ENGINEERING EXTRA

NATIONAL OR LOCAL
Guy Wire discusses the opposing camps on radio programming. Page 22

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Due Diligence Heads Off Acquisition Surprises

Helpful tips for engineers working on acquisitions

A DAY IN THE LIFE

BY CRIS ALEXANDER

As I scan the trades, it is evident that broadcast station trading is ramping up again. Stations are being bought and sold in various locations, and I suspect that this will continue.

Buying a radio station is, in some respects, like buying a used car. Caveat or is the rule, and you can easily ıp with a pig in a poke if great care ›t exercised in the due diligence :ss.

ver the years, I have done a lot of diligence work on station trades, have also worked with local sta- ınd market engineers on such trans- ns. In my dealings on the local , it has been my experience that local station and market engineers are often not equipped for such tasks. It's not that they don't have the skills; it's that they don't have the right mindset.

ENGINEERS ARE CRUCIAL

The decisions that go into the process of a station acquisition are based in large part on the findings and recommendations of the engineer. A mistake or oversight at the technical due diligence stage can have a far-reaching impact on the operation and success of that station.

In this column, I hope to convey some of the things that an engineer should be looking for and thinking about in performing due diligence for a station acquisition.

The primary concern that a station owner or GM will usually have in an acquisition is coverage. That would seem to be a fairly simple question to answer. Drive the signal and find out.

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Use the list of assets to be conveyed as a checklist as you inspect the equipment.

The Power of Wireshark

Packet analyzer provides inside view of computer communications

RADIO IT MANAGEMENT

BY STEPHEN M. POOLE

In previous articles, I've used the Wireshark network protocol analyzer to create various illustrations. Given that this is one of the most powerful tools ever created for network troubleshooting, it's past time that I devoted an article to this marvelous "sniffer."

The usual disclaimer is on a lighted billboard this time: There is absolutely no way that I can cover even a tiny fraction of Wireshark's features in a single article. I will have to oversimplify some things.

My examples were done using Transmission Control Protocol over

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NRSC Revises HD Radio Standards

System improves; but is it in time?

BY MICHAEL LECLAIR

At the Fall Radio Show, the Digital Radio Broadcasting Subcommittee of the NRSC adopted an updated standard for HD Radio. This new standard is available for inspection at the NRSC website, *www.nrscstandards.org*. It reflects a range of improvements and enhancements which improve the flexibility and performance of HD Radio, most of which have been under discussion and testing for some time.

The National Radio Systems Committee is a joint venture of the Consumer Electronics Association and the National Association of Broadcasters. It serves as a standards-setting body for technical aspects of the over-the-air broadcast industry, helping to ensure that working radios are available to consumers as technology evolves.

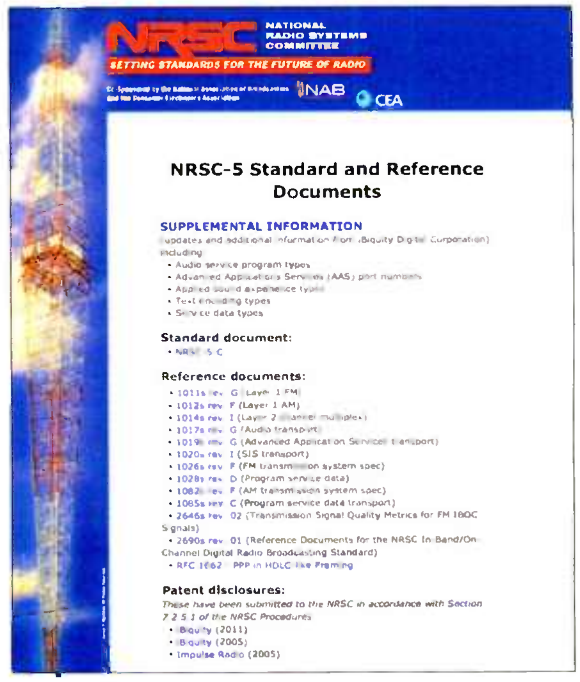
DENSE AND INTERLOCKING

At first glance, NRSC-5-C is a bit hard to dissect. The actual standard, for all its 53 pages, feels incomplete, although the outlines of the HD Radio system can be seen in all the definitions and diagrams. The real meat is to be found in the reference documents, which detail all aspects of the operation and design of the HD Radio system. Reviewing these reference documents reveals all the newest and latest features, both existing and proposed. From the document dates, it appears that iBiquity made a major set of revisions in August of 2011, which are just now being adopted by the NRSC. Some are familiar and some may not be, but the changes reflect the adaptations being made by HD Radio to survive and thrive in the marketplace.

IMPROVEMENTS TO THE SYSTEM

One good example of this flexibility is the formal allowance of asymmetric levels of digital sidebands on both FM and AM stations using HD. This was really an essential answer for instances where digital broadcasting by one station would result in unintended interference to an adjacent or first adjacent station. By permitting lower digital sidebands on one side, HD now offers a satisfactory engineering and regulatory response to such interference.

A second improvement addresses one of the major criticisms of the original HD system — lack of adequate reception, particularly indoors. In response, the new standard includes operation with up to a 10 dB increase of power in the digital sidebands. Coupled with the asymmetrical sidebands above, this allows stations to offer the most digital power possible without analog conversion. HD Radio now has the signal strength to compete as originally intended.



Access a PDF of NRSC-5-C at *www.nrscstandards.org*.

One of the most important strengths of HD Radio is the ability to operate in a hybrid mode during the difficult period of transition when literally hundreds of millions of receivers need to be replaced without losing the entire radio industry. But there were many who complained that the original AM HD systems forced analog stations to limit their audio modulation to a 5 kHz bandwidth. The NRSC-5-C standard unveils two new modes of AM operation that improve the analog bandwidth to 8 kHz and 9.4 kHz respectively. This may encourage some to sample AM HD again, now that limits to the host channel no longer hurt station competitiveness relative to other analog-only AMs.

LONG-TERM PROPOSITION

It seems like a long time since the first test stations began to implement HD Radio on hacked together computer platforms. Although the industry was still embroiled in a furious debate over whether HD Radio was necessary, the early days of conversion saw literally thousands of stations take the plunge into digital radio. But then conversions slowed to a crawl as the economy fell into a recession and they have never really recovered since.

These days if I bring up the topic of HD Radio with fellow engineers, it is easy to see that most were disappointed by digital radio and feel the technology has no future. This feeling is especially strong in the ones who have given their lives and hearts to music formats. Ironically, we're at a moment where millions of HD radios are being sold every year now that car manufacturers have decided to get out of the radio business and back to selling cars. IBiquity President and CEO Bob Struble recently reported to Radio World that over 10 million HD receivers had been sold and that 11 million was not far behind.

My own thinking on HD was that it was never going to be possible to transform the radio marketplace with digital radios unless and until it was accepted by car manufacturers as a standard. With hundreds of millions of radios out there in consumer hands, it was just asking too much to expect them to trade in a radio for a new one that delivered nearly the same performance. However, with the regular 10–12 year replacement cycle of the automobile, it could be accomplished at a measured pace. The question is whether other competitive entertainment services would overtake radio while we waited for digital.

So far that hasn't happened. HD Radio has shown itself to be flexible and on a path to slowly improve its performance and features.

Will it be in time?

Michael LeClair is chief engineer for radio stations WBUR(AM/FM) in Boston; he has been technical editor of Radio World Engineering Extra since its inception in 2005.

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WIRESHARK

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Internet Protocol (TCP/IP) atop Ethernet; while it's by far the most common, it's not the only way. Wireshark's Website (*www.wireshark.org*) and Wikipedia (*en.wikipedia.org*) are your friends and will cover things that I had to omit. Read the documentation!

PROTOCOLS AND PACKETS

A quick refresher is in order before we get started. All network data is sent in packets — short, manageable chunks of data. A large file is split up, sent in separate packets, then reassembled at the other end. Each packet is wrapped with headers containing information on the source and destination addresses and the length of the data. Wireshark lets you examine each of these layers down to the byte level.

A communications protocol is simply a way to format data so that both sides of the conversation can understand it. A higher-level protocol can "ride" atop a lower-level protocol, with each protocol optimized for a specific purpose. The Ethernet protocol "talks" from one MAC address to another. Go up a layer and you'll find the Internet Protocol (IP) "talking" between IP addresses. Riding on top of that might be other protocols, such as HTTP for Web pages and POP for email.

Wireshark understands all common protocols and will display them in a meaningful way. A typical packet capture will allow you dig all the way down into each protocol layer. In Fig. 1, I've captured part of a typical Web browsing session. In the top window, you see individual packets. Click on one and the middle window displays the various protocol layers in the packet. Click the "+" marks to see the details for that particular protocol layer, which will be displayed in the bottom window. The byte data is to the left in hexadecimal, with the ASCII equivalent (if applicable) shown to the right.

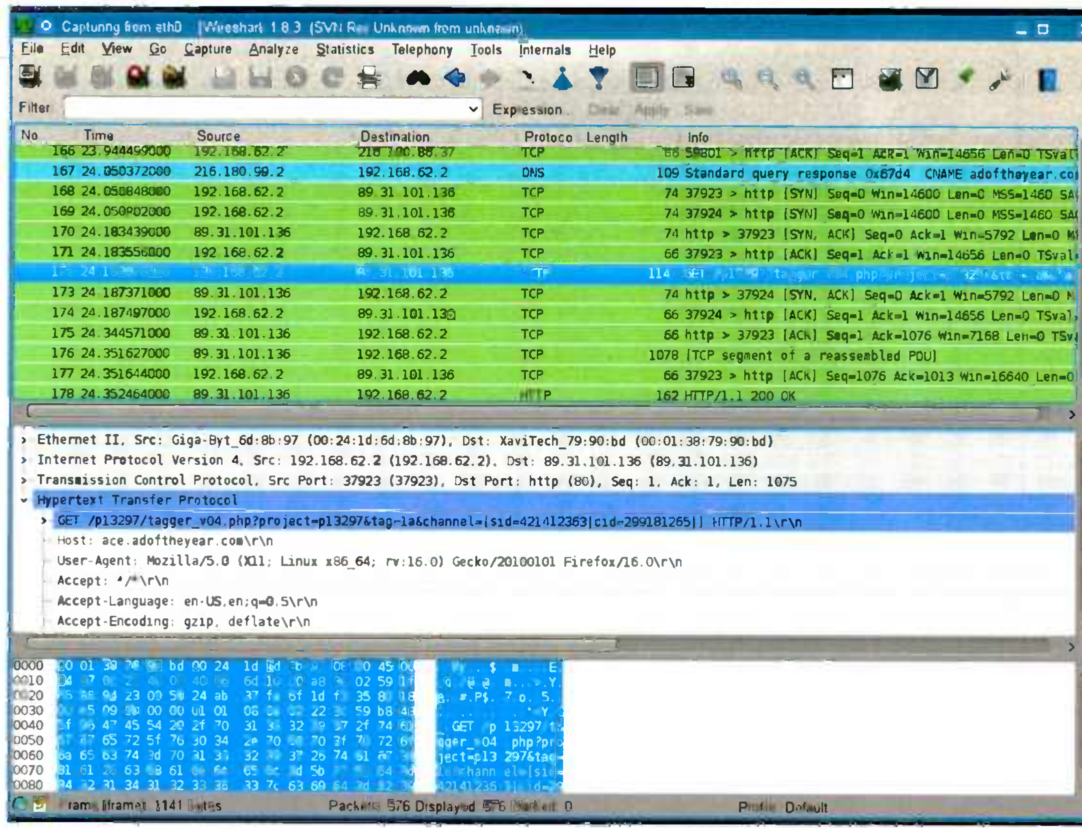

Fig. 1: Wireshark packet capture example. In blue highlight the Web browser performs a 'GET' request to a remote server in plain text.

As you can see in Fig. 1, the HTTP protocol uses plain text commands. In the highlighted example, my Web browser has issued a GET request. It's telling the remote Web server what it wants next. It includes my Web browser (Mozilla Firefox), the desired language ("en," or English) and other information. The server uses this to craft and return a page that satisfies my request.

IT'S EASY TO BEGIN

Download Wireshark from *www.wireshark.org* and install it. When you run it for the first time, don't let all the buttons and menu options scare you off.

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WIRESHARK

(continued from page 4)

You won't need most of them. It's actually easy to start using it right away.

Wireshark provides many different ways to capture data from your network (read the documentation). But I want to get you up and running, so just click the "Capture" item on the menu, then click "Interfaces." This will bring up a list of all network interfaces on your computer. You must select one from which to capture.

Linux uses standard names for all interfaces, such as "eth0" for the first (primary) network card. That's what I've chosen in Fig. 2. Other operating systems (notably, Windows) use a different nomenclature, but in most cases, you should be able to determine from the IP address which one you want. You won't hurt anything if you click the wrong one, so you can try them all (don't be surprised if the "localhost" selection fills the screen instantly; that's an internal virtual network that different programs on your PC use to talk with one another).

Once you select your interface, click "Start." Wireshark will begin "sniffing" the network and capturing a snapshot of everything passing through that interface. When you're ready to analyze the data, click on the stop button.

A warning if you've never done a capture on a busy computer: You will be flabbergasted at the amount of information that flows through your network card. The top window may fill with packets in an instant, making it very difficult to find what you're looking for. We'll address that next.

Go ahead and do some captures to see what happens. You can even amuse yourself. A typical small Windows peer-to-peer network engages in a constant George Carlin routine. Each PC yells, "Who's dere?" followed by, "I'm dere," and "I'm dere, too." "OK, everybody's dere." This repeats over and over, round the clock.

HOW TO FILTER THE ONSLAUGHT

Go back to Fig 1. Near the top, under the menu bar, there's a text box labeled "Filter," with a button to the right called "Expression." This is one of Wireshark's most powerful features. Once you learn how to use it, you can tell Wireshark specifics such as, "only show me packets to IP address 123.1.1.12, for port 1300, which use the XYZ protocol."

Click the "Expression" button to get an idea of just how many different filter options Wireshark offers. Learning all of them requires that you delve into the documentation — and speaking from experience, it takes practice and patience to get a really complex filter worded correctly. But here are some quick-and-dirty filter strings to get you rolling. Print this out and use it as a cheat sheet. You can just type these directly into the "Filter" box yourself.

A typical filter string is of the form, "object==value." (Note the double equal signs.) For us, the most useful objects are:

ip.addr — IP address, source or destination
ip.src — IP address, source (your computer)
ip.dst — IP address, destination (the other computer)
tcp — TCP (Transmission Control Protocol) port number
tcp.srcport — TCP, source port only
tcp.dstport — TCP, destination port only
udp — UDP (Universal Datagram Protocol) port number
udp.srcport — UDP source port only
udp.dstport — UDP destination port only

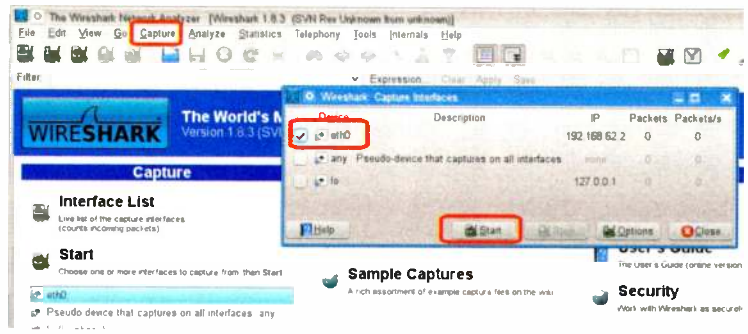

Fig. 2: To begin capturing packets, select the local Ethernet interface on your computer and hit 'Start.'

You can combine strings with "and". For example, suppose my printer is at IP address 192.168.1.12, and receives data on the standard JetDirect port number, 9100. The filter string "tcp.dstport==9100 and ip.addr==192.168.1.12" will let me examine all packets sent to the printer.

There are ways to set up filtering before the capture (useful if you're on a really busy computer), to save and restore files filled with capture data, and more. But what I showed you above just by itself will prove absolutely invaluable right off the bat. Let me close with a perfect example.

HD PAD PROBLEM EXAMPLE

Let's say you're unable to get Program-Associated Data to display on an HD receiver and you're not sure where to look. I'll assume you've checked the cables and other obvious things, that you can ping both the PAD source (typically the server for your audio automation system) and the receiving end (typically your HD exciter/generator). Everything looks OK, but the PAD just won't work.

My audio server is at 100.100.150.204 and the HD exciter is at 100.100.150.233. The exciter receives PAD via UDP on port 1000 (if this information isn't in the manual for your HD equipment, call customer support and ask).

Install Wireshark on the audio server, i.e. the "sending" end. Don't worry, I've never run across a case where Wireshark affected the proper operation of the system. We have it installed on our audio servers. In fact, customer support has used our Wireshark captures to help troubleshoot problems themselves.

In Wireshark click "Capture," then "Interfaces," and select the network card that sends data to your HD transmitter. Click "Start" and let it run long enough for some PAD to be sent (on a music station, a couple of songs should do it). Use the filter string, "udp.dstport==10000 and ip.addr==100.100.150.233"to eliminate unwanted packets and show just the PAD data.

An example of what you might see is in Fig. 3. The audio server for our two talk stations, 101.1 FM and 1260 AM, is sending a stock PAD string for display on the listener's receivers. For the curious, PAD uses the ID3 tagging standard, the same that is embedded in MP3s. Wikipedia has a good article on it so I won't go into it here, but you can see some of the data string as it will appear on the receiver.

Note that Wireshark calls the destination port "ndmp (10000)." If it doesn't understand the protocol (and it doesn't in this case), it will make its best guess. That's why you have the bottom window. You can always look at the raw data yourself.

If you don't see anything, your audio server may not be sending the data at all. A good test might be to edit the filter string by removing the "tcp.dstport==10000" portion. If the data appears, your audio server is sending the data to the wrong port. Next, restore the port number and remove the "ip.addr" string. If data appears, you're sending the PAD to the wrong IP address.

This kind of troubleshooting is especially useful with UDP data, because UDP is a "connectionless" protocol. TCP establishes a connection, and as you'll see from any capture of TCP data, there's constant handshaking: "did you get the data (the SYN and ACK packets)?" But UDP just throws packets at the target and hopes they will stick. If the receiving end gets it, all is good. If not, UDP doesn't care.

With this in mind, let's assume that Wireshark shows that your audio server is sending valid data. We need to troubleshoot the other end. There are plenty of ways to do this, but one of my favorites is to use a laptop to pretend to be the receiving end.

In this case, you'd disconnect the HD exciter and plug in your laptop. Set the laptop to the HD exciter's

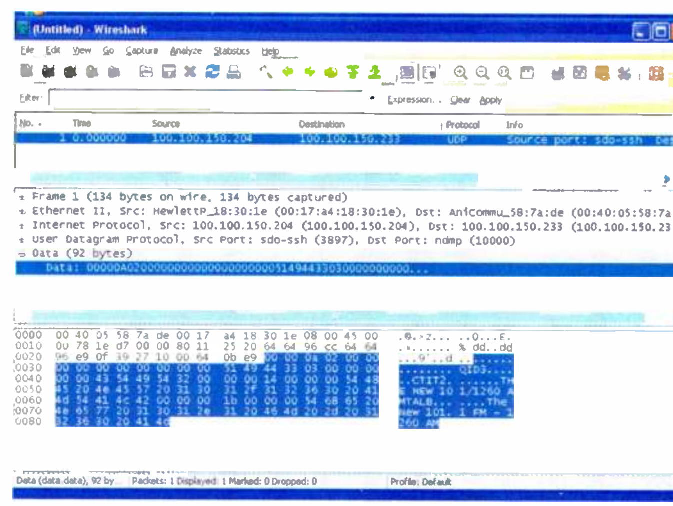

Fig. 3: Program-Associated Data can be seen in the highlighted box in the lower right hand corner.

IP address and run Wireshark on it (Super Tip: you may need to open the target port in the laptop's firewall). If your audio server is sending the data all the way to the transmitter site, you will see it. If so, troubleshoot the HD exciter. If you don't see the data, something's wrong between the studios and the transmitter site. Check your link, whether it's microwave data, DSL or whatever.

WORKS ON ANY DATA

Here's the best part: This approach will help you to troubleshoot any network data communication in your facility, whether it's RDS data to a transmitter site, song/title info to a streaming computer … you name it. I've only scratched the surface, but I hope I've given you enough to start playing with Wireshark. I assure you, once you start using this thing, you'll be sold on it. Have fun!

Stephen M. Poole, CBRE-AMD, CBNT, is market chief engineer at Crawford Broadcasting in Birmingham, Ala.

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DILIGENCE

(continued from page 1)

But it really goes deeper than that. Starting with knowledge of the facility (power, HAAT, directional pattern, transmitter site location relative to the market, etc.) the engineer should make some judgments and predictions before he sets out to evaluate the signal.

For example, if an FM station is a rim-shot, driving the signal may not tell the whole story. The signal as observed in an automobile may seem fine, with just the occasional multipath flutter here and there in the expected places, but it may not be receivable indoors on home, tabletop or portable receivers. I've seen this time and again, and it leads to great frustration down the road.

So in addition to the usual signal drive, take a portable and/or tabletop radio indoors in select parts of the coverage area and see how the station does. If the purchase goes through and there are indoor signal frustrations down the road, at least the decision will have been made with that knowledge and the owner or GM won't be blaming the engineer for not finding out and telling him about it in advance.

Carefully examine the lease agreement for shared sites and be certain what uses and activities are permitted and prohibited.

NIGHTTIME VS. AM

AM stations present even more challenges with regard to coverage determination. All of the above applies — check the indoor coverage in addition to driving the signal — but there is also the element of night coverage to deal with. Time after time, I have to explain to managers that just because a station does not change power or pattern does not mean that the coverage is the same at night. And just because the night coverage is good on the night of the drive test doesn't mean it's good every night. Ideally, the night signal should be evaluated over several different nights.

In one due diligence in the St. Louis market back in the 1990s, our local manager raved about the night signal of the station we were looking at purchasing, saying how it was just as good as the day coverage. As negotiations proceeded, the time came for me to go look at the station, and I found the same thing the local manager did — the night signal was good. Too good. So on the next evening, I set up in a nighttime null location with a FIM and watched the sun go down.

Not surprisingly, nothing changed. I waited 15 min-

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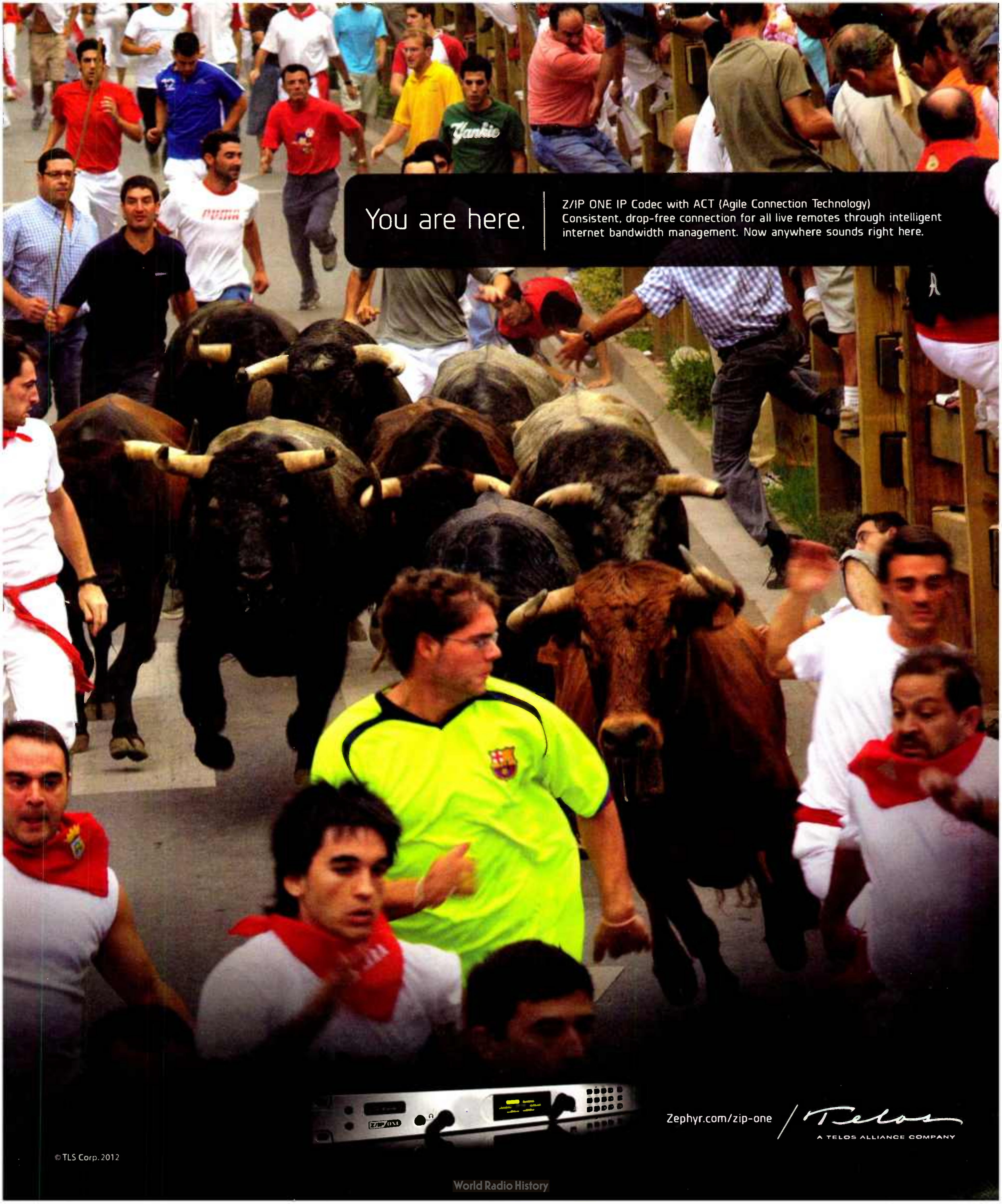
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DILIGENCE

(continued from page 8)

utes past local sunset and still observed the same field intensity at that location. Finally, I called the request line and the jock picked up. I asked him if the station changed power or pattern at night. He said, "Oops" and hung up. I watched the needle on the FIM go to zero, then switching scales I found what should have been there every night. With a deep null right over the city, we decided to pass on the purchase. That would indeed have been a pig in a poke.

KEEP THAT INVENTORY

It almost goes without saying that a thorough inspection of the equipment complement is a vital part of the due diligence process. This is the one part of the process that local engineers are usually well-equipped to handle, so I won't dwell on it here. Still, mistakes can be made.

It's important that the seller provide the prospective buyer with a full inventory of the equipment and assets to be conveyed, and that should be the document the engineer works from as he does the inspection. Check off each item to confirm that it is there and make notes as to condition, operability, etc. Before settlement, the studio and transmitter sites should be re-inspected to confirm that each item on the list is still there and working. It's not uncommon for last-minute substitutions to be made or for equipment to be missing altogether come settlement time.

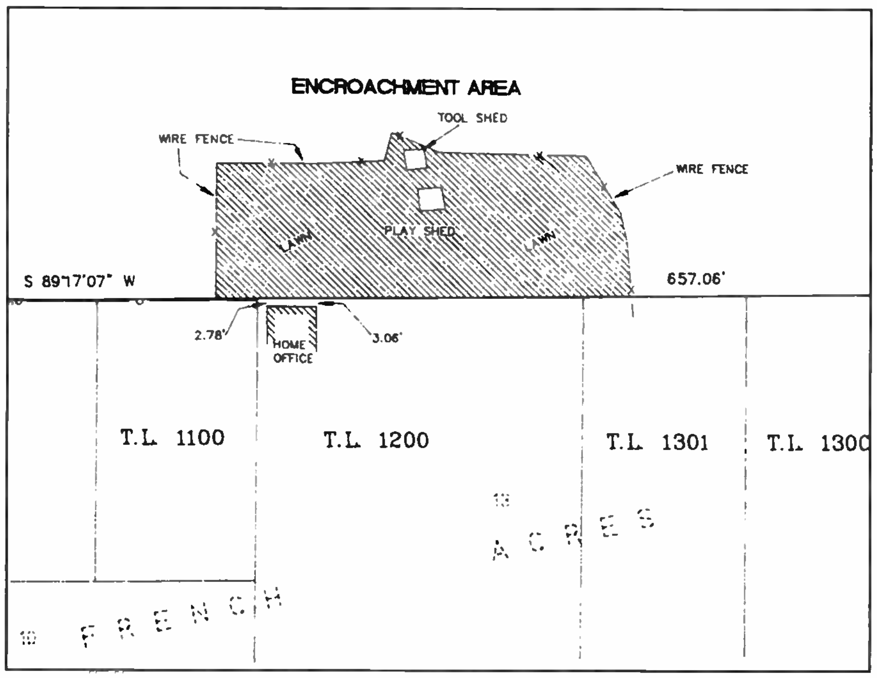
REAL PROBLEMS

One critical area that most engineers may not think about is the real estate, land and/or structures and space that will be conveyed or leases that will be assigned. Land, particularly transmitter site land, can have all kinds of defects that will come back to bite a buyer at some point. It's not uncommon for guy anchors or portions of the ground system to be located off the conveyed property. If not discovered in due diligence, that kind of defect can ruin your whole day.

I like to have a survey made prior to settlement. Often a seller will provide a survey, but the only way to know for sure that what you thing you are getting is what is actually being conveyed is to hand the surveyor the legal description from the agreement of sale, have him stake the property boundaries and then make a survey showing those boundaries and all the improvements.

Sometimes such a survey will reveal an encroachment by a neighbor that can amount to adverse possession down the road. In one acquisition I was involved with, a residential neighbor had actually extended his backyard well into the tower site property — 30 feet or so — and installed a fence and playground equipment for his kids. By dealing with this up front, we allowed the neighbor to continue using that piece of dirt without giving up any of our rights, and we made a friend in the process.

Make certain that STL transmit and receive facilities match the coordinates, heights and equipment/antenna types specified in the license.



This survey revealed an encroachment by a neighbor onto the station's tower site.

The title report should be carefully examined and any exceptions noted. There may well be easements across the land that could impact towers, guy wires, buried transmission lines and the like. Public rights-of-way may extend well into the property, and even though a road may not exist there now, it may come some day and significantly impact the operation. Mineral rights not conveyed can also be a time bomb waiting to go off in some locations. Landowners often have little recourse when the mineral rights owner leases those rights to an energy company and the drilling rig rolls up.

Check with the county or municipality for the zoning, permitted uses, conditional use permits and the like. It's not uncommon for tower sites to have conditional use permits or variances, and sometimes these may have time limits or triggers. If a conditional use permit has a 25-year span and it was issued 23 years ago, the buyer will have to get it renewed or find a new piece of dirt in a hurry. It's best to figure this out before the notice arrives in the mail.

GET A PRO FOR THE BUILDINGS

When it comes to transmitter or studio buildings, hire a credentialed local inspector to do a thorough inspection. There can be all kinds of hidden defects and damage that a station engineer would never pick up on, things that can cost big dollars down the line, such as mold, foundation problems or soft roof decking. Such defects can either be dealt with in the transaction, or the buyer can be prepared to deal with them post-settlement. Either way, it's best to know early what the problems are.

PCBs, lead paint and asbestos aren't nearly the issues they once were, but there can still be environmental hazards and liabilities that can come back and haunt the buyer if not discovered and dealt with. This is another area where the local engineer should insist on an outside professional. A "Phase 1" environmental study is a good place to start, and this consists of a topical inspection and a search of records for events that may reveal a problem. If anything of concern is revealed, a "Phase 2" can be conducted and the hazards can be dealt with.

TRICKY LEASES AND ITINERANT STLS

Leases can be really troublesome, and any due diligence should include a careful examination of the lease by the buyer's attorney and by his engineer. The engineer should pay careful attention to the premises leased, HVAC, hours of operation, rooftop access (for STL antennas, etc.), electrical loads permitted (usually expressed in watts per square foot, which may or may not include lighting), submeters, etc. Watch out for uses and activities that are and are not permitted with an eye to what the buyer intends for the station. It may be necessary, for example, to negotiate additional space from the landlord or get a concession to install a satellite antenna or STL dish. Don't assume the landlord will be willing.

It should probably not come as any great surprise that towers and antennas are not always where they are supposed to be. The due diligence should include a confirmation by GPS that the tower coordinates match those on the station license and antenna structure registration (be careful to convert between data used — Media Bureau records are in NAD27; Wireless Bureau and ASR records are in NAD83).

STL and ICR records are an area of particular concern. It's no great secret that the FCC's database for these stations is full of errors and omissions. The last thing a buyer needs is a notice of violation from an FCC inspector two weeks after settlement. Make sure the STL antennas are where they are supposed to be according to the license, or get the seller to file the appropriate paperwork to fix it.

Finally, it always seems to fall to engineers to inspect the public files and other documents during the due diligence process. It wasn't all that long ago that a story was circulated about a buyer who got dinged by the FCC for missing issues/programs lists from a period of time prior to the assignment. The FCC, in response to the buyer's argument that he should not be liable for the acts and omissions of the prior licensee, stated that the buyer should have performed a "stellar due diligence." That's a word to the wise in any station acquisition. It's going to be impossible to go back and reconstruct missing I/P lists if the old staff is sent packing after settlement, so pay very close attention to these things.

NO SURPRISES

There are many opportunities for the acquisition process to become a less than satisfactory experience, but a careful due diligence can head off most of those things. Dig deep and make certain that you know exactly what is being conveyed in terms of signal, allocation, assets and real property and make certain that what you're getting will do the job you need it to. The best surprise is no surprise.

Cris Alexander is the director of engineering at Crawford Broadcasting and a past recipient of SBE's Broadcast Engineer of the Year award.

MEET AXIA'S NEW, SMALLER IP CONSOLES.

THEY'RE BIG WHERE IT COUNTS.

The more you saw, the more convinced you were that IP consoles made sense for your station. Problem was, you had small spaces to work in. Some behemoth board that looks like a '78 Oldsmobile just wouldn't fit. But there was no way you'd settle for some cheap plastic PA mixer that looked like a refugee from the church basement. "Wouldn't it be great," you thought, "if someone made an IP console that didn't take up a whole room?"

Then you saw the new RAQ and DESQ consoles from Axia, and your problems were solved. With the power and features of a big console, but minus the ginormous space requirements. RAQ will drop right into those turrets in your news station's bullpen – the reporters can send their finished stories right to the studio. And DESQ is perfect for the auxiliary production rooms.

But what sealed the deal was finding out you could run two RAQ or DESQ consoles with just one Axia QOR.16 mixing engine — you know, the one with all of the audio I/O, the power supply and the Ethernet switch built in. That brought the cost down so low that when you told your GM the price, he actually didn't swear at you (for once). Make another decision like this, and you might just be changing the sign on your door from "Chief Engineer" to "Genius."

AxiaAudio.com/RAQ | AxiaAudio.com/DESQ

AXIA

Available in the U.S. from BGS: (352) 622-7700 this simple setup runs rings around any other AoIP network - at ar

Meet the LX-24...Wheatstone's flagship, multi-award-winning advanced modular networkable console control surface

The design initiative behind the LX-24 was to create the world's finest control surface. The result is a console that redefines the entire genre. The LX-24 is an intelligent surface that can store and recall all your settings. Its totally modular design lets you configure it exactly as you like -- you can even hot-swap modules at any time without having to reconfigure.

Assign any source of any type anywhere on your network to any fader. Each input channel can be assigned to four stereo busses, plus four pre/post-selectable aux sends, a stereo CUE bus, four mix-minuses and the panel's own bus-minus. Full Vorsis EQ and Dynamics let you sculpt and control your sound with the quality of the finest dedicated outboard processors. The visually-stunning meter bridge features up to four sets of bright, high resolution LED meters, as well as circular LED displays for auxiliary send levels and pan control. A digital count-up/count-down timer is also included.

The LX-24 is advanced in ways that can make a HUGE difference in your capabilities. But it's also immediately familiar to anyone who has ever sat behind a board at a radio station. Use it to make your programming the best it can be. Just plug it into your WheatNet-IP Intelligent Network - with it, and the BLADES across the page, you can, dare we say it, rule the world.

THE LX-24 CONSOLE CONTROL SURFACE FEATURES

Low-profile table-top design - no cutout required

Meter bridge with up to four bright, high res LED meter sets

Control room and headphone outputs with level control and source selection

Two independent studio outputs

Stereo cue speakers and amplifier, built-into meter bridge

Onboard VGA and USB-Mouse connectors

Event storage (snapshots) and recall

Each input channel features:
- Four stereo bus assigns
- Four pre/post fader aux sends
- Four mix-minuses
- Bus Minus
- Source name display
- A/B source selector
- 2 programmable buttons
- Vorsis EQ and Dynamics including 4-band parametric EQ, High- and Low-Pass filters, Compressor and Expander/Noise gate

T NETWORK

rice. it's called The WheatNet-IP Intelligent Network, and it rules.

Our BLADES carry out your orders network-wide at Gigabit Ethernet speeds – no bottlenecks

As an integral part of the WheatNet-IP Intelligent Network, BLADES interface, move, bend, shape, route and control everything you want to do with your audio. If it's audio a BLADE will handle it – at lightning speed.

Use them organically with our control surfaces, run them from our Glass-E software wherever you have internet access, or control them from the front panels. BLADES make your life incredibly easy and secure.

As you need more functionality, just plug in more BLADES – they come in configurations to handle whatever you need (analog, digital, a/d, mic, MADI). Each BLADE is self-configuring and has the DNA of the entire self-healing network.

With BLADES, you can do everything from a simple (or complex, if you like) snake to STL-over-IP to full-on multi-studio/facility networking – even processing. And because of Wheatstone's partnership with the top suppliers of automation and remote gear, you'll have control over your entire system right from WheatNet-IP. Ruling the world has never been easier.

And this is ALL the extra stuff you need to wire-up the Intelligent Network:

Four CAT-6 cables and a low-cost switch that handles the gigabit speed WheatNet-IP runs at. Let's do the math – plug in eight connectors, power up a console and three BLADES, add your audio and you are ready to rock, roll and rule the radio world. Brilliant, you ask? Nah – just really, really intelligent.

Want to know more?

WheatNet-IP outperforms the other AoIP systems exponentially and is, by far, the most reliable network you can get. Log onto wheatip.com. There is a world of *real* information there. Or, give us a call. There's nothing we like better than talking about this stuff.

EVERY BLADE FEATURES

Two 8x2 stereo virtual Utility Mixers that can be used for a wide range of applications: for example, using Wheatstone's ACI Automation Control Interface, your automation system can control the mix for satellite or local insertion switching

Front panel bar graph meters switchable to display source input level or destination output level after gain trim

Front panel routing control — any system source to any destination on that BLADE

Front panel headphone jack with source select and level control — monitor any system source

Flexible GPI logic — 12 universal logic ports, programmable as inputs or outputs, routable throughout the entire system

Built-in web server so you can configure and control locally or remotely without having to run dedicated software

SNMP messaging for alerts

Silence detection on each output that can trigger alarms or make a routing change

Silent — no fans — can safely be located in a studio with live mics

Wheatstone

How I Quit Worrying and Learned to Live Without POTS

Case studies in VoIP telephony systems

BY JOE TALBOT
WITH KIRK HARNACK

Joe Talbot is product manager of Telos Systems and former director of engineering for ABC Radio's KGO(AM)/KSFO(FM) San Francisco. Kirk Harnack is VP for Telos Products, part-owner of nine stations and host of the podcast "This Week in Radio Tech."

TAKE MY TRUNKS, PLEASE

Just over two years ago, AT&T asked the U.S. Federal Communications Commission to eventually order the shutdown of the Public Switched Telephone Network, or PSTN — the ubiquitous telephone system that provides POTS, T1 and ISDN switched voice service.

In a public response to the FCC's request for comments regarding its forthcoming National Broadband Plan, AT&T acknowledged not the future obsolescence, but the current obsolescence of the PSTN telephone system, the one-time marvel of technology that defined its predecessor, the Bell System, in the 20th century.

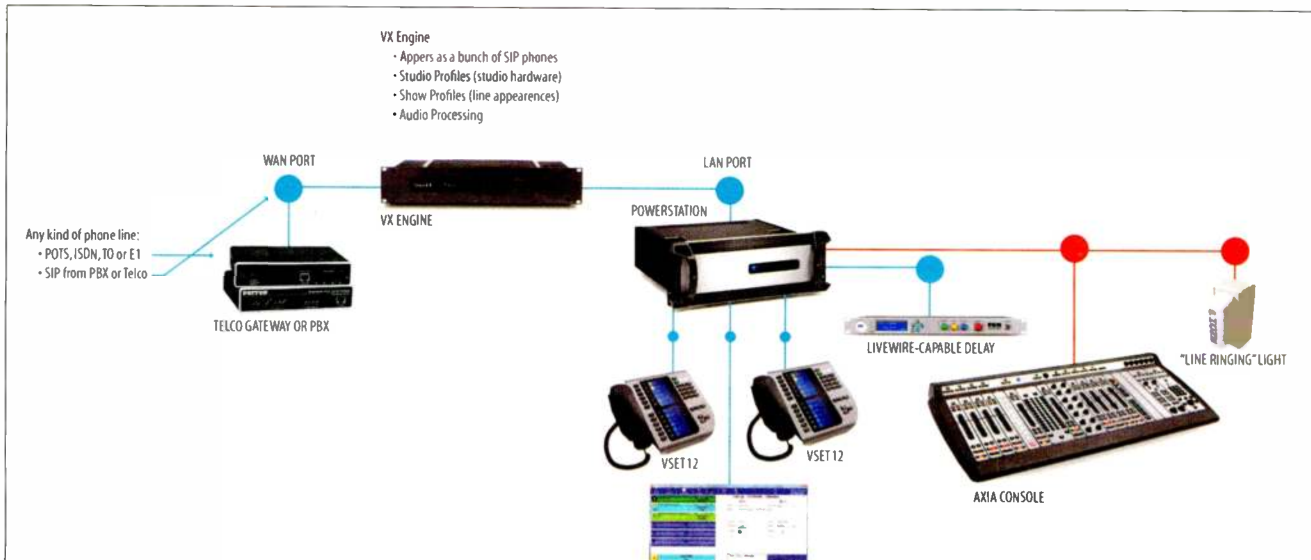
As broadcast engineers, we've witnessed the demise of traditional Broadcast Remote Program Circuits and STL Program Leased Lines quite some years ago. More recently, some telcos have been discontinuing new ISDN installations.

In some locales, traditional, switched phone services are not available at any price for new installations. Broadcasters and other businesses have experienced this in large, modern European cities — not in unwired, third world countries.

Maybe your last-mile, legacy POTS or ISDN service appears secure for now. Consider that more, if not most, of the infrastructure between the caller and your station is already transported as Voice over IP.

Conversely, perhaps your business calls are already brought in as VoIP over SIP trunks. You could be converting these VoIP calls into ISDN or POTS connections locally via your in-house PBX. You're converting these because your studio talk show systems and other phone hybrids don't connect directly to VoIP. They only connect to POTS or ISDN circuits.

By the time a call reaches your studio phone hybrid, it may have experienced several transcodings and several transitions between analog, framed voice path carriage, and packetized voice path carriage, or VoIP. Some of these transitions cause audio degradation, while others do not. None of this outside your facility is under your direct control.



Typical VX VoIP phone system installation.

IN-STUDIO VOIP IS HERE

The Telos VX is a VoIP talk show system for broadcasters. Because broadcasters depend on certain workflows for screening and airing callers, this new standards-based system is designed to fit into all manner of broadcast workflow and program formats.

VoIP technology, and the SIP protocol that carries it, are new to most broadcast engineers. While many of us have connected Analog Terminal Adapters (ATA's) from Vonage or Magic Jack, the specifics of SIP setup remain a mystery to many if not most of us. At this time, a multi-line, multi-studio SIP-based talk show system is not a "take it out of the box and plug in the RJ-11's" proposition.

Two years ago, Steve Church shared his vision for a fully integrated SIP talk show system. Since then, we've completed the beta phase of development and are now actively shipping and installing these SIP-based talk show systems. These case studies demonstrate the various kinds of issues that arise during conversion to a fully digital environment. During the beta period, we learned a lot about SIP

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VoIP

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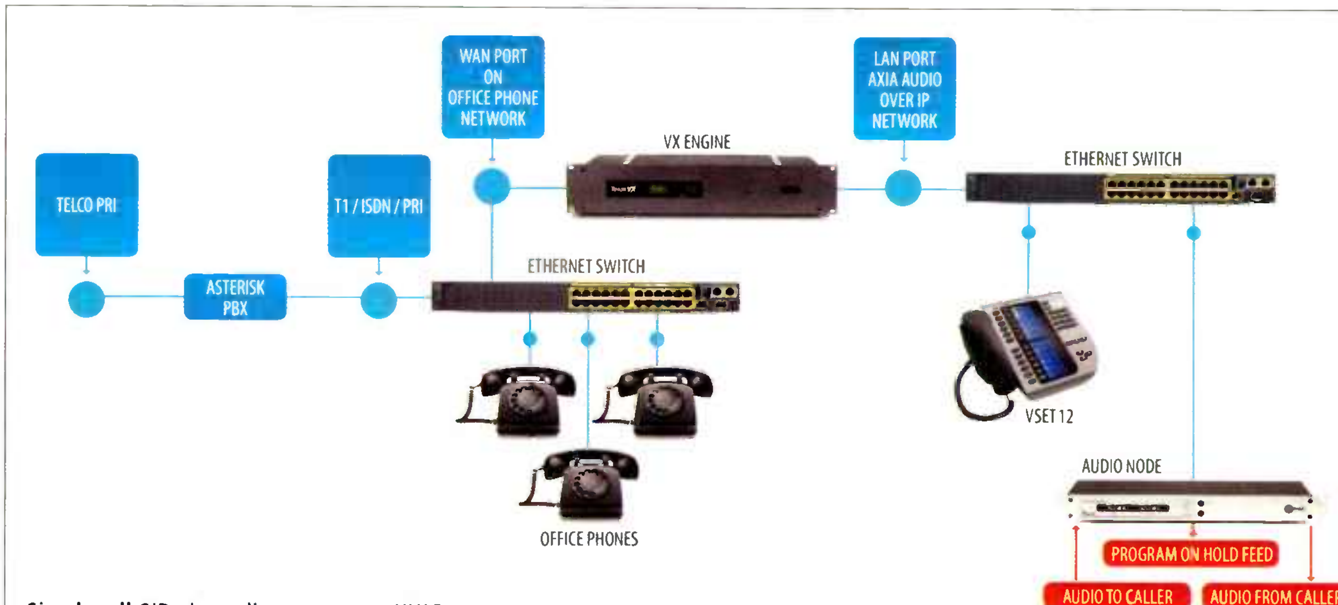
implementations. Many of the issues found are due to differences among service providers, central office switch vendors and network topologies.

The Telos VX system uses the Livewire Audio over IP (AoIP) protocol for broadcast audio I/O. Everything in and out of the system comes over Ethernet, both lines and audio. So any VX installation will have three interfacing considerations: network, audio and telephony. The remaining considerations relate mainly to user preferences and work flow.



Simple, all SIP phone line system at KUIC.

AUDIO EASY

The audio interfacing is the simplest part of an installation. If the broadcaster has Axia consoles, the VX's Livewire streams are simply present on the AoIP network's Ethernet, and can be selected and routed via their normal Web based configuration. Mix-minuses are created for phone channels automatically and control functions are tightly integrated with the consoles. If analog audio or digital AES audio are desired, Telos Audio interfaces convert the AoIP audio streams to and from analog or AES, allowing those signals to be connected to conventional consoles or other audio routers.

TELEPHONY HARDER

More challenging is the telephony side. SIP telephony still has many characteristics of the Wild, Wild West. With multiple service providers, different kinds of gateways, special configurations, and possibly unique customer implementation goals, planning for an installation requires more consideration than with POTS or ISDN.

SIP telephony still has many characteristics of the Wild, Wild West.

The simplest installations have a single connection to the outside world, most often through a VoIP PBX. In these installations, the VX broadcast phone interface or "engine" is connected only to the PBX. All lines and trunks from all providers are also connected to the PBX and routed appropriately by the PBX. The VX is provisioned with "extensions" from the PBX and inbound Direct Inward Dial (DID) numbers are "pointed" to these extensions.

CASE I: KUIC RADIO, VACAVILLE, CALIFORNIA

The first VX system beta test was done at KUIC. The station had just moved into new facilities with a SIP PBX, using the open source Asterisk system. It is connected to the Public Switched Telephone Network (PSTN) via an ISDN Primary Rate Interface (PRI) provided by TelePacific, a regional Competitive Local Exchange Carrier (CLEC). The station has no POTS service. The Asterisk system runs all office and studio phones.

The VX system is connected only to the PBX. All calls, incoming and outgoing, go through the PBX via SIP, and thus there is no further interfacing to the PRI or any POTS lines. This is as simple as it gets.

We used an Axia Power Station to break out the audio and control with this VX installation as it had all of the necessary components "built in," notably a Power over Ethernet POE switch, Analog and Digital audio IO, and GPIO for warning lights and delay dump controls.

CASE II: CORUS COMMUNICATIONS WINNIPEG

Corus in Winnipeg consists of a three-station cluster: CJOB, a full-service 50 kW news/talk station that also originates a fair amount of national network programming; Power 97, an Active Rock station with a high profile morning show; and Groove FM, which is music intensive, though with a live morning show.

The stations' requirements and the availability of SIP trunking from Shaw Communications, a Corus partner company, made the facility an attractive choice to test some new connectivity options with the VX. SIP trunking is the service provided by a telephone company that replaces the functionality of POTS, PRI, T1 or other such services. Trunking is what we call the telephony

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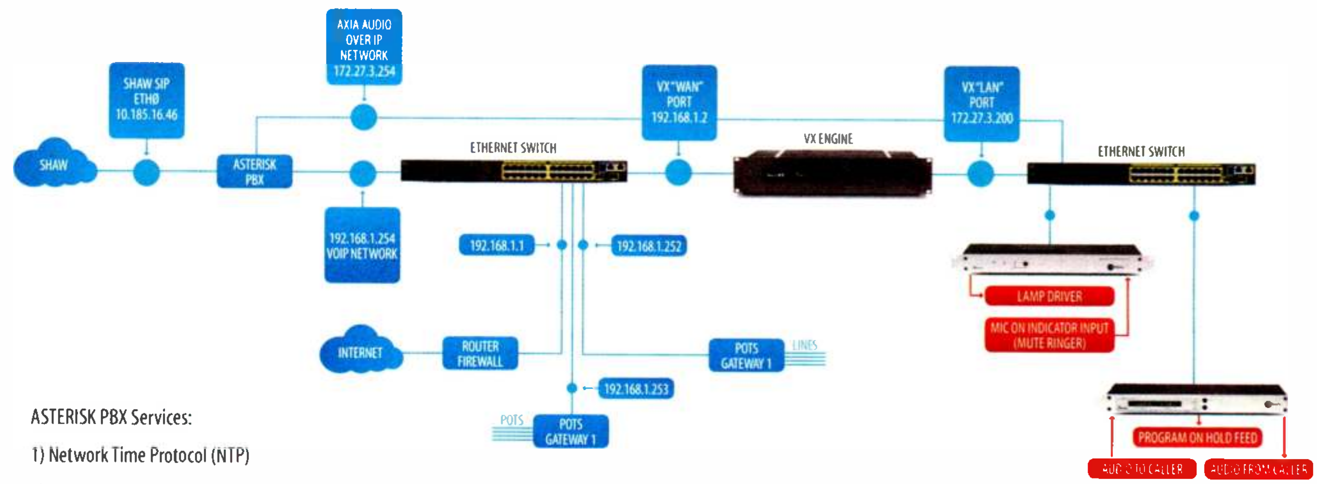
service between a telco provider and our PBX or phone system. Historically speaking, each phone call required a dedicated pair of wires, which was known as a "trunk." SIP trunks require a certain quantity of data per phone connection, which can be part of a multiplexed connection that handles multiple calls simultaneously.

Complementary to SIP trunks are SIP extensions. SIP extensions are Ethernet-transported and are the final link to user devices, such as SIP phones or SIP software clients.

John Wall at Corus wanted to use multiple telephone providers: SIP trunking from Shaw, POTS and a PRI from the local phone utility, MTS, and analog gateways to allow the use of existing Mitel PBX extensions and a few other POTS lines.



ASTERISK PBX Services:

1) Network Time Protocol (NTP)

2) Syslog logging server, logs all devices

3) Call detail recording

4) Voicemail

5) SIP registration

6) VPN access to all networks for maintenance

Combined system at Corus, including SIP trunking, POTS and an ISDN PRI.

DIRECT TO DIGITAL

It's always best to keep lines "four-wire," that is, with separate transmit and receive audio with digital delivery as much as possible. So the station decided to use call forwarding from the analog lines to the new Shaw Direct Inward Dialing (DID) numbers. This way the lines never "went analog," degrading audio performance. This also eliminated the need for additional Analog Terminal Adapters (ATA's), often called "Gateways," used to take two-wire POTS lines and convert them to four wires. We would only have to use the gateway boxes to convert Mitel PBX extensions from the office phone system for use with the VX.

Upon further investigation, it was learned that the Mitel PBX could support SIP stations directly, though with an expensive software upgrade and substantial additional licensing expense. It was decided that this could wait a while as the new studio construction was running on an aggressive time line.

Once the Livewire AoIP audio network was set up, initial tests were made with the newly installed Shaw SIP circuit. Unexpectedly, the Shaw trunk required registration, which at that time the VX did not support (it has since been added to VX and is now fully supported). Registration is the process where the client system logs into the provider's network with an "Auth ID" and password, sometimes called a "secret." Once any one line is registered, all of the trunk's Direct Inward Dial (DID) numbers become available for use. Registration re-occurs regularly as negotiated by the systems.

We also learned that registration status is a good troubleshooting aid. A successfully registered line or trunk will generally have no connectivity problems and its status is easily checked from the VX's web GUI.

ASTERISK: THE TELEPHONY SWISS ARMY KNIFE

With the immediate need for registration support and very limited time to get the whole system working, it was decided to build an Asterisk PBX from available hardware, configure it to perform the trunk registration function and connect via SIP extensions to the VX. Once complete, the trunks successfully came up and calls were routed to the pre-programmed extensions on the VX.

With the Asterisk PBX on the network, it was simple to add additional useful services, such as centralized logging via Syslog, a Network Time Protocol (NTP) server, CDR and Voice Mail with interactive Voice Response (IVR) functionality. The detailed log data has been useful for troubleshooting and allows secure remote network access to the Telos support team as needed. Call Detail Recording (CDR) has been helpful to the station for research, showing the source of all calls and the number and length of those calls, including call attempts to busy numbers.

At the time of this writing, the system has been operating for well over a year and now processes about 120,000 calls per month. It has never crashed or required a reboot.

Finally, the analog gateway was configured, and connected to the existing Mitel PBX. For the studio lines, it was set to forward incoming calls back to the Asterisk PBX, which sends them to a specific extension button on the VX. Outgoing calls could now be placed through the Mitel. The gateway device was configured to send its own logging data to the Syslog server on Asterisk, providing a complete picture of everything happening on the network.

The Syslog server, running as a service on the Asterisk PBX, monitors the Asterisk PBX itself, all Axia hardware, the VX engine, the analog phone gateways and even the Internet router. All devices are time synced via NTP to ensure that logged events are all time correlated. It's been extremely helpful in debugging and the use of Syslog is suggested for most large or complex installations.

CHOKE LINES

Everything was working well — with one exception. We'd tried to "port" the high-volume "choke" numbers over to Shaw. On the day of the porting, we got a call from MTS saying that porting the choke numbers couldn't be done. This has been our experience with all attempts to port choke numbers in North America.

The fallback plan was to manually call-forward the choke lines to the Shaw DID numbers in an attempt to keep the calls "four-wire all the way." A tech was sent to the old studio location to forward the phones. When he dialed the feature code used to activate call forwarding, instead of the expected dial tone that would prompt you to enter the destination number, a confirmation tone was immediately heard. This essentially meant that you couldn't forward the line to the destination of your choice. After working with MTS, we found that the lines were provisioned long ago, with an unusual

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It's always best to keep lines 'four-wire,' that is, with separate transmit and receive audio with digital delivery as much as possible.

ASTERISK — AN OPEN SOURCE SOLUTION

Asterisk turns an ordinary computer into a communications server, which is used to power IP PBX systems, VoIP Gateways, conference servers and other custom solutions. The Asterisk Project was begun in 1999 when Mark Spencer released the initial code. Asterisk is now maintained by the combined efforts of Digium and the Asterisk community.

Asterisk can become the the basis for a complete business phone system, or used to extend or enhance existing systems.

Asterisk is free to use. It requires a working knowledge of Linux, script programming, networking and telephony. On-line training is available. See *www.asterisk.org* for more information.


MULTIBAND STEREO ENHANCER
EXCLUSIVE "UNDO" TECHNOLOGY
VERSATILE DYNAMICS PROCESSING
DISTORTION MASKING
COMPOSITE EMBEDDER
DIAGNOSTIC / ANALYSIS TOOLS
SPEAKER CALIBRATION
RDS ENCODING
FM
STUDIO PROCESSING CORE
HD-1
PATCH POINTS SELECTOR
HD-2
BUILT-IN ENCODING FOR
STREAMING AUDIO
HD-3
TO SAY THIS AUDIO PROCESSOR MULTITASKS
WOULD BE AN UNDERSTATEMENT
Omnia

VoIP

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variation of the call forwarding feature that required the destination number to be set in the central office switch by telco technicians. Eventually, MTS simply changed the type of call forwarding feature on the lines.

For the first few days, only one call at a time was forwarded through to the Shaw DID's from the choke network. This was clearly not the desired behavior! Our discussions with MTS led them to understand and correct the problem. There is a setting in the Central Office switch that limits the number of simultaneous calls or "paths."

A LITTLE HISTORY OF CHOKE

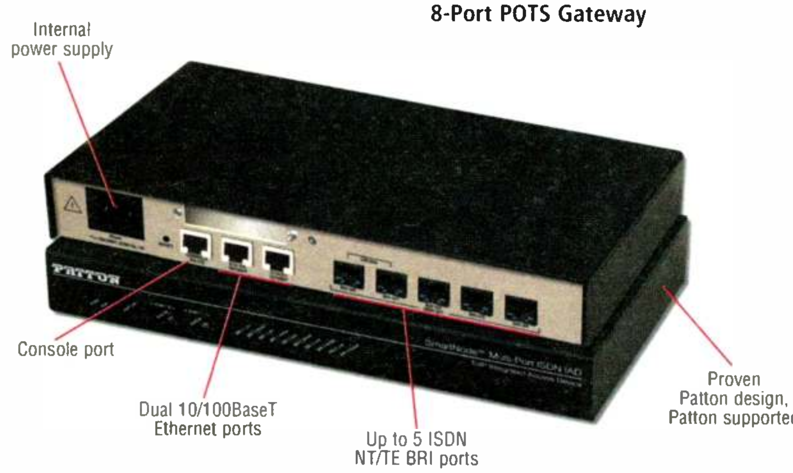
There are many provisioning options in a central office switch, and much history and related lore. To telco guys, the choke network is generally not understood and is usually feared. They are convinced that your massive call volume will bring down their entire network while having no idea what your actual traffic volume looks like.

They also don't want to cause problems that a popular morning radio show host will talk about on the air, so they tend to be overly cautious.

The first choke networks were implemented in the 1960s. The problems that the "choke" solution prevented simply don't exist anymore. The system was created when trunks were the scarce item, to be shared and rationed. Calls to busy numbers would use up a trunk just to play the busy tone to callers.

TI/PRI Gateway

8-Port POTS Gateway



ISDN BRI Gateway

The Public Switched Telephone Network is now much more robust and powerful than the network of say, 1980. There are no more "mechanically switched" central offices and few (if any) analog trunks between central offices. When a call is made today, and it's busy, most often the busy tone comes from your own central office or even the instrument itself. Each call is simply a data transaction.

The telco engineers who came up with the early high-volume solutions are retired and thus not available for consultation about network design challenges or explanations of the system that they designed long ago. So choke lives on, frozen in time and purpose.

When a call is made today, and it's busy, most often the busy tone comes from your own central office or even the instrument itself. Each call is simply a data transaction.

The truth is that most stations really don't need choke numbers anymore. Unless your station is giving away cars or houses every hour, it's unlikely that the local telephone company would even notice your traffic. Still, if you do frequent contests, you may wish to consider asking your SIP provider or Telco to limit the number of calls that it allows through to an individual DID number. This would be done mainly to avoid overloading your PBX or VX that is handling all of the extra call setup messages, which simply tell the phone company to return a busy signal to callers. There's no good reason to keep your systems busy handling call setup messages for calls that will never actually get to you.

IN THE NEWSROOM

The CJOB newsroom has small consoles at each workstation with a microphone and routable inputs.

Phone interviews are recorded at these workstations from calls that come in from the main switchboard. We needed to find a way to get those calls into the VX, and get the hybrid audio of interest into the consoles.

The VX has a feature that allows for a dedicated console fader per line, and dedicated lines also have an available auto-answer function. We simply created a few lines in Asterisk and set them up to auto-answer in the VX. When a call comes in, it is answered on the Mitel office phone by the reporter. If the reporter decides to record the caller, he simply transfers the caller to the VX extension by pressing two buttons. The call is then auto-answered by the VX and the conversation continues through the news workstation console and microphone.

CASE III: COX BROADCASTING, TAMPA, FLORIDA

The Cox broadcasting cluster in Tampa/St. Petersburg asked us to create a system that used every line delivery option available, all at the same time. We like a challenge and knew that this installation would provide us with valuable real-world experience. It did.

Lines were delivered via SIP trunks from Cox's preferred carrier PAETEC. Some PBX extensions from their older Nortel Option 11 PBX would also need to be available in studios. Some POTS lines would also need to get into the system. The engineering department led by Director of Engineering Roz Clark and Chief Engineer Dylan Scott was also intrigued by the possible use of mobile apps and the G.722 wideband codec, supported by the VX directly.

Since the facility has an existing Harris console/router infrastructure, AES and Analog I/O Nodes were used to break out the VX audio for console

MOBILE APPLICATIONS FOR SMARTPHONES

With the widespread availability of smart phones, stations can obtain a high-quality working audio circuit in the field wherever there is wireless phone service. There is a range of free or inexpensive smartphone apps for SIP calling that allow direct audio over IP connections in the field.

Most applications run on either iPhone or Android smartphones and offer a range of audio codecs, including G.722, G.711 or Silk (used with Skype).

By adding an external microphone or headset, you can turn a smartphone into a remote broadcast device that connects via SIP to your compatible phone system or audio codec.

See: Acrobits (*www.acrobits.cz*), iSip (*www.vnetcorp.com*) and csipsimple (*code.google.com/p/csipsimple*) for more information on SIP calling over smartphones.

connection. GPIO would be made available for control of delays, to light warning lamps, and allow the Harris gear to control the VX line functions.

The analog gateway for the PBX extensions and POTS lines was set up easily, leaving the Public Internet/G.722 piece to complete. There were several different ways to approach this. While it's possible to have an app connect directly with the VX, the benefits of one of the features that SIP registration provides became clear: because wireless and WiFi networks present a constantly changing IP address and network environment to the app, the remote device or app couldn't be called if the IP address was unknown.

We decided to implement an Asterisk server to provide registration functionality and a simple and consistent dial plan. The Asterisk server would connect to the Public Internet via the station firewall, allowing SIP access to mobile devices, such as iPhone or Android smart phones. Each station would have a unique private phone number, for example "Hot 101.5" would be dialable by calling 41015. Each app user would have a unique dial in number as well. It would be easy to see at a glance if there was connectivity or not, and overall troubleshooting would be far easier. The presence of that Asterisk machine on the network also gave us VPN remote access, Syslog server, NTP and audio recording features.

CONCLUSIONS

The installations described here illustrate that it's possible to dramatically improve audio quality and operational flexibility, while reducing costs and the number of components in the studio on-air phone system

The installations described considered these important goals when system implementation choices were made:

- How can we keep as many phone lines as possible in the four-wire domain?
- How can we make the system as reliable as possible while still consolidating delivery facilities?
- How can we best position ourselves for the future of telephony, which will be 100 percent IP?
- How can we identify the best service provider choices in our area and for our situation?
- How can we make these dramatic changes without adding complexity or adversely changing our user's experience?
- What are our risks and what is our fall back plan? How much backup do we need?

With some good choices and a little planning, you'll have the best on-air phones that have ever been possible.

Over the past two years, we learned that you should consider more than "is this PBX SIP capable" when shopping. The overall "openness" of the system and manufacturer should be considered when choosing a PBX. In several locations, we've found that it would be relatively simple to connect the VX directly to the PBX, but when we found out that the arbitrary licensing costs to the PBX vendor would be several thousand dollars, other arrangements were made.

We also found that a complementary Asterisk server is a useful addition to any studio phone system using SIP for the additional support and troubleshooting functions it provides when dealing with VoIP calls.

We found that SIP providers will often assume that you want the G.729 compressed codec, not understanding that audio quality is a primary concern that you're willing to pay for. The standard bandwidth G.711 codec is a far superior choice. Never lose sight of why you are making changes in the first place!

Consider how your lines are delivered and how many you actually need. You will get fewer channels (17) on a T1 IP connection than a TDM T1 connection (23). If capacity is an important issue, your best choice is to have SIP trunks delivered in some other (non T1) fashion if possible, or simply use a SIP gateway, bringing in TDM on a PRI from the carrier, then converting to SIP. That scenario is all digital and four-wire.

We learned that the ILEC phone companies aren't the best choices for SIP service at this time. They're just not familiar with the technology yet and really assume that they'll be connecting to you through a PBX.

We've also found that moving to VoIP is more of a network-engineering job than it is an audio or telephony job. You must consider what to expose to the Internet and how. Further, you must consider maintenance, remote vendor troubleshooting and error logging.

As we look forward, we'll find that converting to VoIP and SIP trunks will become easier and, eventually, ubiquitous. We'll also see wideband codecs becoming standard in mobile phones and tablet computers, so your VoIP broadcast talk show system will be communicating with them, too, achieving audio quality you've never experienced over a "telephone."

Nationalization vs. Localization

Two views duel for the future of radio

BY GUY WIRE

Guy Wire is the pseudonym of a veteran broadcast engineer.

We have all heard the hype about radio needing to be live and local to compete with satellite, Pandora and all the other new options coming out of the cloud. However, at least two of the largest group owners are pushing a contrary model. Except for their largest markets, live and local is apparently not succeeding for Clear Channel and Cumulus.

At the core of their new strategy is a move to "nationalize" the programming and overall operations. They won't call it that, and actually say it's a better way to save and improve radio's appeal. These are the primary ingredients:

- Homogenized formats for all markets, planned and managed by corporate and regional team leaders;
- Distant voice-tracking across most markets;
- Syndicating the most popular shows across multiple markets;
- Elimination of most local station personalities;
- Fully centralized automated delivery systems and support functions for all stations.

It's really no surprise these two and a few other group owners have been steadily moving down this road. They're heavily leveraged and compelled to show their lenders they are doing everything humanly possible to reduce expenses and maximize revenue to remain solvent.

THE TRIED AND TRUE

While all radio owners have had to cut expenses and tighten their belts to keep their companies profitable, most are not adopting a nationalized or corporate mandated operational template for all their owned stations. These owners are committed to the importance of the tried and true, live and local formula as radio's key advantage in the multi-media war of attracting and holding audiences.

iStockphoto/Barry Gregg

Distant voice-tracking by talent not familiar with local names, places, popular events and important issues gets exposed rather quickly. In particular, late-breaking news stories with heavy local impact are always best covered by real people in real time who are on the scene.

These owners also know that their stations have to earn their place in their communities as respected contributors to the overall health and welfare of those communities. It is critically important to gain the support of their local area businesses for advertising revenue. They hire the best managers and PDs they can find and let them do their jobs.

THE BATTLE OF VISION

What we have shaping up is a battle of two very different operating philosophies. Which one will better allow radio as a business to survive and prosper in the future?

On one hand, we have a highly indebted group committed to the idea that central planning and control is the best way to survive and succeed. On the other hand, we have another group that champions the importance of local service and control while reducing or carrying minimal debt to remain successful and grow. Some owners will cherry-pick and blend the methods they like best from both groups.

What we have shaping up is a battle of two very different operating philosophies. Which one will better allow radio as a business to survive and prosper in the future?

Companies that are highly-leveraged and betting on their one-size-fits-almost-all approach to management are really in a race against time. Some are staying afloat with financing that could be labeled funny money. Their debt, much of which was acquired when valuations were grossly inflated, simply gets restructured and refinanced over and over with the assistance of private equity firms.

Some investors do lose money, but the big ones get protection and incentives to stay the course. Sadly, the CEOs and upper managers get paid every time the debt is rolled over. They keep promising better performance is just around the corner. Everybody buys in and they all keep kicking the can down the road together.

The game stays alive with business as usual as long as interest rates stay low and the bankers remain willing partners. If and when inflation returns and rates spike up dramatically, the game could end rather quickly.

WHAT'S COMING

As new media players continue to chip away at radio's traditional base and profitability, radio station valuations will probably continue to slowly erode. At some point, the Wall Street bankers and venture capital groups will run out of patience and want to cut their losses.

The petition to move Nassau into Chapter 7 liquidation by Goldman Sachs last May is a good example. The accumulated debt in excess of $200 million simply became too large for primary lender Goldman Sachs to accept the eventual risk. The group of approximately 50 stations was largely liquidated in a court-run auction at a small fraction of its book value. Other distressed groups may face similar proceedings in efforts to bring debts in line with the underlying value of their stations.

Breaking up debt-ridden groups and selling off the assets in pieces is usually the best way the debt holders can move on to more promising investment opportunities. Owners unburdened by heavy debt who want to stay in the business and grow will be the logical buyers of such stations when prices fall to attractive enough levels.

Unless of course, and perish the thought, there is some kind of government bailout that comes to their rescue. It's ironic that the companies pushing the nationalized model are saying it's the best way to actually make local radio better.

MONKEY BUSINESS

Where the nationalization approach is ascendant, the original programming and local management decisions have been judged as "not good enough" by others sitting many thousands of miles away. Where else do we see this kind of thinking being promoted?

One size certainly doesn't fit all in the changing landscape of making radio succeed and survive. Our monkey-see, monkey-do business is just one of many in this country coping with a similar metamorphosis. Will more of the monkeys see and do like some of the so-called leaders of our industry are doing?

Read Guy Wire's archive under the Columns tab at radioworld.com.

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NewBay Media

Digital PowerRadio: Advances in HD Radio

Receiver techniques will produce improved coverage and reception quality

WHITEPAPER

BY BRANIMIR R. VOJCIC, HAKAN DOGAN, WOOKWON LEE, STYLIANOS PAPAHARALABOS AND FARNAZ SHAYEGH

In HD Radio (HDR) systems, improved reception quality and increased coverage of the primary signals represent key drivers to improve the domestic and international adoption rate of HD Radio. Also, one of broadcasting's biggest opportunities is assuring strong HDR reception into smartphones and other mobile and portable devices, as well as car and tabletop radios.

The current HDR is based on 1990s technology, similar to what was used for second-generation mobile cellular systems. Meanwhile, other competing technologies have been continually improving, such as mobile cellular evolution into third generation (3G) or even fourth (4G). Similarly, some of the other digital broadcast systems have evolved in the past decade, for example DAB in Europe.

Current HDR performance issues are an obstacle to consumer adoption, as well as to international adoption. The state of HDR presents itself the last opportunity to change the dynamics and offer a superior system that 1) achieves maximum performance, 2) meets consumer expectations, 3) promotes international adoption and 4) provides a growth vehicle for the broadcasting industry.

In this paper, we present advanced receiver techniques that achieve improved performance of HD Radio and change HD Radio dynamics such that HDR receivers can operate at much lower signal-to-noise ratio (SNR).

These enhanced techniques do not require changes in existing HD Radio transmission format or infrastructure, only very doable changes in the baseband receiver chip in radio receivers.

Enhanced receivers will enable broadcasters to maximize coverage and performance. This will in turn improve customer satisfaction, promote HDR adoption and increase the listener base, and ultimately enable growth of new digital and interactive applications. The performance also will significantly improve coverage for handsets/portables and indoor radios in FM, which will enable larger listener base in FM and help international adoption of the HDR standards.

This work was supported by Digital PowerRadio LLC and is based on the intellectual property developed by Digital PowerRadio.

The advanced receiver techniques are intended to enhance the performance of processing techniques used in the current HDR receivers without any impact on the already deployed

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Practice Makes Perfect

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Saving AM: Time for Radical Change?

Structural problems need correction

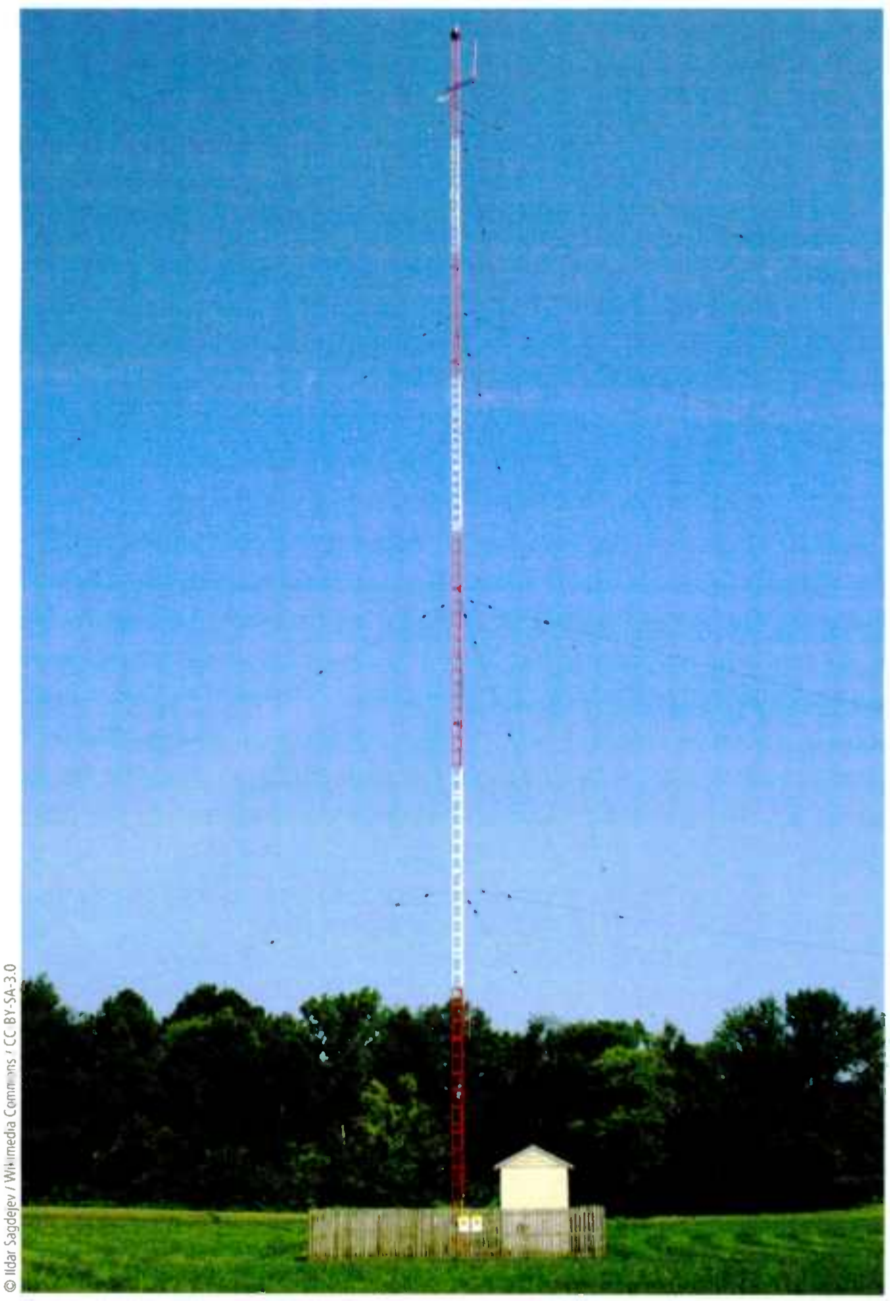
BY MICHAEL LECLAIR

I'm encouraged by the recent groundswell of concern for the AM radio band. Every year seems to make the operating conditions for AM stations a little worse. Each year the land required for an AM directional antenna field gets more expensive to acquire or maintain. Each year electromagnetic noise grows a bit more, especially in desirable urban areas. Each year it seems the receivers get worse. Every year the cost of electricity gets a bit higher while the reliable coverage area shrinks. Every year it seems fewer people take AM seriously, as owners, listeners and radio suppliers. New portable electronic devices that sometimes feature FM radios rarely, if ever, offer AM tuners.

NOT WISELY BUT TOO WELL

The problems of AM can only be described as structural and fundamental to the band itself. Key to today's challenges are the overly optimistic allocation policies of the post-World War II FCC in its mission to bring radio to everyone.

In the process, stations were created that were crippled from the beginning: daytime-only or so completely interference-limited that service could only be expected to carry about 8 miles at best. As if this wasn't enough, the widespread use of very expensive and complex directional arrays created a tier of stations that could scarcely afford to operate, let alone legally maintain their limited patterns. Everyone knows a painful example of one of those.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This "allocate at all costs" approach worked until real competition came along with the growth of FM. Today it is hard not to see the underlying flaws of creating a system that mixed a few very powerful stations with thousands of severely limited signals in a hierarchy that worked only for the largest players.

THE INTERFERENCE SPIRAL

Then there were the attempts to increase audio bandwidth to compete with FM. This virtually guaranteed that stations would interfere with their first- and second-adjacent neighbors, again due to a defective allocation scheme that placed stations within 5 and 10 kHz of their neighbors.

The adoption of the first bandwidth limiting NRSC standard in 1988 further demonstrated that AM is often its own worst enemy, with stations now encouraged to use vicious amounts of pre-emphasis as long as they complied with an overall bandwidth limit of 10 kHz. Radio manufacturers countered with ever-reduced quality in AM radios, noting that listeners dislike interference more than they wanted high-frequency response. Subsequent revisions to the NRSC, including the recent NRSC-2-B in September of 2012, have tried to refine and improve the bandwidth mask in the hope that it will encourage receiver manufacturers to make better receivers.

Interestingly, a simultaneous standard, NRSC-G100-A, began finally to address the problem of encouraging structural interference by exploring the quality improvements that could be met by limiting bandwidths below 10 kHz. Unfortunately I haven't yet seen the discussion move to what is an obvious

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The Best Process for HD Radio Audio

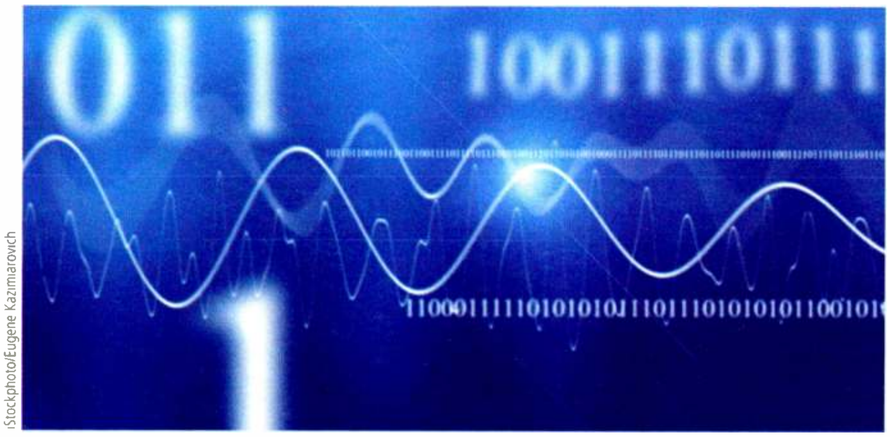
BY STEPHEN POOLE

In the discussion forum at the amusingly-named GearSlutz.com, there's a thread from July 2011 with the title, "HD Radio sounds like junk." The poster admits that the high end is better than analog FM, but complains that it's "swishy," like a low-bitrate MP3 file. The first commenter to that question offers, "Possibly this is a fault of the radio station processing?" Indeed, it very well might be.

HD Radio *does* use a relatively low bitrate. With our present "hybrid" (i.e., analog+digital) systems, AM HD-R typically runs at 40 kilobits per second. A single FM channel in HD-R is typically 96 kbps. But many FM stations multicast, splitting that bitrate across several HD "streams." The result in both cases is that the final bitrate is comparable to that for a Web stream.

Most of us made this mistake when we started streaming over the Internet: We split the audio from our existing processor to drive the encoder. We then discovered that it sounded *awful*. The same is true of HD Radio, and the low bitrate is the primary reason.

Omnia's Frank Foti released an important paper about this several years ago titled "Audio Processing and HD Radio." It's available at *omniaaudio.com/tech/Audio_Processing&HD-Radio.pdf* and is recommended reading.



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In that paper, Foti says flatly, "A processor for HD Radio has a completely different set of requirements. The most important issue is in dealing with *data reduced audio*" (emphasis mine).

Foti's paper details how clipping, in particular, must be given special treatment. Most top-of-the-line processors, including those from Omnia, actually have two separate peak limiters. A more traditional "clipper" might be used with the analog, but a softer "look-ahead" limiter will be required for a digital stream. *Hard clipping is an absolute no-no with any digital "stream" — HD Radio included.*

My buddy Jeff Keith with Vorsis (*www.vorsis.com*) had this to add when I queried him via email: "In any coded environment it's important to not give the codec something to code that's (1) difficult, and/or (2) takes bits away from audio that was supposed to be coded." In his words, codecs are "very good at 'hearing' all of the things that are in the audio." But they can't differentiate between the good and the bad. Therefore, Jeff says, *we* must supply the smarts in the processing and "not give it anything bad."

For the best performance, you'll buy a processor that's "digital stream-ready." But if you're on a strictly limited budget, you may have to be prepared to make compromises with an older, pre-HD and pre-Web stream processor.

Here are some tips: First, reduce the clipping as much as possible. Next, watch the levels! Don't ever overdrive the codec's input. Remember not to use

> **For optimum performance, you'll buy a processor that's digital stream-ready.**

standard FM pre-emphasis. Finally, you may need to slightly roll off the highs after the processor to further ease the load on the codec.

It's not ideal but you can get good sound on a low-bitrate digital stream like HD Radio even with an older processor. This will buy you some time to save up for that dedicated HD audio processor.

Send your HD Radio tips or column suggestions to rwee@nbmedia.com.

Stephen M. Poole, CBRE-AMD, CBNT, is market chief engineer at Crawford Broadcasting in Birmingham, Ala.

SAVING AM?

(continued from page 3)

ous solution: Get rid of the pre-emphasis curves and remove all that extra energy from exactly where it does the most interference damage, right on top of adjacent stations. At least the conversation is proceeding in a rational direction with the realization that all you are going to get is 2–3 kHz of audio bandwidth unless AM stations themselves stop creating all this unnecessary interference. A frequency response of 5–7 kHz can actually sound quite good. It's sad to observe that it has taken nearly 25 years to see the first inkling of this understanding. Meanwhile AM radios remain uniformly dismal.

STILL IN DEMAND

If AM is so fundamentally flawed, why try to save it at all?

It's a reasonable question and sometimes it feels as if we have reached a de facto consensus to just let AM crumble away and die from neglect.

Personally, having grown up listening to AM radio in the 1960s, and having worked for many years at AM stations, I feel there is still something there worth keeping. One advantage of good AM is that all it requires is a simple and inexpensive tuner and transmitter design. This makes it a perfect free consumer technology. AM offers advantages in coverage compared to FM when it isn't so severely interference limited. There's no multipath or terrain shielding.

And there are still so many voices that strive for broadcast coverage, as witnessed by the number of applicants for new licenses that occur at every FCC window. Why not keep AM alive to serve the many that want to become broadcasters but don't have the money to compete for FM licenses with huge corporations like Clear Channel or Cumulus?

RADICAL CHANGES

Given the number of structural problems faced by the AM band, perhaps it's time to consider more radical solutions than we've seen in the last 25 years.

I recall 20 years ago it was sometimes joked that the way to fix AM was to eliminate all daytimers, DAs with more than two towers and stations with less than 5,000 watts. Perhaps there is truth in that old chestnut, and the only way forward is to come up with a method to reduce the structural defects of too many allocations. The trick is to come up with the method of doing so that shares the pain and the cost amongst the winners. I'm guessing the FCC would be open to just about any approach and might be able to offer some incentives.

While we're at it, is there really a justification for the remains of clear-channel allocations clogging up the band? Regional service is pretty neat, especially in the wide open parts of the country. But in densely populated areas allocating just a few stations with signals that reach many millions delivers a solid benefit to just a small group of owners. Those owners are especially motivated to keep AM profitable, but if AM disappears they will suffer with the rest. Some might say these stations are the only ones with the resources to save AM.

Further work remains to be done to improve receiver bandwidth. If there are fewer allocations this will be easier. Digital AM has some promise in the long run, though it has been criticized by many as a cause of even more interference to the existing allocation scheme.

I also like the ideas proposed by Ron Rackley and Ben Dawson recently in the pages of RW (see *www.radioworld.com/freshlook*). For example, if we eliminate the minimum efficiency and ground system requirements for AM antennas, it would make it far simpler to "repack" the AM band in such a way that separations can be increased in a meaningful way. Currently this is impossible; the technical standards meant to maintain AM "quality" are a lead weight clamped on a lifeline. It would also greatly reduce the cost of operating AMs in terms of land and local regulatory burdens.

We should take a page from the DTV band repacking and see what would happen if we actually started to eliminate some channels and move the others around with the antenna burdens removed.

We have already tried the small stuff. For AM it's time to consider just about any idea.

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Real-Time FM Stereo Separation Management

New technique helps reduce perceived multipath distortion

BY JEFFREY A. KEITH

The author is senior product design engineer at Wheatstone Corp. in New Bern, N.C.

New findings indicate that real-time stereo separation control can help reduce the audible effects of multipath-induced stereo receiver distortion. It has been shown that the perception of multipath interference can be reduced via program-controlled modification of the program audio in the L–R difference signal.

It is important to note that this method is not intended to replace other methods for mitigating FM multipath, such as the use of single sideband suppressed carrier (SSBSC) or SSB. Indeed, Wheatstone is one of three major broadcast audio processor manufacturers offering SSB in the stereo generator of its high-end FM audio processor.

However, field testing has revealed that the SSB modulation method can cause stereo decoding issues with certain consumer receivers; some don't decode it at all and blend to mono.

FM stereo multipath control based on adaptively adjusting the L–R signal offers another, and possibly a superior method of reducing the perceived effects of multipath because this method is compatible with *all* stereo receivers regardless of their demodulation scheme.

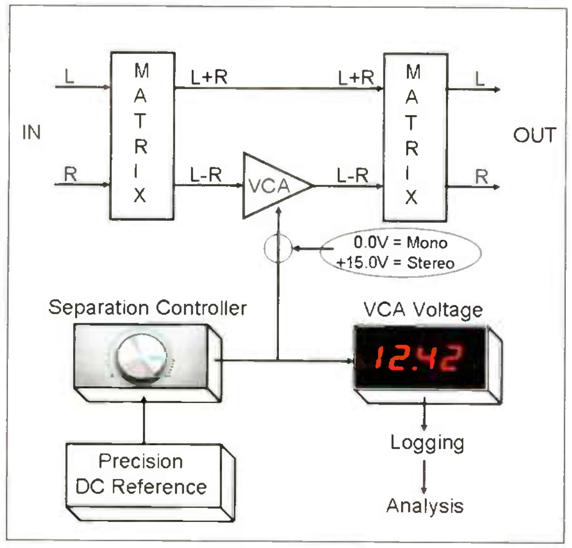

Fig. 1: Stereo Separation Test Fixture

CORRELATION BETWEEN STEREO ENHANCEMENT AND PERCEIVED MULTIPATH

FM transmission technology has evolved to the point that FM stations can easily achieve stereo separation beyond 60–70 dB. Unfortunately, most consumer receivers rarely achieve 40 dB stereo separation in the mid-audio frequencies.

Field experience has revealed that such high stereo separation isn't required to deliver a great stereo program experience to a station's audience. Supporting this hypothesis is that listeners with consumer-grade receivers can't even hear all the stereo separation transmitted by their favorite FM station.

At a radio station where I once worked, we had noted that our loudness was compromised when airing early "ping-pong" stereo recordings. While those songs were playing, our loudness was lower than other stations in the market. Given this challenge, I set out to design a pre-processor that could manage our stereo separation on the fly.

DETAILS OF LISTENING TESTS

First, we needed to determine the amount of stereo separation required for the average listener to have a favorable stereo experience. Nearly five dozen test subjects were willing to participate in my listening experiment, not a statistically high sample, but nonetheless useful for lending insight into what was suspected but not documented before.

Each subject used a few of their favorite CDs as their "reference." The sole objective was to determine how much stereo separation someone required in order to believe they were hearing normal, natural stereo.

A special test fixture was built that would allow

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FM STEREO

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stereo separation to be adjusted by an unlabeled knob between mono and normal stereo. Each person could listen on headphones or speakers, or both, at their discretion. Fig. 1 shows the test setup used for the experiment.

Each listener was instructed to start with the Separation control fully CCW (mono) and stop turning it clockwise when they believed stereo separation was normal.

As each subject found the control setting that pleased them, the VCA control voltage for that setting was measured and documented. Later, the settings for each result were duplicated and the actual stereo separation measured.

TEST RESULTS A SURPRISE

The data shown in Fig. 2 reveals a surprise finding: few subjects needed more than about 25 dB stereo separation to perceive a "stereo" listening experience!

The data gathered during the experiment can be summarized by stating that male subjects required more stereo separation than females, with a difference of about 12 dB. All subjects required far less than the >90 dB stereo separation available from the test fixture and audio source.

No test subject considered themselves to be an "audiophile"; they were average radio listeners. Fig. 3 shows the stereo detector circuit.

OTHER SUPPORTING EVIDENCE

Later, while searching for more data on the perception of stereo separation, a pair of papers (see references below) were discovered that addressed it in a multichannel sound environment. Fig. 4 is a summary of the data from Stuart and Schroeder, showing how little stereo separation is required for a given amount of perceived stereo image width. The data corresponds well with what was measured during the experiment related to this project. It seems that our perception of stereo separation and the actual stereo separation are only indirectly related. In fact, so far the evidence suggests that the stereo separation of most typical FM program content could be reduced quite dramatically to solve the mono loudness problem and without compromising the stereo listening enjoyment for typical radio station listeners who are not audiophiles.

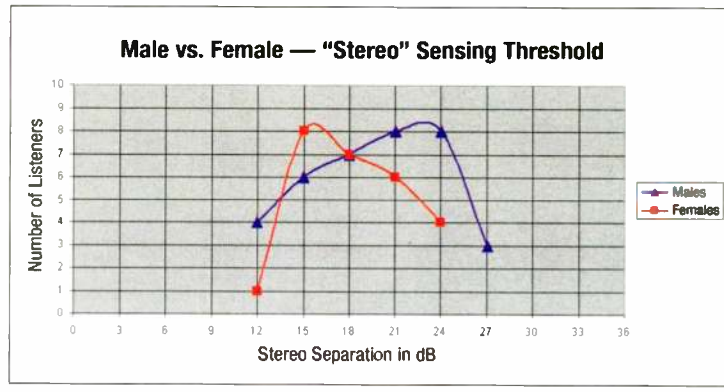


Fig. 2: Male and Female Stereo Sensing Thresholds

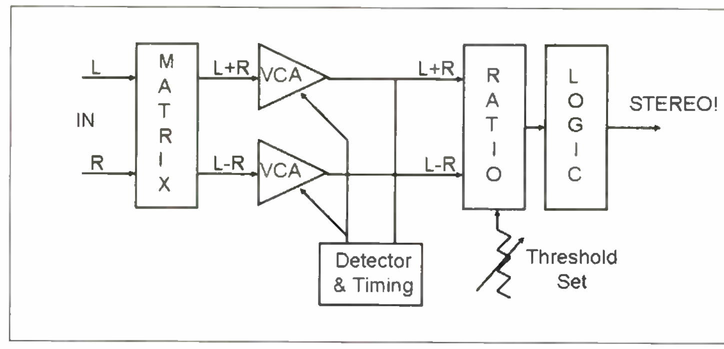


Fig. 3: Stereo Detector

STEREO CONTROL DEVELOPMENT FOR ON-AIR

Since no product existed that could perform the function of "Stereo Width Management," I decided to develop a new type of audio processor, one that could dynamically manage the width of stereo program material in order to even out our station's loudness on mono radios.

When the Width Management processor was put on the air, the loudness of stereo and mono program material was now equal on mono radios. Stereo programming sounded completely normal, even when the processor's metering showed that it was performing aggressive reduction in electrical stereo separation.

The processor cured the mono loudness issues that had plagued the station, but it had a second and surprising benefit. In areas where the station's stereo signal had previously been unpleasant to listen to because of multipath-induced receiver distortion, it was noticeably cleaner; when blending did occur, it was virtually unnoticeable. This was *not* expected!

Whenever the processor was switched to bypass, the noisy receiver blending returned. In the Operate mode, blending was much less noticeable. This confirmed that the reduction in apparent multipath was due to the processor's management of the amount of L–R signal being transmitted.

A highly refined DSP version of the original analog prototype is now a feature in many Wheatstone FM audio processors and is called the Automatic Multipath Limiter. Customers report that it usually reduces multipath-induced receiver disturbances, and in some cases seems to extend their stereo coverage area.

Of course, additional tests are welcomed and are being conducted by Wheatstone to determine if there are other characteristics of a stereo broadcast signal that can be controlled to further reduce the perception of blending on stereo receivers — including continued exploration into the feasibility of using single sideband (SSB) instead of double sideband (DSB) for the L–R subcarrier.

Stereo Separation in dB	Apparent Stereo Image Width in %
15.2	74
12.8	67
10.9	60
9.4	53
8.2	48
7.1	42
6.2	37
5.4	33
4.7	29
4.1	26

Fig. 4: Stereo Separation vs. Perceived Image Width

This white paper is based on a presentation prepared for the 2013 NAB Broadcast Engineering Conference.

REFERENCES

[1] Stuart, J., R., "The Psychoacoustics of Multichannel Audio," *AES Conference: UK 11th Conference: Audio for New Media (ANM)*, March 1996, pp 7, Figure 16.

[2] Schroeder, M., R., "Listening with Two Ears, *Music Perception*, Vol. 10, No. 3, 1993.

The Atomic Option

Cheap and accurate clocks, radiation-free

BY AARON READ

Image courtesy of NIST.gov

A few years ago, cheap "atomic clocks" flooded the market. Now it's possible for anyone with thirty bucks to own a simple LCD clock that has high precision, is visible across a room, and displays seconds — all things that are very useful to radio broadcasters!

While none of these clocks contain radioactive materials, the name is meant to invoke the accuracy of a laboratory standard clock.

However, these clocks' precision relies on regular, daily reception of a radio signal from the WWVB transmitter in Fort Collins, Colo. And some folks find that no matter what they try, they can't seem to reliably receive that signal. Accordingly, the clock's time starts to drift, sometimes by as much as 10 seconds per week. For time-sensitive radio networks, that's unacceptable. So today, we'll talk a bit about how you can better improve your WWVB reception, and keep your clocks "on time."

First, you must understand that WWVB transmits in the "Low Frequency" band, specifically at 60 kHz. Like any LF transmission, it propagates much farther after dark. Ideally, you'll time your efforts to synchronize when it's at least an hour after sunset both at your location and at all locations between you and Fort Collins (it's near Denver). For areas on the East Coast of the U.S., this is a practical necessity.

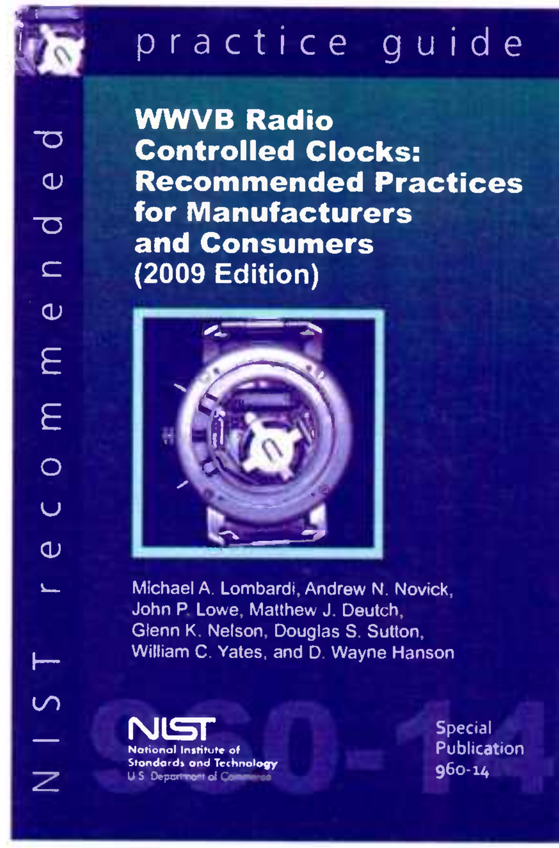
You'll want to locate the clock as far away as possible from anything that is actively drawing AC power, as it can cause RF interference to the WWVB signal. Things that cause a buzz on an AM radio are prime suspects: dimmers, computers, certain fluorescent lights, etc. This can be tricky in 24/7 operations like radio stations, and doubly so if you're not sure where the electrical outlet wiring is located in the walls you might be hanging a clock on. Experimentation is the only answer.

Inside every clock is an antenna tuned to 60 kHz, and orienting that antenna properly can improve reception. Hanging the clock on a wall or in a window that faces towards Denver should help.

If nowhere indoors is working, and you have station vehicles that are parked outside, try leaving the clocks on the dashboard every few days.

For more information and a handy signal map, check out the NIST WWVB website: *http://1.usa.gov/emdFCG*.

Aaron Read is director of engineering at Rhode Island Public Radio.



You can dowload this booklet from the NIST WWVB website.

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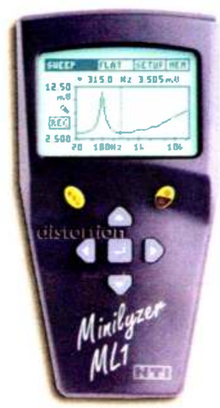
The ML1 Minilyzer is a full function high performance audio analyzer and signal monitor that fits in the palm of your hand. The comprehensive feature set includes standard measurements of level, frequency and THD+N, plus VU+ PPM meter mode, scope mode, a 1/3 octave analyzer and the ability to acquire, measure and display external response sweeps generated by a Minirator or other external generator.



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The MR2 pocket-sized analog audio generator is the successor to the legendary MR1 Minirator. It is the behind-the-scenes star of thousands of live performances, recordings and remote feeds. With its new convenient thumbwheel control system and ergonometric package design, the MR2 is the compact and economical choice for a high performance analog source.

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- Programmable Swept (chirp) and Stepped sweeps
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- AES/EBU, SPDIF, ADAT signals
- 32k to 96k digital sample rates
- Measure digital carrier level, frequency
- Status/User bits
- Event logging
- Bit statistics
- VU + PPM level meter for the embedded audio
- Monitor DA converter and headphone/speaker amp
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The DR2 Digirator not only generates digital audio in stereo & surround, it is a channel transparency and delay tester as well, all condensed into a handheld package. Delivering performance & functionality challenging any digital audio generator made today, it produces all common audio test signals with sampling frequencies up to 192 kHz and resolution up to 24 bit. The Digirator features a multi-format sync-input allowing the instrument to be synchronized to video and audio signals. In addition to standard two-channel digital audio, the DR2 can source a comprehensive set of surround signals.

- AES3, SPDIF, TOSLink, ADAT outputs
- 24 bit 2 channel digital audio up to 192 kHz SR
- Sine wave with stepped & continuous sweeps; White & Pink Noise; Polarity & Delay test signals
- Dolby D, D+, E, Pro-Logic II, DTS and DTS-HR surround signals
- Channel Transparency measurement
- I/O Delay Measurement
- Sync to AES3, DARS, word clock & video black burst
- User-generated test signal files

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follow the gigabit road

Somewhere in this, the Emerald City, there is an Intelligent Network with more than a million crosspoints connected through a 1,232 × 1,232 audio matrix shared between 21 studios and seven stations, all via AoIP running at gigabit speed. The equipment tally so far: 77 Wheatstone IP88a BLADE access units with 15 Wheatstone control surfaces, 12 crosspoint controllers, three Producer Turrets, 43 Headphone Panels, 23 Mic Control Panels and 45 Mic Processors. Still to come are at least seven more IP88 Blades, 17 mics, 17 headphone panels and 17 mic processors.

Wanna know more about it? Learn how you can benefit. Get the whole story here: **seattle.wheatstone-radio.com**

pictured: Wheatstone LX-24

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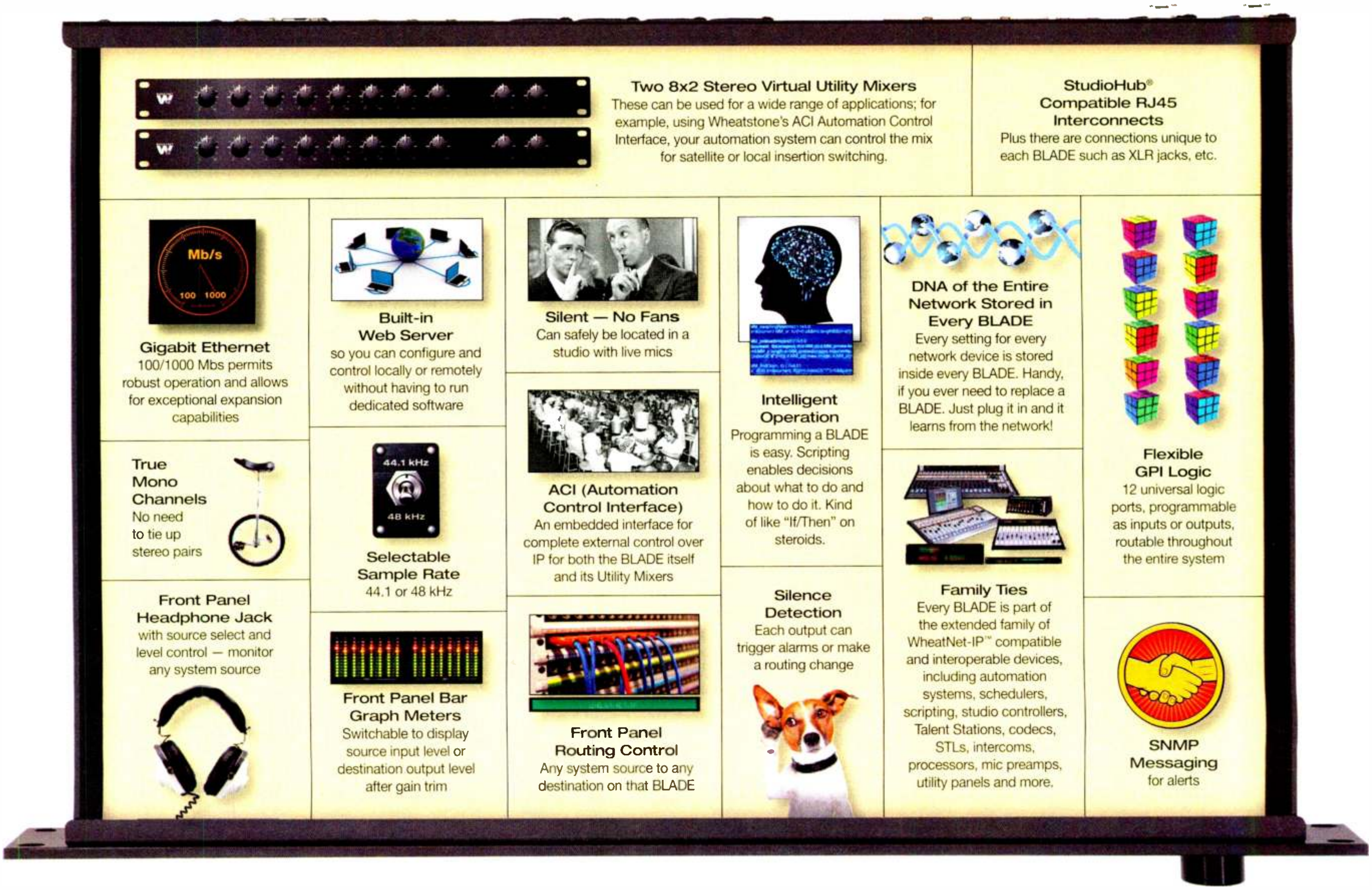


Each BLADE on the Intelligent Network is exceptionally powerful... but do you know about the incredible functionality inside *EVERY* BLADE?

By now, it's a good bet you're aware of the WheatNet-IP Intelligent Network. You know about its advantages - how it's obsessively compulsive about redundancy. How it can repair itself, configure itself, run rings around the competition while still having much greater bandwidth (due to its Gigabit Ethernet throughput) - enough, in fact, to not only handle our increased functionality today, but well into the future.

So we figured it's time to let you know a little more about those boxes you plug into the Intelligent Network. The ones that contribute to its intelligence. They're called BLADES and from inception have been far more advanced than any boxes on any other networks out there. Take a look and consider how you'd put all that to work in YOUR WheatNet-IP system.

WHEATNET-IP: THE INTELLIGENT NETWORK

Wheatstone

Make Internet Magic

A quick view behind the scenes at your website

BY STEPHEN M. POOLE

Your listeners expect your station's website to be filled with eye candy: animated graphics, "what's playing now" and more. A plain text "Welcome to WXYZ" website just won't cut it.

Station engineering staff won't normally be called upon to design and create the *content* (pages, graphics, etc.) for that website. This will be farmed out to a third party or done by someone on staff using something like Adobe's popular Dreamweaver package (*www.adobe.com/Dreamweaver*).

Therefore, I'm not going to cover content creation and Web programming here. There are tutorials online that literally will walk you through this step by step if you're interested. (One good site is *www.w3schools.com*.)

Engineering staff *will* be asked for opinions on technical issues. We're also expected to help troubleshoot the inevitable problems that crop up.

This edition of "IT stuff that an engineer should know" will take a brief look at what goes on *behind* your station's slick Web presence.

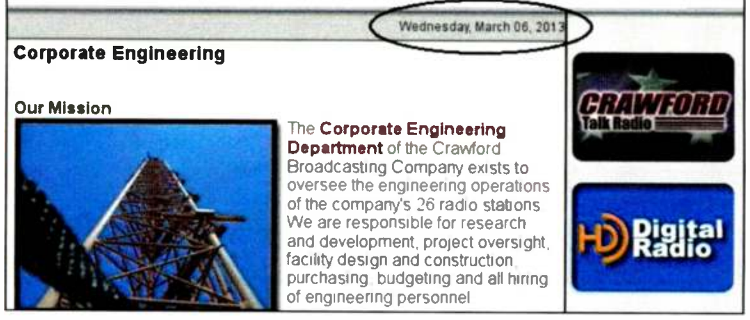
BASIC HTML

A classic HyperText Markup Language (HTML) page is a plain text file with embedded "<tags>" that tell your browser how to display it. In most browsers, you can right click in the page and select "View Source." I won't cover those tags here in any detail, but I can illustrate with an example.

Fig. 1 is the Engineering page from our company's website. The HTML source is at the top; the result is at the bottom. "<tag>" switches on an effect and "</tag>" (note the forward slash) turns it back off. For example, the "<b>" tag activates bold text and "<font color=#800000" results in red text.

Another interesting tag is "<img" near the top, which inserts an image. When the browser sees that, it will send a request to the server for the file "images/tower_small.jpg" and insert it in the display. If the file is unavailable, it will insert a stock "broken image" graphic.

```
<b>Our Mission</b><br>
<img border="0" src="images/tower_small.jpg" width="252"
</font><b><font face="Arial" color="#800000">Corporate
Engineering Department</font></b><font face="Arial"> of
the Crawford Broadcasting Company exists to oversee the
engineering operations of the company's 26 radio
stations. We are responsible for research and
development, project oversight, facility design and
construction, purchasing, budgeting and all hiring of
engineering personnel. </p>
```



HTML at top and the resultant Web page below.

The new HTML5 specification supports slick animations, but at present, most are built with Adobe's Flash. Links to these files (which usually have a ".swf" extension) can also be embedded in HTML. Your browser will fetch the Flash animation and, assuming that you have the Flash Player installed, will display it at the indicated spot in the page.

If we ignore the fancy stuff, though, basic HTML is analogous to a simple word processor. You create a document with tables, bold print and italics, then send that file to your co-workers for viewing. All of the formatting is embedded in the document, just as HTML has these embedded "<tags>."

DYNAMIC CONTENT

I've circled one other interesting feature. The date that I accessed this page was inserted by our Web server. We run a *LAMP stack* (Linux, Apache, MySQL and PHP — more on this in a moment) for stuff like this. If your Web server has a "stack," your content creators could insert the line

<?php echo date("m/d/Y") ?>

... into the file for that page and save it with a ".php" extension. Something like "03/06/2013" would be inserted into the HTML for that page, then sent to the browser. I'll have more on PHP in a moment, but this is called *server-side scripting*. This is essential for *dynamic* (i.e., changing/custom) content.

Imagine if you could automate a word processor to do custom search-and-replace just prior to printing and you'll get the idea. The operative term is "replace." Before sending that page to your browser, the Web server scans for <?php?> tags, executes the commands and then inserts the result. This is a very basic example too. When you couple server-side scripting with a *database*, all sorts of slick things are possible. If you click "View Source" in this case, you won't see the "<?php?>" stuff.

DATABASES

I covered Linux and Apache in the 2011 article "Four Steps to Build a Test Web Server" (*radioworld.com*, keyword Apache). Now for the "M" in the LAMP stack, which stands for MySQL, a popular database server.

Databases are the high-tech equivalent of the slotted shelves in your mailroom: different things get stuffed into labeled slots. The "slots" are called "fields" and you can define these any way you like.

My purpose here isn't to teach SQL; again, there are tutorials on the Web. But the people creating your website's content will almost certainly want at least one database, because they're used for all sorts of things nowadays. For example, your database could store info on advertisers, then use a script to pull up different ads for each visitor to your website.

Note that MySQL is just one of many. MariaDB, Microsoft's SQL (MS-SQL) and Postgresql are some other popular database systems. Your Web server will need to support the database that your content creators want.

CONTENT MANAGEMENT SYSTEMS

Another example is a so-called Content Management System (CMS) like Joomla or WordPress. These allow you to create and edit pages online, then "publish" them with a single click. They're popular for personal websites and blogs.

When you post a new page with a CMS, it actually stores that page (and associated content, such as graphics) in "chunks" in the database. When a browser requests that page, the CMS assembles it from "chunks" on the fly. It then sends the browser this fresh-baked HTML file for display.

What you may not know is that these CMS packages are just a collection of PHP scripts!

PHP

PHP isn't the only server-side scripting language. Microsoft's Internet Information Services (IIS) supports ASP.NET, and there are many others. But PHP is by far the most popular, which is why I'll discuss it here.

This stands for "PHP: Hypertext Preprocessor" (that's a "recursive" acronym, referring to itself in the acronym; geeks love the things). It's a powerful scripting language that runs *on the Web server* to create custom content on the fly.

I've been emphasizing "server side" for a reason. This is the counterpart of JavaScript, another popular Web scripting language that runs on the *browser side*. That's why your browser can disable JavaScript, but there's no way for you to "disable" PHP. Server-side scripting, by definition, occurs before the HTML is ever sent to your browser.

Where PHP really shines is in combination with a database ... whence the classic LAMP stack. It's a complete system for dynamic, custom Web page generation.

As I showed in that simple date example above, PHP commands can be embedded in a regular HTML file with the "<?php" tag. Some files contain nothing but PHP script(s). In either case, these are saved with the ".php" extension. A properly-configured Apache server will see that extension and give that file the "scan, execute and replace" treatment.

A complete, if simplified, example of the process is shown in Fig. 2. We've asked our content creators to set up a "current events" page on our website. They've created PHP files and have loaded the database with concerts, remotes and other station activities. When someone browses to our "events.php" page, the scripts fetch events from the database and construct an HTML page on the fly.

To repeat: My purpose in showing you this wasn't to teach you how to use PHP or set up a database. I just wanted you to understand the process so that you could make intelligent decisions if called in to help. And on that note ...

IN CLOSING: SITE HOSTING

Not many years ago, it was more cost-effective to build your own server and run it in house. It's easier and cheaper now just to use a *hosting provider*. If our company were starting from scratch today, we'd probably do this.

Companies like APlus, Dreamhost and GoDaddy have plans, complete with LAMP stack, starting at a few dollars a month. They'll register "your-station-slogan.com" with the Domain Name System (DNS). They'll maintain the server and they'll provide the bandwidth. Let them handle the headaches! All you need to do is come up with the content, upload it to their server, and you're done.

What should you look for in a hosting plan? Ask local businesses with nice-looking websites what company they deal with. Get their opinions. Next, as already mentioned, be sure that the provided database is what your content creation team needs.

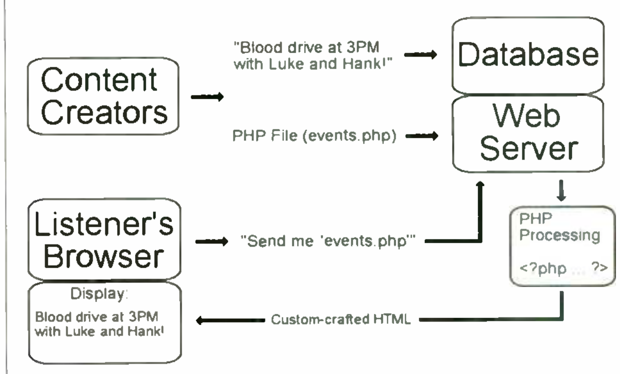


Fig. 2: A simplified example of dynamic HTML creation.

Beware of bargain-basement, $3-a-month "unlimited" plans. These are usually just a "best-effort" kind of thing with no guarantees. You're sharing that server with a bunch of other users, all competing with you for processor time and Internet bandwidth. It's better to get a *dedicated server* — simply put, a server allocated for your use only. That costs more, but it's worth it.

If you're just getting started and don't know how "big" of a plan to purchase, it's easy: Just ensure that your hosting provider will be flexible. Ask how difficult it will be to upgrade to a "bigger" plan. You'll probably find that they'll bend over backwards; after all, they want your money!

Stephen M. Poole, CBRE-AMD, CBNT, is market chief engineer at Crawford Broadcasting in Birmingham, Ala.

Multimode Fiber Provides Option for Microwave Data

A solution to Cat-5 limitations

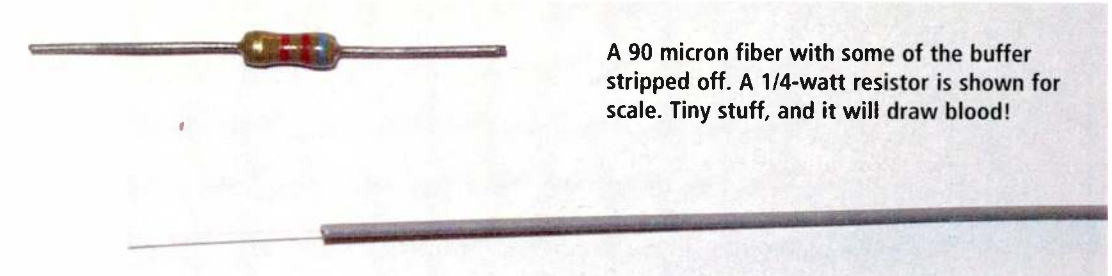
BY CRIS ALEXANDER

Since 2003 or so, I have been experimenting with alternative STL systems, and have written about some of my experiences in these pages. On a recent project, however, I took the opportunity to go a different direction with part of the link.

These "alternative STLs" have, for the most part, taken the form of some kind of 802.11 unlicensed link, a Part 101 licensed microwave system or both. All provide some quantity of Ethernet bandwidth between one end of the link and the other, and we have employed various codecs to transport audio from end to end in either or both directions.

Because they carry data, each of the links is fed with a piece of Cat-5 or Cat-6 cable. That is good in one sense because Cat-5 and Cat-6 cable is a lot cheaper than coaxial cable or waveguide, but it's bad in a couple of others because there is a 100-meter length limit on such cables, and it's tough to get them across the base insulator of a series-fed AM tower in AM station applications.

Those limitations led me to a relatively inexpensive solution: employing a short-hop 802.11 link to get from the top of the tower to the transmitter building with the data. There are many options commercially available, but my favorite is the Ubiquiti NanoBridge M5 link. It's inexpensive (less than $200 for a complete link), reliable and has a lot more bandwidth than any long-haul microwave link you're likely to use. This configuration requires power on the tower very near the microwave and 802.11 link antennas, which is probably not a big issue or expense, and it also requires a small Ethernet switch at that location.



A 90 micron fiber with some of the buffer stripped off. A 1/4-watt resistor is shown for scale. Tiny stuff, and it will draw blood!

FAILURE POINT

Over the decade or so that I have been using this kind of link, those Ethernet switches have been the number one point of failure. They don't fail often, but let's face it — in this application, they're operating in a tough environment. These simple consumer-style switches are mounted outdoors, probably in a poorly ventilated NEMA enclosure, with ambient temperatures ranging from 10 below zero to 140 above … and don't forget about the occasional magnetic field spike from a lightning strike on the tower! It's amazing they work at all!

In my new project, I was looking for a way to do away with the on-tower Ethernet switch and the 802.11 link, thus reducing possible points of failure in the system by three or so active devices. Because I was working with a series-fed AM tower, part of a 50 kW array, I had to deal with the issue of crossing the base insulator without adversely affecting the impedance of the tower. My options were few, but one kept jumping out at me: multimode fiber.

Except for the short piece of fiber in the old Continental "Power Rock" series of transmitters, my experience with fiber-optic cables was limited to gimmick Christmas decorations. I knew it was possible to pass a lot of data over a long piece of fiber, but I knew nothing of how to convert Ethernet to optical, put on connectors, make splices or any of that. It was time to do some investigating.

So I started talking to people, surfing the Web and learning all I could about it. Multimode fiber is certainly nothing new to the folks who work with data. I found some of my amateur radio friends who regularly use it as a transmission medium within large data infrastructures. As such, there is a lot of information on fiber-optics out there. That's the good part. The bad is that none of the information fit my specific application.

That's where we engineers get to do our thing — figure out how to apply different technologies to meet the needs of particular situations.

AN EDUCATION IN FIBER-OPTICS

The microwave radios I planned to use were made by Dragonwave, and a brand-new option at the time was for multimode fiber inputs and outputs. It was so new, in fact, that evidently none had yet been shipped.

Dragonwave provided me with the information I needed — the type of fiber, connectors, etc. required; and with that in hand, I began searching for equipment that I could use on the other end of the piece of fiber to get the blinky red light back to Ethernet ones and zeros.

That search turned out to be short. Cisco and other manufacturers of high-end gigabit switches offer units with GBIC (gigabit interface converter) ports, and for under $40, there are any number of GBIC fiber-optic transceivers available. Buy one, plug it into the switch's GBIC port and you've got yourself a gigabit switch with a fiber port on it.

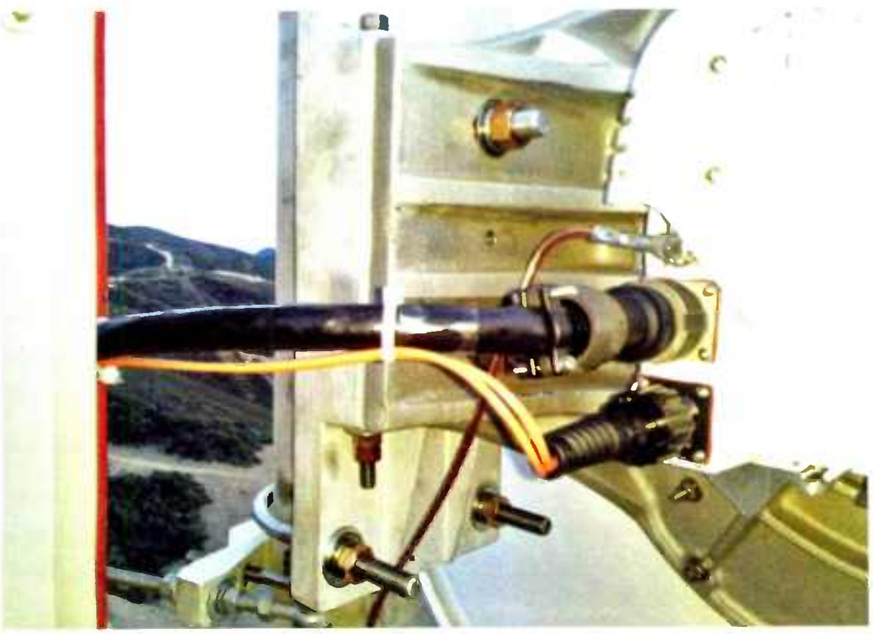
The next part of my education came in learning how to put connectors on multimode fiber. That was quite a learning curve. On the recommendation of a friend who works with fiber, I purchased a Corning fiber-optic connector installation toolkit, something that set me back some $1,400. It included all kinds of tools used to work with fiber, the connector jig, crimp tool and continuity tester with several different connector adaptors.

I had to learn first to be very careful, very gentle. Fiber is incredibly robust, but nick it and it's extremely brittle. Removing the "buffer" (what you and I would think of as "insulation") from a 90-micron fiber is a delicate operation that can easily end with a broken fiber and ruined installation.

I also had to learn that everything had to be very clean. You clean the bare fiber with a special cloth and cleaning solution before putting on the connector. And you don't cut fiber, you "cleave" it, essentially scoring it at the desired cut point and then breaking it. Weird.

Once I got the hang of it, though, putting on the connectors wasn't so hard. My third or fourth effort worked (the connectors I wrecked were $12 each!). For a test, I used a pair of Cisco switches with GBIC fiber transceivers to connect my laptop to the office network.


Contents of the Corning fiber-optic toolkit.


Rear of Dragonwave 11 GHz transceiver showing power (upper) and fiber (lower) connections.

I was amazed when I was able to connect to the Internet and pass data over that piece of fiber! But that experiment proved to me that it would work, and it showed me that I could successfully put the connectors on.

IT ALWAYS PAYS TO TEST FIRST

I had the Dragonwave transceivers shipped to my office in Colorado and my next step was to configure them and connect them with fiber to the LAN.

I got that done and tested, and then programmed in the frequencies to the Dragonwaves. Next, I ran the power all the way down and actually pushed data through the microwave link and fiber, just to make sure it would work (it did). Better to confirm all that on the ground than try and troubleshoot it 300 feet in the air!

So I shipped the Dragonwave transceivers, power supplies and a 1,000-foot roll of duplex multimode 90 micron fiber-optic cable to California for the installation. The tower crew installed the antenna, a four-foot dish, on the tower along with a NEMA enclosure for the –48V power supply for the radio. The enclosure provided a pass-through for the fiber-optic cable to the radio.

My initial plan was to use a solid run of duplex multimode fiber from the transmitter building all the way to the top of the tower, and indeed, that would be the ideal way to install it. But after much thought and discussions with the tower crew, we decided it would be too difficult to pull a cable with delicate connectors installed in a piece of 3/4-inch rigid conduit *up* a 300-foot tower. If a connector were damaged, it would all have to come back out — there's no way I could install a connector hanging from a harness at the top of a tower!

The plan, then, changed a bit. I opted to terminate the fiber-optic cable from the transmitter building (some 500 feet away) inside the ATU where it would be spliced to the cable coming down the tower. The conduit from the transmitter building terminated into the bottom of the ATU cabinet, so that part was easy. For the run from the tower, I set a 6x6x4 PVC J-box right on the transom plate of the tower, terminated the vertical

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RECEIVER

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transmitter infrastructure. Such enhancements include, but are not limited to, improved decoding of System Control Data Sequences (SCDS), adaptive and iterative channel and noise/interference estimation in collaboration with soft-input soft-output (SISO) Log-MAP decoding of convolutional codes, list Log-MAP decoding of convolutional codes for protocol data unit (PDU) bits, soft decoding of Reed-Solomon code and iterative decoding between Reed-Solomon and convolutional codes.

Numerical results of simulations are provided to justify the merits of the proposed enhancements, which are shown to be about a 7 dB improvement in reception for audio packets in hybrid FM HD Radio.

We also present even more significant improvements that could be achieved by modified HD Radio air interfaces for systems that have not been deployed yet, such as all-digital AM and all-digital FM HD Radio. These improvements are achieved by means of novel nonsystematic irregular repeat-accumulate (IRA) code that approaches the Shannon capacity and also exhibits superior performance in the presence of impulsive noise/interference.

In addition to improved performance, the use of new code can also simplify the physical layer as only one FEC code may be employed instead of multiple codes for different logical channels.

The rest of this paper is organized as follows. The second section provides brief descriptions on the processing techniques devised for performance improvement in the current HDR receivers and numerical results in exemplary cases. In the third section, a new non-systematic IRA code is presented along with its simulation results.

Improvements for Existing Systems

I. DECODING OF SYSTEM CONTROL DATA SEQUENCE

Decoding of the SCDS bits is important for the overall system performance in HDR systems. For the processing at the receiver, in the current system, the differential phase shift keying (DPSK)-modulated SCDS bits are first non-coherently decoded and then a majority voting follows on the repeated bits to make a final bit decision of all repeated bits collectively. In this paper, we explore the benefits of integrating log-likelihood ratio (LLR)-based iterative decoding techniques into the decoding of SCDS bits to improve the accuracy of SCDS-bit decisions and also the quality of channel state information (CSI).

The main improvement in SCDS is based on "soft" non-coherent maximum likelihood diversity combining of appropriate bits from different reference subcarriers. In addition, multiple symbol observation interval based differential decoding yields improvements in many scenarios. Also, the parity bits in the SCDS sequence are used for soft single parity check code decoding, such that the least reliable bit in a field protected by the parity bit is flipped.

In this paper, we present advanced receiver techniques that achieve improved performance of HD Radio and change HD Radio dynamics such that HDR receivers can operate at much lower signal-to-noise ratio (SNR).

II. ADAPTIVE AND ITERATIVE CHANNEL ESTIMATION

In conventional HD Radio receivers, CSI estimation (including the noise power estimation) over HDR's OFDM spectrum band of 1093 subcarriers utilizes pilot symbols carried on a limited number of reference subcarriers. Thus, use of interpolation techniques is essential and FIR smoothing filters are often applied to reduce the effect of additive noise on the CSI estimation. However, the performance of the conventional methods is inherently limited when the FIR smoothing filters do not take into account channel variations in time, and interpolation is not sufficiently accurate to track rapid channel variations, particularly in fast time-varying channels.

In the proposed advanced receiver processing, channel selectivities in both the time and frequency domain are estimated based on the level crossing rate (LCR) and the lengths of FIR smoothing filters are dynamically adapted to the channel variations based on the estimated values of LCR.

Moreover, adaptive decision-directed channel estimation (DDCE) based on data symbols is applied selectively, depending on the SNR and channel selectivity conditions. With these methods integrated into the receiver processing, particularly, in the initial estimation (thus called the advanced initial CSI estimation), the accuracy of initial CSI estimation is enhanced and the additional iterations of key receiver processing including further refinement of CSI estimates within the iterative processing result in improved decoding performance.

Note that unlike in the advanced initial CSI estimation, in the subsequent iterative processing, adaptive DDCE is not employed, as estimated "soft" symbols (based on the coded-bit log-likelihood ratios obtained for the convolutional log-MAP decoder) serve the purpose.

For the iterative processing, the coded-bit LLRs produced by the SISO/Log-MAP decoders for decoding of convolution codes are utilized. Coded-bit LLRs of all logical channel signals in HDR (e.g., P1, PIDS, P3 and P4 for hybrid FM) are properly interleaved and multiplexed in the receiver in the same way as processed in the transmitter. Then, the composite LLRs are supplied to advanced CSI estimation and subsequent SISO/Log-MAP decoding for the next iteration. There are multiple alternatives to implement joint iterative decoding and CSI estimation, and the number of iterations depends on the stopping criterion.

III. SOFT IN/SOFT OUT LIST DECODING OF CONVOLUTIONAL CODES

For decoding of protocol data unit (PDU) bits of the Transport layer in HDR, the conventional HDR receivers typically rely on Viterbi decoders to decode convolutional coded bits and produce hard-decision (e.g., logical bit 1 or 0) information bits of the Layer 1 (L1) logical channel frames. Then, these L1 information bits are segmented into smaller blocks depending on the format of the logical channel.

For logical channel P1, for instance, the decoded L1 information bits would be segmented into a main program service (MPS) PDU header, a program

(continued on page 19)

service data (PSD) PDU, and multiple audio packets for further processing to reverse the processing done in the transmitter to form the PDUs and logical channel frames. For audio, packets of these segmented L1 information bits are passed to a conventional CRC decoder for error detection, and then to a source audio decoder. For data packets, L1 information bits are passed to an algebraic Reed-Solomon (RS) decoder, possibly enhanced with erasure decoding, which produces hard decision bits, followed by a conventional CRC decoder for error detection. Each decoding is performed separately and in a sequential manner. The algebraic Reed-Solomon decoding on hard bit-decisions from the Viterbi decoder results in suboptimum performance and such an approach is not amenable to potential improvements in performance with iterative decoding.

In the advanced receiver, however, L1 coded-bit LLRs from the last iteration of CSI estimation are first segmented into blocks depending on the format of the logical channel, and a list Log-MAP decoder is employed to produce, from each block of coded-bit LLRs, information-bit LLRs and a predefined number, denoted by *M_value*, of most-likely hard-decoded bit sequences that correspond to the information bits in the block. These information-bit LLRs and the list of sequences are further processed in a way that can improve the ultimate accuracy of information bit decisions.

For audio packets, once the list Log-MAP outputs the information-bit LLRs, bit decisions are made in reference to a threshold value of zero for antipodal signals, and the CRC-syndrome of the packet is checked for packet error. If the packet passes the CRC syndrome check, the bit decisions are declared as the final decisions of the information bits in the audio packet. If the CRC-syndrome check on the packet fails, the bit sequences in the list are checked for the CRC syndrome in the order of most-likely correct sequences. The sequence that first passes the CRC syndrome check is declared as the decoded packet. If all sequences fail the CRC syndrome check, the first sequence in the list is used as the decoded packet.

IV. SOFT DECODING OF REED-SOLOMON CODES

Advanced SISO decoding of Reed-Solomon (RS) codes is employed for both MPS PDU header and for advanced data services. The SISO RS decoder is based on belief propagation and includes several novel aspects and achieves best known SISO RS decoder performance with much smaller computational complexity that the next best known soft input RS decoder in the literature.

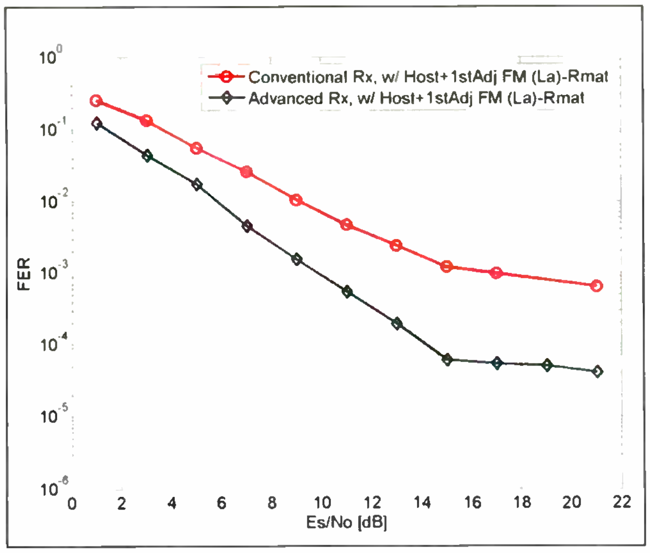

Fig. 1: Audio FER — USLOW2, Digital at –10 dB, With First Adjacent

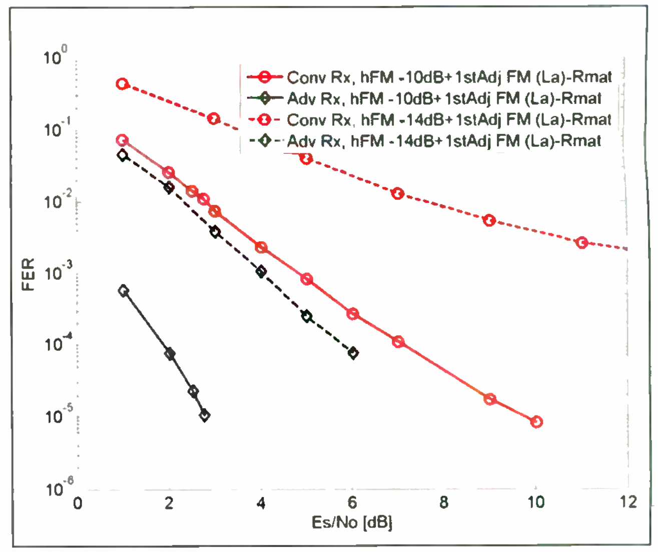

Fig. 2: Audio FER — UFAST60, Digital at –10 dB and –14 dB, With First Adjacent

V. NUMERICAL RESULTS

Performance of the proposed advanced receivers is evaluated via computer simulation in terms of audio packet frame error rate (FER), or equivalently block error rate (BLER). In this simulation, the digital, OFDM signal power is assumed to be at –10 dB or –14 dB relative to the host FM analog signal.

The first adjacent channel interference is assumed to be present at –20 dB relative to the host FM signal. Three different types of time-varying, multipath fading channels are considered, i.e., slow urban fading channel with a vehicle speed of 2 km per hour (USLOW2), fast urban fading channel with a vehicle speed of 60 km per hour (UFAST60), and a 3-path fading model (3RAYS).

For decoding audio packets, the advanced receiver employs the advanced initial CSI estimation and one additional iterative CSI estimation stage after the first forward error correction (FEC) decoding of P1 frames employing tail-biting Log-MAP decoding. The proposed Rmatrix decoding is used before the CSI estimation. After iterative CSI estimation on L1 frames, tail-biting list Log-MAP decoding is employed for audio packets. The list Log-MAP decoder uses a list size, *M_value*, of 32. On the other hand, the conventional receiver processing includes a single stage CSI estimation using time- and frequency-domain filter lengths suitable for the range of mobile speeds and frequency selectivity of the channel. For the conventional method, the MATLAB tail-biting Viterbi decoder provides essentially the same performance as the tail-biting Log-MAP decoder. In both receivers, metric calculation of LLRs is implemented in the form of a linear clipper, suitable for Laplacian noise, to account for impulsiveness of host FM interference that exhibits approximately Laplacian distribution. Such a metric ensures more accuracy of performance evaluation in practical systems than a LLR metric meant for AWGN.

Fig. 1 shows the performance of the audio packet of P1 logical channel in USLOW2 corresponding to very low mobility conditions. It can be seen that the advanced receiver is about 6 dB improved at FER = 10^{-3} in symbol energy to noise power ratio (Es/No). Eventually, both receivers exhibit error floor performance, but the error floor for the advanced receiver based on the proposed processing techniques is close to the target performance FER = 10^{-5} desired for commercial operation. Fig. 2 shows the performance of the audio packet of P1 logical channel in UFAST60. The advanced receiver with the proposed approach achieves about 7 dB gain in Es/No at the FER = 10^{-5} desired for commercial operation at –10 dB digital (i.e., solid line with circle and solid line with diamond, respectively), and more than 8 dB at FER = $2x10^{-3}$ when the digital signal is at –14 dB (i.e., dashed line with circle and dashed

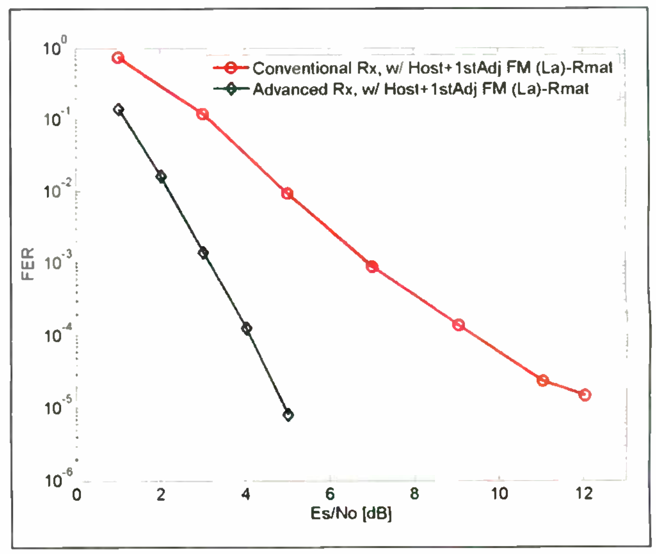

Fig. 3: Audio FER — 3RAYS, Digital at –10 dB, With First Adjacent

line with diamond, respectively). Thus, in the presence of an increased amount of analog FM interference, the performance gain of the advanced receiver is larger. Fig. 3 illustrates the performance of the audio packet of P1 logical channel in 3RAYS at a vehicle speed of 100 km per hour. The advanced receiver achieves about 7 dB gain in Es/No at FER = 10^{-5}.

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Fig. 4 illustrates the computer-simulated FERs of P3 AAS data of the FM HD Radio receivers in UFAST60. It can be seen that at FER = 10^{-4}, the advanced receiver achieves a gain of more than 1.5 dB relative to the conventional receiver. Also, with additional multiple iterations between outer RS decoder and inner convolutional Log-MAP (or list Log-MAP) decoder, the performance is further improved as shown in the figure (dashed line with square).

Finally, Fig. 5 shows the performance of the audio component of P1 logical channel in the same scenario as in Fig. 2 but with additional effect of shadowing with standard deviation of 8 dB considered. It can be seen that in the presence of shadowing, the gain of advanced receiver over conventional receiver is comparable to, and at some FERs even bigger than, the gain in the case with no shadowing, e.g., more than 10 dB in Es/No at FER = 10^{-5}.

Improvements for New HDR All-Digital Standards

I. NON-SYSTEMATIC CHECK IRREGULAR IRA CODE

IRA codes are a class of LDPC codes that feature lower encoding complexity than general LDPC codes, with comparable error rate performance. It is commonly perceived that capacity achieving codes (e.g., turbo, LDPC and IRA) would need to be systematic, in which the information bits are transmitted over the channel together with the coded or parity bits. The ratio of the number of information to the number of parity bits depends on the coding rate (R). In non-systematic codes, information bits are not transmitted but only the coded bits. Until recently there has been a scarcity of work on non-systematic capacity-achieving codes.

A typical non-systematic IRA encoder is shown in Fig 6. The encoder is composed of a set of bit repeater nodes of different degrees, collectively referred to as irregular repeater that repeats the information bits sequence u in an irregular fashion. For example, a bit repeater of degree m produces m identical replicas of an information bit. These repeated information bits represent the first-stage coded bits. Then, an interleaver performs pseudo-random permutations on the repeated bits sequence v, and produces interleaved first-stage coded bits v'. A set of check-node combiners of different degrees, collectively referred to as check nodes operate on the interleaver output v' to produce the second-stage coded bits, check bits sequence c. A check-node combiner of degree n performs modulo-2 addition of n inputs bits represented in the {0,1} domain. A check node of degree 1 is a by-pass check node that simply passes the input bit to the output. The second-stage coded bits are processed by an accumulator, also known as a differential encoder, producing the third-stage coded bits, coded bits sequence a.

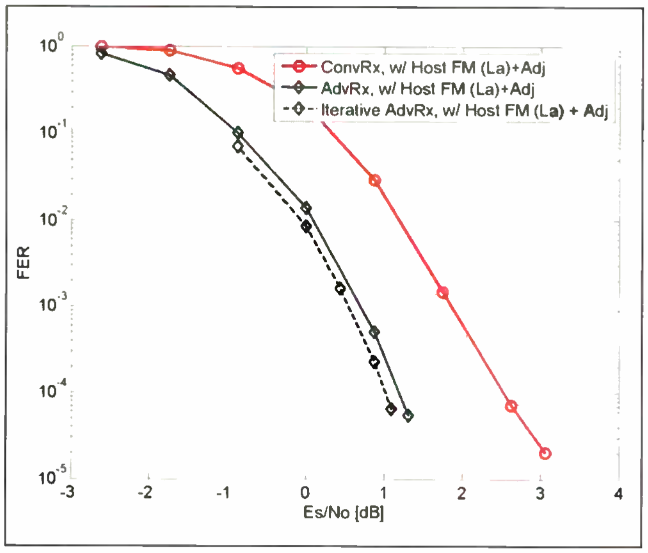

Fig. 4: P3 AAS Data Packet FER — UFAST60, Digital at −10 dB, With First Adjacent

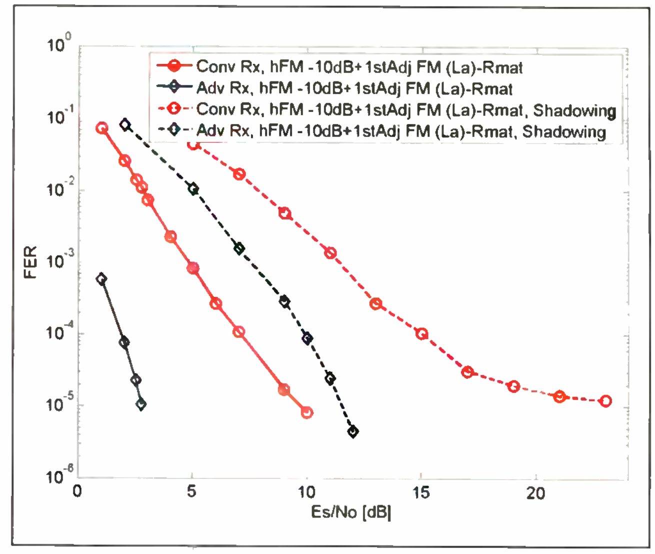

Fig. 5: Audio FER — UFAST60 With Shadowing, Digital at −10 dB, With First Adjacent

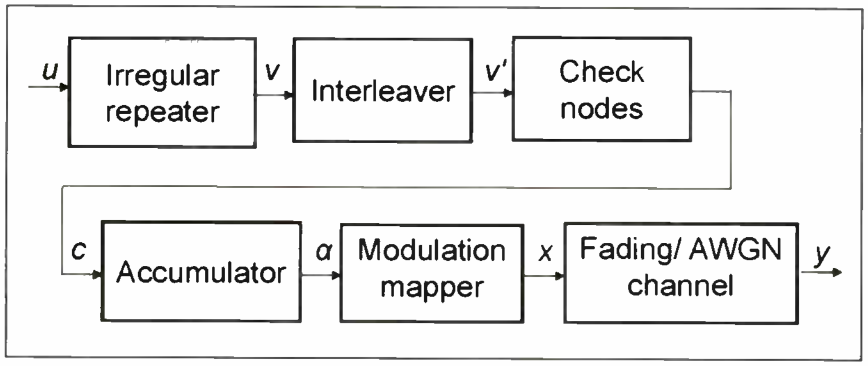

Fig. 6: IRA Encoder Block Diagram

The third-stage coded bits are transformed into modulation symbols using a desired modulation mapping, such as BPSK, QPSK, M-QAM or other desired modulation mappings, producing modulation symbols x to be transmitted over a channel. For simplicity, an equivalent baseband model is considered here, omitting steps such as carrier modulation, power amplifications and other steps as known in the literature. The channel may include additive white Gaussian noise (AWGN), multiplicative fading, other forms of multipath fading and possible impulse interference. Finally, the sequence y is the received baseband signal, including the transmitted symbols x distorted by various channel impairments mentioned above.

The non-systematic check-irregular IRA codes provide extra flexibility in designing a variety of desired coding rates by having the freedom to vary both the bit and check-nodes degrees. The use of non-systematic IRA code can simplify the physical layer of all-digital AM and all-digital FM HD Radio as only one FEC code may be employed instead of multiple codes for different logical channels. In this case, the information bits input of non-systematic IRA encoder can represent information bits coming from one or more logical channels and at least one of audio and data. With this, the larger the information bits input of non-systematic IRA code the better BER/FER performance is obtained.

II. NUMERICAL RESULTS

Numerical results for performance are provided for two kinds of codes, i.e. non-systematic IRA against nonsystematic convolutional, and for different types of channels, showing the advantages of non-systematic IRA codes in terms of achievable error rate (BER/FER) performance at smaller SNR values.

Fig 7 shows performance comparison of non-systematic IRA code against non-systematic convolutional code with coding rate $R = 5/12$ and information length equal to 30,000 bits, assuming BPSK modulation, independent Rayleigh fading with known fading amplitudes and noise power at the receiver and when: (i) both complementary codes (two sidebands) are received; and (ii) one complementary code (one sideband) is completely lost due to channel impairments. A non-systematic convolutional code of constraint length $K = 9$ is used with generator polynomials, i.e., g1 = 561, g2 = 657 and g3 = 711. For this code, the puncturing patterns to obtain coding rates $R = 5/12$ and $R = 5/6$ were used. As shown in the figure, the performance gain of the non-systematic IRA code (with 50 decoding iterations) against non-systematic convolutional code is more than 5 dB in bit energy to noise power ratio (Eb/No) when both sidebands are received (i.e., solid line with circle and dashed line with square, respectively) and more than 10 dB in Eb/No when only one sideband is received (i.e., solid line with upper triangle and dashed line with diamond, respectively) at FER below 10^{-3}, and the gains are larger at lower values of FER.

Similar to Fig. 7, performance comparison of nonsystematic IRA code against non-systematic convolutional code is depicted in Fig. 8 but assuming 64-QAM modulation and with Laplacian noise. As shown in the figure, the performance gain of the non-systematic IRA code against non-systematic convolutional code is more than 10 dB in Eb/No when both sidebands are received and more than 8 dB when only one sideband is received, at FER of about 10^{-4} and the gains are larger at lower values of FER. For the results shown in Fig. 8 and non-systematic IRA code, 5 demapping operations (i.e., initial and additional four IRA decoding operations) were performed.

Concluding Remarks

We have presented receiver processing techniques and simulation results that demonstrate substantial improvement in the performance of HD Radio receivers. These enhancements only require very doable changes in the baseband receiver chip in radio receivers, but no changes in existing HD Radio

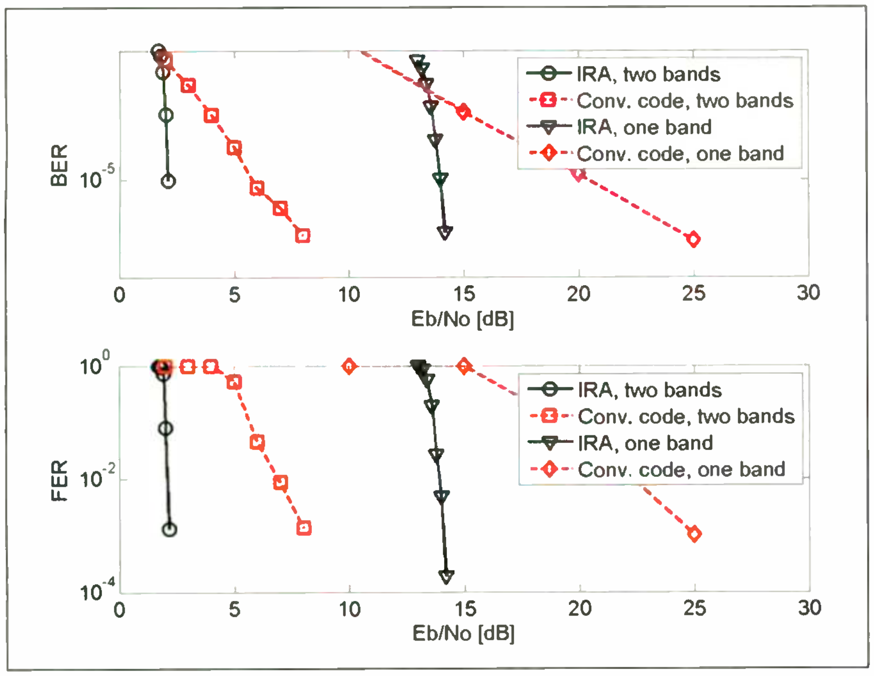

Fig. 7: Non-Systematic IRA vs. Convolutional Code Performance Comparison, R = 5/12, 30,000 Bits Frame, BPSK Modulation, Independent Rayleigh Fading Channel

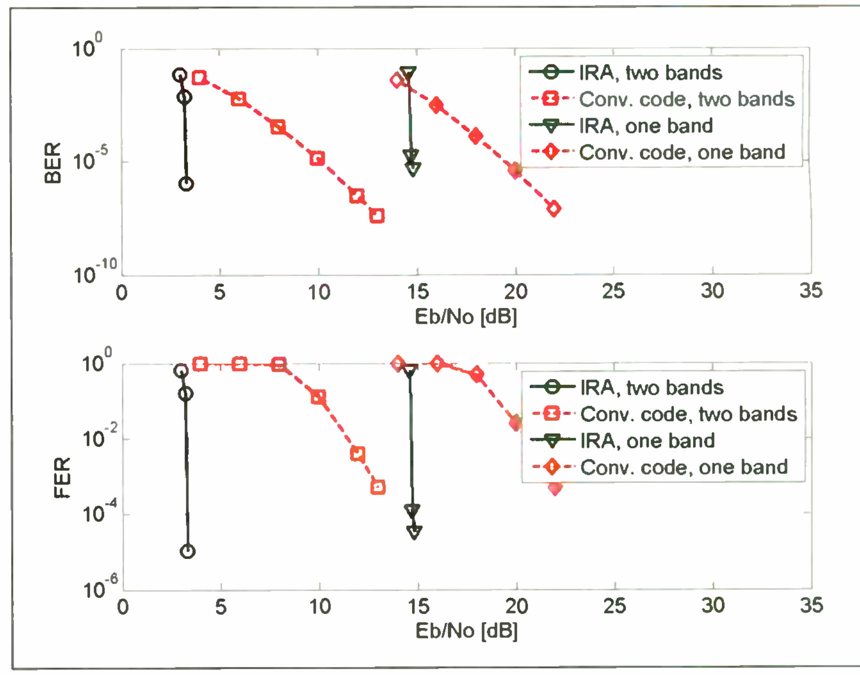

Fig. 8: Non-Systematic IRA vs. Convolutional Code Performance Comparison, R = 5/12, 30,000 Bits Frame, 64-QAM Modulation, Laplacian Noise Channel

transmission format or infrastructure. It is expected that the demonstrated enhancement techniques will enable HD Radio receivers to operate at much lower signal-to-noise ratios. This in turn will enable broadcasters to maximize coverage and performance, and thus customer satisfaction, and ultimately help international adoption of the HDR standards. We have also demonstrated potential significant improvements in performance of all-digital HD Radio receivers that may employ novel non-systematic check-irregular IRA codes.

This white paper is based on a paper presented at the 2013 NAB Broadcast Engineering Conference.

Dr. Branimir R. Vojcic is a professor at George Washington University in Washington. Dr. Hakan Dogan is an assistant professor at Istanbul University in Turkey. Dr. Wookwon Lee is an associate professor at Gannon University in Erie, Pa. Dr. Stylianos Papaharalabos is a post-doctoral fellow at the Institute for Space Applications and Remote Sensing, National Observatory of Athens in Greece. Dr. Farnaz Shayegh is a post-doctoral fellow at McGill University in Montreal.

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FIBER

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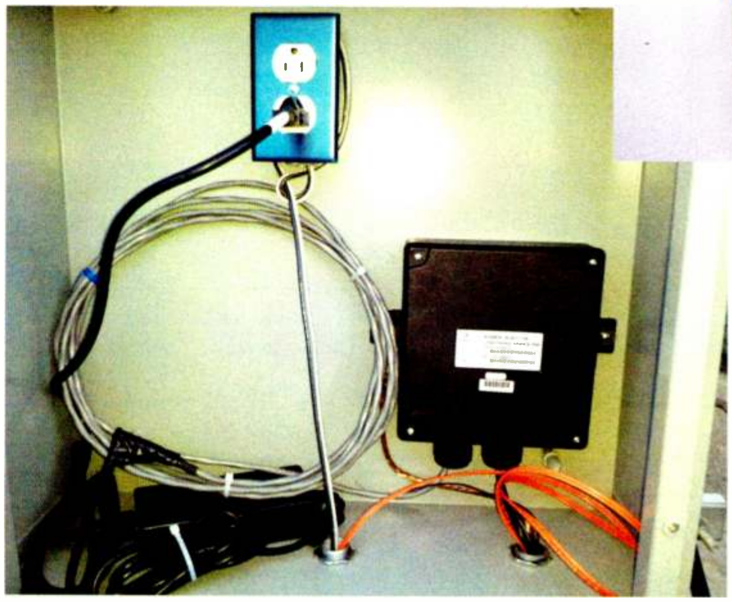
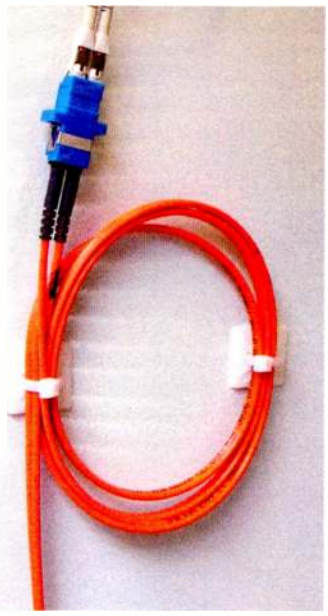
conduit run from the top into that box, then ran a piece of Seal-Tight to the lighting control box on the tower to pick up 120 VAC.

To cross the base insulator with the fiber, I used a length of 100 percent PVC non-metallic Seal-Tight, connecting one end into the J-box on the transom plate and the other end into the underside of the ATU cabinet. I had some reservations about this (there would be 15 kW or so across that base insulator and thus across the piece of Seal-Tight), and I expressed them to Bobby Cox of Kintronic Laboratories. Bobby replied that KTL has been using PVC to manufacture isocouplers for years with no issue. That made sense, so I stopped worrying. So far so good with all the RF!

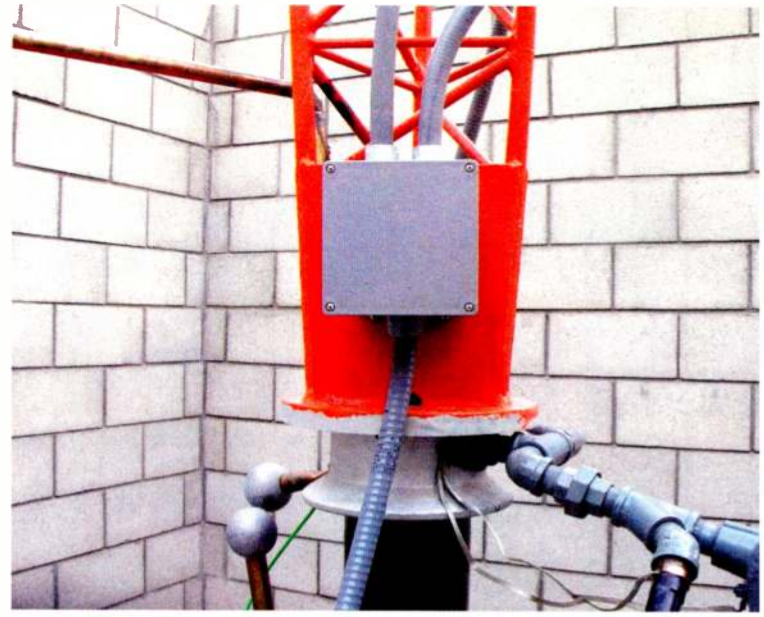
With the splice made inside the ATU, I applied power to the Dragonwave transceiver on the tower, and just like that, the GBIC port lit up with a valid gigabit connection and we were immediately able to push management data to the transceiver. In short order we had the path aligned with the predicted signal strength at each end, and we were pushing data between studio and transmitter in both directions.



Interior of the NEMA enclosure on the tower containing the surge suppressor, –48V power supply and loop-through for the fiber. The –48V power supply employs a coaxial-type connector. The coaxial m-f connection was secured with electrical tape by the tower rigger to keep it from vibrating apart.



Fiber-optic cable splice employing LC-connectors and a dual coupler inside the ATU. The duplex cable, with buffer and outer jackets, resembles zip cord.



PVC J-Box on transom plate. The conduit coming in on the top left is from the NEMA box on top of the tower and contains both 120 VAC and fiber-optic cables. The top right conduit carries the 120 VAC power conductors to the lighting control box on the tower. The PVC conduit on the bottom contains the fiber cable and crosses the base insulator to the ATU.

PROBLEMS SOLVED

The link now carries STL audio, remote control, security, video surveillance and telephone to and from the site. I sure feel a lot better without an Ethernet switch on the tower and an unlicensed RF link between tower and transmitter building. The removal of those additional possible points of failure changes everything.

Going forward, I plan to make fiber a primary design element in my AM and FM STL links. It solves both the base insulator and length limit problems with Cat-5/6. And besides that, I think it's pretty cool.

Cris Alexander is director of engineering at Crawford Broadcasting, a longtime RW contributor and a past recipient of SBE's Broadcast Engineer of the Year award.

READER'S FORUM

STL ANTENNAS — TOO MUCH HEIGHT?

Regarding the article "STL Antennas — Too Much Height?" in the Feb. 20 edition of RWEE:

Another factor is interference. You may be able to better cooperate with other licensees, reducing or eliminating interference concerns by reducing the height of your STL antennas.

Dave Barnett
Nevada City, Calif.

MORE ON STL HEIGHT

Jeffrey Gehman correctly points out the problem of reflection of microwave (and even VHF/UHF) STL paths. Lowering the antenna may help in some cases, but in my many years of microwave communication link experience, that may not be the best solution.

First of all, those "helpful trees" that he shows blocking the reflection have a nasty habit of growing unexpectedly taller and infringing upon the Fresnel clearance zone. Secondly, what happens if those trees are cut down by property owners, or if not of the evergreen variety, lose their leaves in the fall?

Finally, radio wave propagation isn't limited to simple and straight line-of-sight paths. Particularly on longer hops, climatic changes of weather and temperature fronts will drastically alter the apparent point of reflection due to "curvature" of the wave path, so it is likely the hop will still experience degraded signal strength at times.

A better solution if budget permits is to mount *two* antennas at the receive end, separated vertically on the tower, and both fed into a diversity reception detector that monitors the quality of signal from each path and selects the best signal to be demodulated.

A space diversity system like this can compensate for variable propagation characteristics much better than simply lowering one antenna. As with any RF link, careful design of the antenna mounting locations and spacing is still required, but for paths that need the highest reliability, space diversity (or the somewhat related frequency diversity) systems are preferred.

Leon Amstutz, CBRTE
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Upland, Ind.