. I also developed spreadsheets assigning IP addresses, sources and destina-

Jim Hibbard in Studio

tions to each node.

I found it necessary to bring on Fred Morton, a local contract engineer, to make sure we had the equipment available and ready for installation as needed. Because so much of my time was needed to keep the overall project on schedule, the decision was made to bring in several specialists. Another call was to Mike Schweizer, a phone contractor. Due to the large number of T-1s, ISDNs and the PRIs required for the Telos 2101, we needed someone who could dedicate his time to making sure the installation of the phone services would be correct.

To oversee the day-to-day buildout, we brought in Jim Hibbard, a contractor who specializes in studio construction. Working with me to make sure that the look and feel of the project was what I had envisioned, Jim headed up the studio team.

This included in-house AudioVault expert AD Rigmaiden; studio guru Orlando Valdivia; Assistant Chief Engineer Bill Hartman; staff technician Thomas McDaniel; Sandy Johnson, a contract engineer; and Don Hackler, another contract engineer who works closely with Univision Radio. Rigmaiden and Valdivia are assistant engineers. These folks made my vision of a broadcast facility come to life.

AD Rigmaiden mounts BE AudioVault servers.

Jim brought new ideas. He was able to take what I had in mind and expand it to create something more. He showed us ways to do things we had never thought about and things we could do that would make our lives easier. He was also a fanatic for quality control. I wanted a showplace and under Jim's guidance that is what we got.

LESSONS

Overall, building our new facility went pretty smoothly. We encountered unexpected problems but were able to overcome them and continue on due to the ingenuity of the folks involved.

Here are lessons I learned:

- Plan, plan and plan some more. This helped us find and solve problems before the buildout began and gave us the opportunity to develop backup plans in case something went wrong, as it occasionally did.
- Draw layouts and diagrams for everything, and post them. This proved invaluable and kept me from having to answer the same question repeatedly.
- Make sure you bring in the right people to help in critical areas where you or the staff may not have time or expertise — especially important if you are an early adopter of new technologies as we were.
- Find good vendors. You do not want to deal with just sales people. You want to deal with folks who can listen to your situation and help find solutions that fit. Do not hesitate to ask for a demo and for time for all involved to evaluate their product. This is especially true when you are building a plant with new technology such as IP for audio.
- Delegate responsibilities. On a project of this size it is impossible to do everything yourself. Give others the opportunity to shine. Tell them the desired result and let them figure out what needs to be done.
- Keep communication between everyone flowing. You need to know that others are achieving the results you need. This also ensures that everyone knows what areas need additional help and gives all the opportunity to share ideas for solving problems.

IP is the way of the future. Many stations are streaming and some are using private streaming as a multi-location distribution system over the Internet. As technology and the reliability of new equipment advance, we will be using IP as a replacement for Marti RPU systems, ISDN remotes and even plain old telephone call-ins. The cost, reliability and quality of these methods are going to surpass the way we have done things for years.

Today, more pieces of equipment come with some type of TCP/IP connection than ever. Being able to control a station from somewhere else in the world is now a reality. With a laptop, Internet access and a VPN (virtual private network) connection, we can do many things that once required a visit to the station. Changing audio sources, controlling audio codecs, reconfiguring console layouts are just a mouse click away.

Racks With Layout Sheet

IP is moving in fast; there are few stations, if any, not touched by it. If we want to survive we not only need to embrace the new technology, we must chase it. Waiting could leave us standing out in the cold.

Marty Scruggs has worked as a chief in both radio and TV and has been in broadcasting for more than 27 years. ■

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