ENGINEERING EXTRA

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NewBay Media

ONE, TWO, THREE, FOUR!
Stephen Rutherford and H. Hawaii Media find that HD4 in stereo is possible. Page 14

Kevin Gross Talks About AES-X192

The society's effort seeks a high-performance streaming AoIP interoperability standard

Digital audio routing, switching and control have grown dramatically in broadcast facilities. TDM-based (time-division multiplex) systems were the first to appear in the 1990s. But in recent years, audio-over-IP, or AoIP, systems have gained widespread popularity.

NEWSMAKER

Unfortunately, all these systems use unique protocol features and proprietary components, making interoperability among the different platforms and brands difficult, if not impossible. As Radio World reported in its Aug. 1 issue, the Audio Engineering Society is stepping up to tackle the challenge of standardization so that interoperability might be achieved.

The AES Standards Committee has initiated a project called AES-X192, with the goal of developing a high-performance streaming audio-over-IP interoperability standard. The issues involved are reminiscent of when AES became involved in the serial digital audio standards setting process that eventually produced AES3 (AES/EBU) in 1985.

AES-X192 was conceived two years ago and is being led by Kevin Gross, a media network consultant with AVA Networks, which developed the CobraNet standard. About 100 members including most of the major console and router manufacturers are taking part in this project.

Radio World Technical Advisor Tom McGinley interviewed Gross via email to glean more background and how the effort is progressing.

Kevin, you've been interested in this topic for a number of years and now have the AES involved pushing for a formal new AES industry standard for interoperability. Tell us about your background and current work as a media consultant, and how X192 got started.

I've been working in audio networking since the mid-1990s when we came up with CobraNet at Peak Audio. CobraNet is now owned by Cirrus Logic, and I'm no longer involved with it. I have continued as an independent consultant helping end users, system designers and equipment manufacturers with media and networking technology. In this role, I keep abreast of the IT industry in general and media networking specifically.

From that position, I noticed two things. First, the IT industry has been rallying around the Internet Protocol (IP) as a framework for all things networking. Second, there has been an emergence of a new generation of media networking protocols. Unlike previous generations, the protocols in this new generation use similar technical concepts and standards. Like previous generations, these new protocols do not interoperate.

I recognized an interoperability opportunity between these technically similar protocols and approached the AES with the idea of developing an interoperability standard.

Describe how the AES Standards Committee approaches and conducts the standards setting process.

Standards development at the AES is conducted in a very open manner. Membership in task groups that develop standards is open to any "materially affected individual." This basically means that standards are developed by individuals, not companies, and the participants have a stake in the outcome. The task groups are given great autonomy in how they choose to organize themselves and conduct business. This autonomy and openness has allowed for quick progress on X192.

What is the proposed schedule for meetings of the X192 group and when do you estimate that a new standard could likely be finalized and emerge for the pro audio and broadcast industry?

We have a draft standard and have been holding teleconferences twice a month to edit that to everyone's satisfaction. We are nearing the end of that process and are working to have a final draft approved by the task group later this year. There are several further AES approval steps beyond that before it is published as a standard. "X192" is the AES designation for the project. The X192 work product will be published as an AES standard named AESxx, where xx is some number to be determined.

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FROM THE TECH EDITOR

iStockphoto/Alex Slobodkin

BY MICHAEL LECLAIR

There are some readers who have noticed the change regarding the FCC's public file requirements. Television stations in the top 50 Designated Market Areas, and stations affiliated with the top four national television broadcast networks, must begin to upload their public files to an FCC maintained website. All other television stations must comply within two years.

This change will produce unprecedented public access to information about television operations. TV stations have been readying compliance to this new requirement since details were published in April, following a long legal battle.

But don't turn the page and walk away just because you're radio. While the new rules only affect television immediately, a close reading of the commission's reasoning would suggest that the rest of us broadcasters be prepared to follow along in our turn.

EXTENDED ARGUMENTS

The history behind this dramatic change came to public light as part of a Notice of Proposed Rulemaking in October of 2000.

In what must have been a fairly radical proposal at the time, the requirement that TV stations place their public files online was adopted in 2007 in a Report and Order. This proposal was challenged by many, including the NAB and Walt Disney Corp. After the issue was stuck in court challenges for several years, the FCC issued a Further Notice of Proposed Rulemaking that vacated the 2007 Report and Order entirely and set up the requirements for online public files being implemented today.

While some may find these public announcements from the FCC to be a bit dull, I have to admit that often I'm intrigued; they afford an internal look at the philosophy of governing and industry regulation straight from the most important players in our industry.

As a public process, all affected parties are invited to comment either for or against proposed rule changes. In making a final order, the FCC must address all public comments and explain why it may accept or reject a particular reasoning. It is this process of proposing, accepting comments and finally deciding on a set of rules that is at the heart of how the FCC regulates our industry.

The commission goes to great pains to explain why it feels public file materials should be public in ways that make television station operations more transparent than any time in previous history. Its reasoning is that by putting the public file online, the information can be accessed by virtually anyone over the public Internet at no expense. This is, in the end, the only way to truly make public files an instrument of full and transparent disclosure. For those of us who are used to having complete control over access to the public file, and may indeed have used public file requests in the past as an early warning system of possible license challenge, this is indeed a new world order.

Reasons given to justify an online public file form the heart of the Second Report and Order. For example in paragraph 12 there is this quote: "...making public file information available through the Internet should facilitate public access and foster increased public participation in the licensing process." Or in paragraph 13: "We also agree with commenters that access to the public files has been inconveniently (and unnecessarily) limited by current procedures."

But in particular it is interesting to note the discussion about a station's political file, which includes a record of all political advertising sold and who was the purchaser. The commission specifically points out the legal precedent of the *Citizens United vs. Federal Election Commission* case, in which the Supreme Court ruled that political advertising was a form of free speech and thus could not be restricted.

The FCC responded that at the very least, information about political advertising would become as transparent and publicly available as possible to provide an essential support to the democratic process.

Given the strength of the FCC's commitment to transparency, it seems only a question of time before all broadcasters, including radio stations that also benefit from political advertising, will be brought into this ruling. That said, the FCC has only hinted at the possibility so far and has no timetable for requiring radio stations to move their public files online.

For a more detailed view of this discussion you can read the Report and Order yourself. I've posted the link at *radioworld.com/links*.

IT'S NOT ALL BAD

For many in the radio industry I'm sure the first reaction to such a requirement will be anger and denial, similar to what occurred amongst television broadcasters.

The commission has tried to address some of the more practical objections and minimize the burden on broadcasters. Most importantly, the FCC will take on the burden and expense of maintaining the file server system that is used to make these materials available online. It has taken pains to accept just about any standard format possible for uploading information without the requirement of reformatting them to the commission's

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THIS ISSUE

OCTOBER 17, 2012

Come Here Mr. Watson, CBNE!

Sometimes the expression 'lost in transmission' really means that

BY CHARLES S. FITCH

SBE certification is the mark of professionalism in broadcast engineering. To help you get in the exam frame of mind, Radio World Engineering Extra poses typical questions in this column. Although similar in style and content to the exam questions, these are not from past exams nor will they be on future exams in this exact form. Today's question is at right.

Normally our questions arise from the combined genius of your erudite RW Engineering Extra editor and your humble author, mirroring readers' requests to sojourn into certification topic areas that interest or perplex them.

This month is unique in that we asked the SBE certification folks for a typical CBNE question in honor of this new level of engineering certification. The Certified Broadcast Networking Engineer level exam focuses on computer networking skills, now an essential component of broadcast engineering.

At first blush I chose one of the wrong answers, (c). Obviously a nick in one of the lines is not good and jeopardizes the reliability of the cable and connection but it is not the most correct answer.


Tools of the trade

The Mismatch Patch

Question from the Aug. 22 issue
(Exam level: CBNE)

A cabling impedance mismatch can be caused by which of the following?

a. Using RJ-45 connectors on Cat-6 cable
b. Running cable in an overhead cable tray near electrical conductors
c. Nicking cable conductors when stripping
d. Mixing shielded and unshielded twisted pair cable in the same segment
e. Not following ITE/EIA/IEEE standard UTP-101 regards wire twist format (i.e. left overhand)

This question was incorrectly listed as a CBNT level question in the last issue of Engineering Extra. The correct level for this question should be for Certified Broadcast Networking Engineer, or CBNE.

CLASSIC BUC BOGUS

Let's move expeditiously through the other wrong answers.

As anyone who has installed a computer to a network knows, the RJ-45 connector mentioned in answer (a) is the most ordinary eight-pin connector used on

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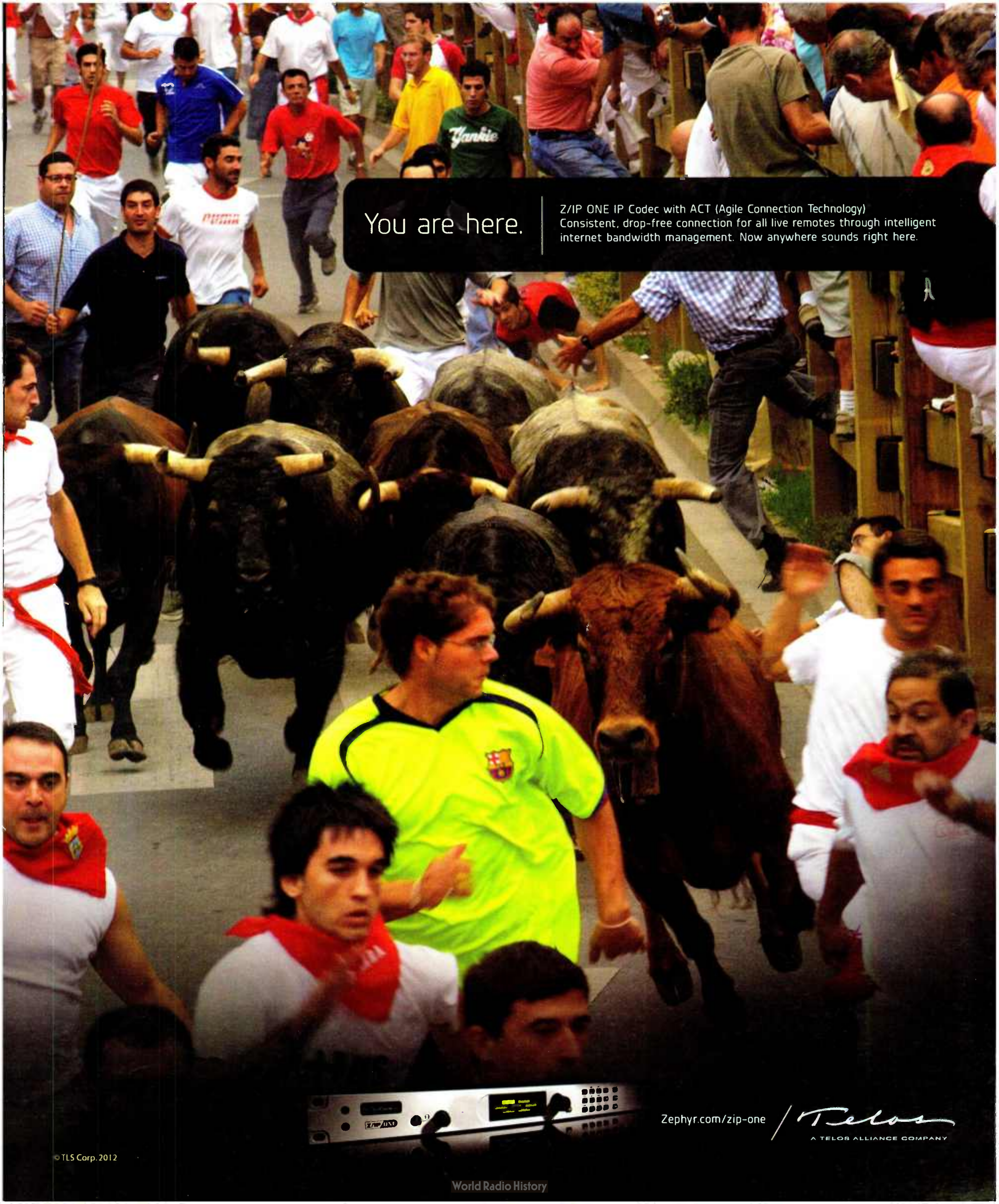

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CBNE

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Cat-6 cable and thus should not be source of impedance mismatch. Technically, a poor installation of the RJ-45 could cause this problem but that isn't what was written.

Data cable is optimized for common mode rejection; if properly made up and connected, it is just about impervious to induction fields caused by nearby electrical cables. The fields might mess with the data, but I cannot see any connection to an impedance mismatch. Endless miles of this cable coexist in cable trays throughout the world so (b) is not our answer.

Although it reads with the patina of technical veracity, answer (e) is a classic Buc Bogus answer. "VUFB" is an inside joke amongst my confreres, an acronym for Victor's Ultra-luxurious Flat Black spray paint, which we use liberally around here to cover rust and wear recklessly.

This makes our most correct answer (d), mixing shielded and unshielded twisted pair cable in the same segment.

By way of reminder, actual SBE exams usually have only four choices. We use five to allow an enlarged discussion and makes it a little more challenging (as well as allowing us an opportunity for a little humor on occasion).

A HISTORY OF REACTANCE

Our devoted RWEE readers know that a large part of our work is researching technical points for our articles, sometimes very fine ones. In doing that, I often am amazed at how early in history certain aspects of our technology appeared.

For a moment, let's look back to circa 1880 and Alexander Graham Bell's seminal work on the telephone.

Voice information is transferred to a varying current and then reconverted to voice again at the end of the circuit. Not exactly digital data, but nevertheless the effort to send this information a great distance brought forward the problem of reactance on the lines. The voice varying current, even though very low frequency, behaved over long distances as alternating current and reactance began to affect the signal.

Shunt capacitance is a major attenuator of varying current, and one early solution was to cancel that with peaking coils. If there is enough interest, we can return to this telco phenomenon in a future column, but we mention this now to point out that when conveying audio or digital information over long distances, reactance must be dealt with.

LIMITATIONS

Cable reactance sets the ultimate bandwidth (highest frequency of data possible and the maximum distances this bandwidth can be maintained). As capacitive reactance increases, it will tend to shunt high-frequency signals between conductors or to ground. This ultimately limits the highest frequency data pulse that can be transmitted. Capacitance between conductors also creates what is known as crosstalk, wherein data pulses moving in opposite directions mix together and become unusable due to corruption. Category cables (e.g., Category 5 or 6) are designed to minimize these effects.

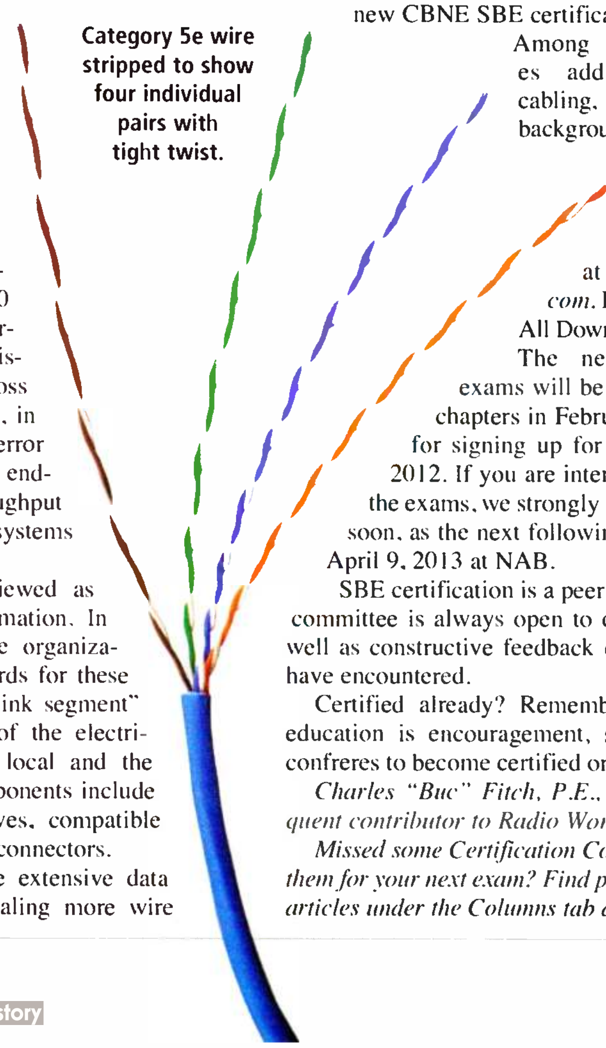
In this day and age of advanced electronics, it is interesting that the simple mechanical structure of category wiring is its power. By running digital signals on individual, balanced transmit and receive pairs, the cable connects easily between different areas in a building without creating ground loops. And by the simple addition of twisting the paired wires, category cables reduce the potential for inductive coupling or crosstalk. The more twists, the better the crosstalk performance.

Current wiring standards call for a minimum of Cat-5e or Cat-6 cabling. Each of these is capable of passing data at a rate of 1 gigabit per second when installed properly.

Cat-6 cabling has a nominal design impedance of 100 ohms. Mixing cables of different impedances will cause a mismatch and increase return loss (essentially data VSWR). This, in turn, will create a higher bit error rate (BER), which will result in endless data re-sends, lower throughput and a tendency for networked systems to crash.

Data cabling should be viewed as a channel, a conduit of information. In the parlance of the IEEE, the organization that formalizes the standards for these systems, what we have is a "link segment" — the physical components of the electrical data connection between local and the remote equipment. Those components include the main cable runs themselves, compatible jumper/connection cables, and connectors.

Today's radio stations have extensive data cable systems, quite often totaling more wire than the analog connections. The important work of installation and maintenance of these data wiring systems is part of the job description of station and contract engineers, so knowledge in this area is important. Once again, recognition of that knowledge is most often indicated in our industry by the CBNT and the new CBNE SBE certification.



Category 5e wire stripped to show four individual pairs with tight twist.

Among many Web resources addressing Ethernet/data cabling, an easy comprehensive backgrounder on cable installs is a white paper from Wheatstone, "The Role of Wiring Integrity ..." available at *audioartsengineering.com*. Look under Support, then All Downloads.

The next SBE certification exams will be given in the local SBE chapters in February 2013. Closing date for signing up for the exams is Dec. 31, 2012. If you are interested and ready to take the exams, we strongly suggest that you sign up soon, as the next following exams are scheduled April 9, 2013 at NAB.

SBE certification is a peer process. The 14-person committee is always open to question suggestions as well as constructive feedback on exam questions you have encountered.

Certified already? Remember that nine-tenths of education is encouragement, so motivate your own confreres to become certified or advance a grade today.

Charles "Buc" Fitch, P.E., CPBE, AMD, is a frequent contributor to Radio World.

Missed some Certification Corners or want to review them for your next exam? Find past Certification Corner articles under the Columns tab at radioworld.com.

A Question of Peak Performance

Question for next time

(Exam level: CRBE)

What is the mathematical relationship of the peak voltage and the RMS voltage value of a uniform triangular waveform?

a. The peak value and the RMS value in volts are numerically equal

b. The peak value is the sine value and the RMS value is the cosine value of the square root of 12 times Vp to p in volts

c. The RMS voltage value is the peak voltage value divided by the square root of 3

d. $\text{RMS E} = \sum_{k=0}^{n} \binom{n}{k} Vp^{n-k}$

where n is the number of cycles considered (normally based on a nominal 1 second or the waveform frequency) and k is Boozeman's constant (after Victor Boozeman) normally rounded to 1.57 or $\pi/2$

e. RMS E = Vp times the square root t divided by T where T is the duration of the waveform (nominally ¼ of the cycle) and T is the total count of cycles or the frequency of the waveform

PUBLIC FILE ONLINE

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requirements; for example, the most common document formats such as Microsoft Word or Excel are acceptable. Additionally, any materials that are available online from the FCC itself are not required to be included in the online public file, such as coverage maps, applications for license, ownership reports or copies of the Public and Broadcasting manual. Letters and correspondence to the station are exempt for privacy reasons, but must still be kept as they are today in a correspondence file at the station.

Given the reality that an online public file is in the first stages of implementation it would seem a wise choice to review that aspect of your station operations with an eye to preparing for the change.

But don't be in too much of a hurry. Indeed, the FCC ruled that some NCE FM stations associated with public TV stations that had voluntarily requested a chance to participate in the online public file would not be permitted to do so. Their reasoning: If too many stations try to join in at once it may overwhelm the process. That decision doesn't seem to offer much hope that radio stations will remain forever exempt — only that when they get the process figured out and operating smoothly it will be extended to everyone.

As always, your comments on this and any other radio engineering topic are welcome. Does this sound like the end of the world to you or a long delayed step forward? Let me know at *rwee@nbmedia.com.*

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(continued from page 1)

Telos/Axia got the AoIP ball rolling first for console and routing systems. Other manufacturers like Wheatstone and Logitek are now offering their own AoIP solutions. Briefly explain the major similarities and the differences within the various AoIP platforms.

Here's a summary at right of AoIP protocols from the Objectives and Benefits document you can download from *www.x192.org* [find the direct URL at *radioworld.com/links*]. The similarities are use of User Datagram Protocol or Real-time Transport Protocol for transport and IEEE 1588 for synchronization. But it is possible for two systems to use the same protocol in different, non-interoperable ways. The other differences are in management protocols used for discovering other devices on a network and setting up a connection.

Technology	Purveyor	Date Introduced	Synchronization	Transport
Ravenna *http://ravenna.alcnetworx.com/*	ALC NetworX	In development	IEEE 1588-2008	RTP
AVB *www.ieee802.org/1/pages/avbridges.html*	IEEE, AVnu	2011	IEEE 1588-2008 advanced profile (IEEE 802.1AS)	Ethernet, RTP
Q-LAN *http://tinyurl.com/QLANWP*	QSC Audio	2009	IEEE 1588-2002	UDP
N/ACIP	EBU	2007	Data packet arrival times	RTP
Dante *www.audinate.com/*	Audinate	2006	IEEE 1588-2002	UDP
Wheatnet-IP *www.wheatip.com/*	Wheatstone	2005	Proprietary	RTP
LiveWire *www.axiaaudio.com/livewire/*	Telos/Axia	2004	Proprietary	RTP

Table 1: Summary of AoIP Protocols

U.S. broadcasters are familiar with Telos/Axia and Wheatstone, but the others listed on your chart are less familiar to many of us. Can you tell us a little more about those?

Ravenna is most well known in the broadcast industry in Europe. It is a technology developed by one of the Lawo companies, but available to other manufacturers for a reasonable fee.

AVB is a set of IEEE standards jointly developed by the pro audio and networking industries. The automotive industry is now also contributing to development. Special AVB network equipment is required to make this work.

Q-LAN is the networking piece of QSC's integrated Q-Sys signal processing and distribution system. This is currently used mostly in commercial audio systems though there are some prominent broadcast applications (e.g., U.S. Senate recording studio)

Dante was developed by a network startup in Australia. The technology is licensed to audio equipment manufacturers. Dante is most commonly found in commercial and live applications.

To what extent will details of the proprietary features have to be revealed for AES to be successful in crafting a new standard with X192?

One of the ground rules we've used in X192 development is to reference existing standards wherever possible, rather than invent new techniques and protocols. I believe this approach reduces exposure to intellectual property issues. You never eliminate the risk entirely, however.

Broadcast engineers have hoped that generic, off-the-shelf Ethernet switches could be employed in AoIP systems. However the switches now being used have to be qualified, managed switches and how they are configured is critical. Why is this issue of switch implementation so fussy and confusing?

An important piece of a successful AoIP system is network quality of service features. QoS standards and technology have existed in the network world for years but have not, until now, been given a real workout. Network vendors are learning from fussiness experienced by their customers and the situation is rapidly improving. Anyone building an AVB switch has to explicitly deal with these issues, for instance.

I was involved in the development of early systems for recording direct to hard disks. That was also initially quite fussy. Media networking feels like it is on a similar technology curve. We're not going to remember any of this in a few short years.

In some cases, only a specific manufacturer's switch is compatible. Won't there need to be some kind of standardization of switch architecture, firmware and programming established as part of any X192 standard?

One of the stated goals of X192 is to work with standard IP network equipment. I have tested a fair amount of network equipment for QSC, and my general findings are that some of the equipment (even otherwise high-performance equipment) does not give the real-time performance or offer some of the QoS features required by real-time media networking. As I say, this situation is improving quickly.

Is it realistic to expect that after an AES-X192 standard is established, engineers might eventually be able to buy a generic $50 switch from Best Buy and run a multi-studio broadcast facility through it?

If you're going to build a small audio-only network, we're already there. Surprisingly, the cheap switches often have the best raw performance. That's because everything is done on one IC. Things have to slow down to go between ICs in the larger and more complicated equipment. If you want to mix audio with

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other IT functions, you will need QoS and that generally requires a managed switch. Fortunately, almost all gigabit switches you can buy today are managed and have the QoS features. They're not exactly $50, but the Cisco Small Business switches, for instance, get the job done for less than $1,000.

Most of the U.S.-based AoIP systems use the same Cisco or HP Ethernet managed switch models. What other switches are being used?

Just about anyone who builds a gigabit Ethernet switch has a switch that has the required QoS feature set. Examples can be found from all these manufacturers: Dell, Cisco Small Business (formerly Linksys) Extreme Networks, Juniper, Brocade, Netgear, 3Com, Alaxala, Allied Telesys, Avaya, Moxa and Hirshmann.

Beyond the IP network itself, the other major hardware link of any AoIP system is how source devices get connected to the network. To what extent will X192 set any protocols or rules for these components in the new standard?

X192 describes how audio data is formatted into network packets and how everything is synchronized. Additionally, X192 specifies protocols that allow devices to find one another on the network and then exchange commands to set up the aforementioned audio transmissions across the network.

Other aspects of configuring these boxes such as preamp gain settings, logic inputs and contact closure outputs are outside the scope of X192.

I don't know whether X192 makes these network interface products commodity items. I don't think it is a question of great import because where I see the real value of interoperability is where, instead of using a second piece of gear to interface a product to the network, the network interface is built into audio products such as automation equipment, processors, consoles, amplifiers, microphones, speakers and transmitters.

Synchronization is a vital element within AoIP. Some systems are using the IEEE 1588-2002/8 standard while others are proprietary. Will X192 likely specify the IEEE standard?

Standards development rules of participation discourage me from confirming or denying this to the press, but, yes, it is very likely. :)

Aside from the hardware, what other factors and considerations within an AoIP system complicate the goal of forging an interoperable standard?

I think business issues are really the important complicating factors. There are business models out there that rely on proprietary systems. I think many of these would work better for everyone if opened up, but that can be a scary transition to make.

Axia/Livewire is collaborating with Lawo/Ravenna from Germany and appears to be leading the effort to leverage an open standards approach for AoIP interoperability. Axia's Mike Dosch has even said that he would expect any AES-X192 standard to look very much like "the new Livewire."

What is your position on this development and how might it affect the X192 process?

Axia and ALC (Lawo/Ravenna) have been very active in the development of X192. It is clear to me that open interoperability is a business priority for both of them. With X192 we're going to make interoperability available to the industry. Anyone who sees the value in it can pick it up and run with it. We appear to have a game going with Axia and ALC even before the standard has been finished. To me, that's a good sign that X192 will achieve the critical mass of acceptance required to make real and useful interoperability.

Compared to the AES3 standard setting process and result, what similarities and differences do you see with the X192 process as it proceeds?

AES3 was first published in 1985, so I can't make a first-hand comparison. I have encountered some sour attitudes about the standards process and results. What is important to understand about AES standards is that there's a lot of autonomy given to the individual groups developing standards, and so there's no single process. Because of the generosity of my sponsors, Axia and QSC, I'm able to devote a significant amount of my time leading the effort, researching basic technology, drafting the standard and interfacing with related standards bodies. I believe this level of attention has helped move things forward relatively quickly.

While full interoperability of the major systems and protocols would be an ideal goal of X192 as a new AES standard, in the end it may come up short of establishing or requiring that for all participating systems and specific protocols. Expand on that as a possible outcome.

I think what you're getting at is described here [see comic below]. I'm not going to claim that I have a surefire way around this pitfall, but we are not trying to solve everyone's problems with X192. Full interoperability is not the goal. We're trying to define a fairly basic interoperability mode. We expect manufacturers to go beyond that and develop their own extensions. If X192 is successful, maybe we'll have the opportunity go back and standardize some of those extensions.

What major players have not yet stepped up to be part of the X192 effort?

There are over 100 individual members of the task group. Members come from networking, silicon, broadcast, live sound, commercial audio, music production and other fields. End users, system designers, integrators and equipment manufacturers are represented. X192 is a technical effort, but there has not been a lot of publicity yet (publicity doesn't usually help get technical work done), so I'm sure there are some that are simply not aware of the initiative.

Is there the expectation that similar to AES serial, there will be two standards developed ... one high-performance interface for pro audio and a cheaper one for consumer systems?

We're including in the standard several options in terms of latency, bit resolution and sample rate. It is not feasible to require all devices to implement all modes so I expect AoIP products will self-classify themselves in the market based on the capabilities they implement. Because of the protocols we're using, X192 will have a certain amount of intrinsic interoperability with VoIP systems. A VoIP device could be dialed up to minimally support X192.

Tom McGinley is technical advisor and longtime contributor to Radio World.

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OMNIAAUDIO.COM/GameChanger
Omnia
192.168.2.199
Fault: None
Config: <none>
Preset: CGSmooth
Input: Livewire 1
Pri
Sec
Input
AGC
ST
Wide Band
AGC
Limiters
FM
B
M
HD
S
L
FM
HD
Output
Analysis
NESS MONITOR
7.5 kHz
Preset
Wizard
Basic
Advanced
SL
LO
ML
XO
MH
HI
SH
Lim Thresh
0.0 dB
SL to LO Sync
-12.0 dB
Attack Offset
8.3
Release Offset
5.6
Gate Offset
-4.0 dB
11
Now Available
in FM-Only and
FM+HD Versions

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.

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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

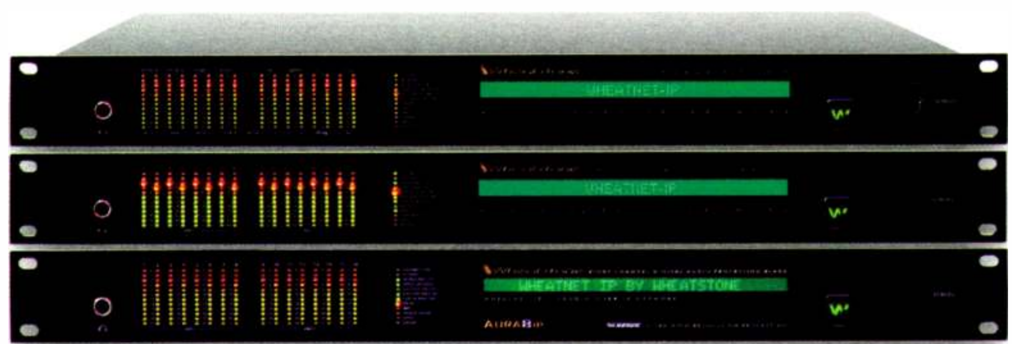
Onboard VGA and USB-Mouse connectors

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.

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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.

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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

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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

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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

HD4 Rollout With a Three-Translator Chaser

HD Radio plus translators are used to support three unique formats in Hawaii market

BY STEPHEN RUTHERFORD

The phone call from George Hochman of H. Hawaii Media was not unlike most other calls from prospective clients. He needed a new studio built out and the modification of two existing program rooms to accommodate two foreign language formats. These two new formats and an existing jazz format would be fed to three FM translators, currently not operational. The interesting part of the project was transmitting all three on the "yet to be installed" HD on the KORL(FM) signal.

The project was originally started in February of 2011, and three previous engineers had been involved in some portion of the design and acquisition of the equipment. Some unnecessary equipment had been acquired and, as expected, necessary equipment overlooked. Overall system documentation was sketchy and only the discussed "concepts" of how the system would work were available.

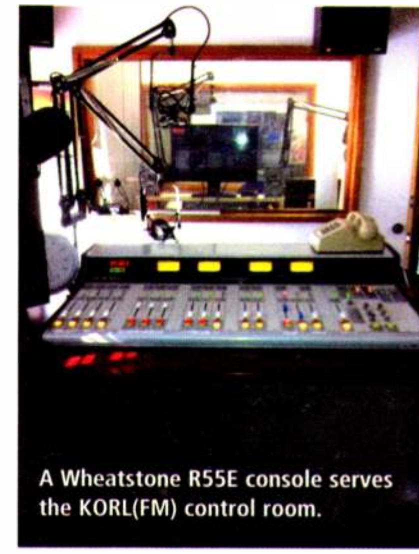
GET THE STUDIOS READY FOR NEW HD

The new studio centered on the Arrakis ARC-10U console, a great choice to accommodate the majority of the unbalanced consumer-level audio equipment to be installed. The Program bus was configured to feed the KORL HD3 sub-channel while the Audition bus output was configured for backup production needs.

The studio cabinetry remained unchanged. After three days wiring the studio was commissioned to provide a +4 dBm output to the central equipment racks for the future HD encoding.

The existing program rooms consisted of the original KORL(FM) control room, the KHPI(AM) control room and the existing production room. The KORL(FM) control room console, with a Wheatstone R-55E, was configured to feed the KORL analog signal from the Program bus output, with the KORL(FM) HD2 sub-channel to be fed from the Audition bus output.

The new Arrakis console is mounted in the studio.



A Wheatstone R55E console serves the KORL(FM) control room.

The production room console was configured to feed the KORL HD4 sub-channel from the Program bus output while the Audition bus remained for the production needs of the facility.

Three new BSI Simian automation systems were configured at the BSI facility and only required minor set-

(continued on page 16)

THE NEW MATRIX

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- Data & setup memory storage and recall on micro SD card

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Analog Audio Analyzer

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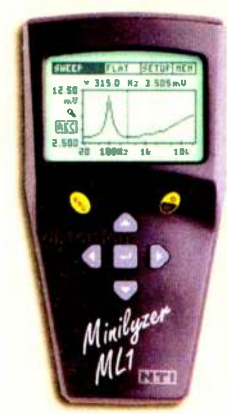
Add the optional MiniLINK USB computer interface and Windows-based software and you may store all tests on the instrument for download to your PC, as well as send commands and display real time results to and from the analyzer.



- Measure Level, Frequency, Polarity
- Automatic THD+N and individual harmonic measurements k2 – k5
- VU + PPM meter/monitor
- 1/3 octave analyzer
- Requires optional MiniSPL microphone for SPL & acoustic RTA measurements
- Frequency/time sweeps
- Scope mode
- Measure signal balance error
- Selectable units for level measurements

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High performance Analog Audio Generator + Impedance/Phantom/Cable measurements

The MR-PRO Minirator is the senior partner to the MR2 below, with added features and higher performance. Both generators feature an ergonomic instrument package & operation, balanced and unbalanced outputs, and a full range of signals.

- High (+18 dBu) output level & <-96 dB residual THD
- Low distortion sine waves
- Programmable stepped or glide sweep
- Pink & white noise
- Polarity & delay test signals
- User-generated custom test signals & generator setups
- Impedance measurement & speaker power calculation
- Impedance measurement of the connected device
- Phantom power voltage measurement
- Cable tester and signal balance measurement
- Protective shock jacket

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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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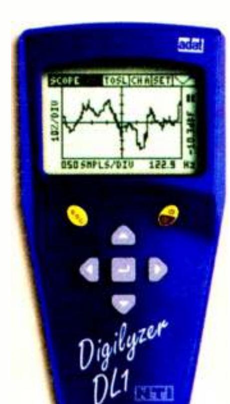
- New higher output level (+8 dBu) & low distortion
- Programmable Swept (chirp) and Stepped sweeps
- Sine waves
- Pink & White noise
- Polarity & Delay test signals
- Illuminated Mute button
- Set outputs in Volts, dBu or dBV
- Hi-res backlit display
- Balanced and unbalanced outputs

DL1 Digilyzer
Digital Audio Analyzer



A handheld digital audio analyzer with the measurement power & functions of more expensive instruments, the DL1 Digilyzer analyzes and measures both the digital carrier signal (AES/EBU, SPDIF or ADAT) as well as embedded digital audio. In addition, the DL1 functions as a smart monitor and digital level meter for tracking down signals around the studio. Plugged into either an analog or digital signal line, it automatically detects and measures digital signals or informs if you connect to an analog line. In addition to customary audio, carrier and status bit measurements, the DL1 also includes a comprehensive event logging capability.

- AES/EBU, SPDIF, ADAT signals
- 32k to 96k digital sample rates
- Measure digital carrier level, frequency
- Status/User bits
- Event logging
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- 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
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H. HAWAII MEDIA

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ting changes to adapt the system to the facility.

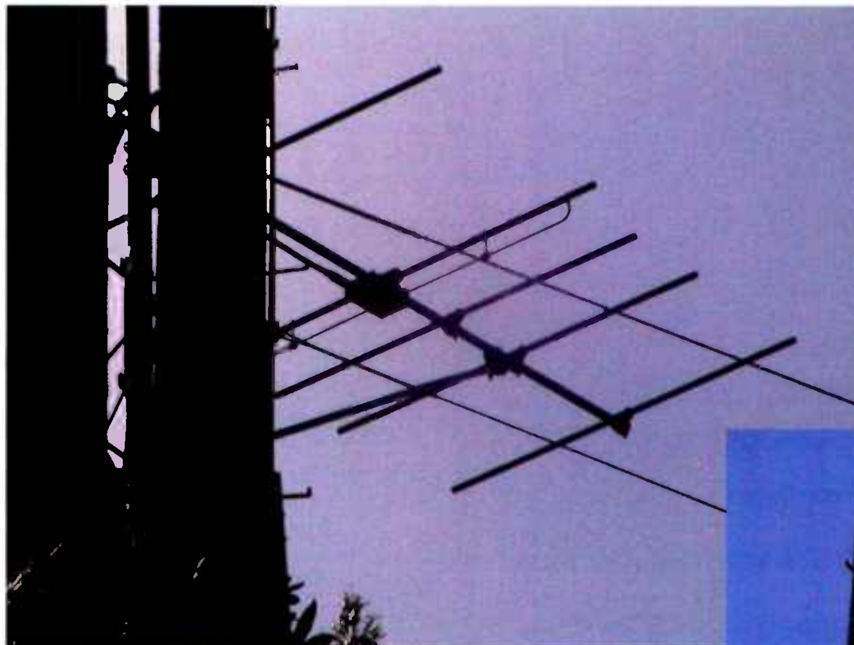
FIX THE TRANSLATORS FIRST

The triplexed FM translator site (three transmitters sharing a common antenna) is located on Mount Tantalus and was not operational for several reasons. First, the Bext model TFC2K four-bay antenna system was originally installed with the top bay just above the surrounding tree and bamboo growth, and the bottom two bays were bisected by two of the supporting 3/8-inch galvanized guy wires to stabilize the mounting pole.

Mr. Hochman engaged the rigging services of Ross Putnam to rehang the four-bay antenna system to a higher position on the antenna mast as well as run a new feed line into the translator equipment shelter. After rehanging the main antenna, Mr. Putnam also installed the Kathrein-Scala five-element yagi antenna that would feed the FM translator receivers.

When the rigging operations were completed, Mr. Putnam conducted VSWR sweeps at the feed-line input, using his Anritsu Site Master, which indicated a VSWR of 1.68:1 at the 101.5 MHz frequency.

We discussed this situation with Luca Bononato, the chief engineer for Bext in San Diego. He recommended testing the individual antenna elements with each of the three translator transmitters to determine their match quality. The next was to substitute a known good power divider for the existing unit to ensure no issues with the matching transformer.



A yagi receive antenna is used to feed the three FM tuners, which in turn feed the HD2, HD3 and HD4 subchannels.

This Bext four-bay antenna is used to combine the three translator signals. The antenna needed remounting and some repairs before it could be used for the project.

The Production Room would also be used to relay one of the HD sub-channels on its program bus, with production work being done in Audition.

Bext shipped out the following materials: Four new inner-bay coaxial cables, because it appeared that the EIA flanges were not installed properly on the original inner bay coaxial cables; a 16-foot antenna test cable to run from the end of the feed line EIA flange to test the VSWR of each antenna element; and a replacement matching transformer to ensure the antenna system components could be verified and replaced as needed in the field. All H. Hawaii Media had to do was return the original components and cover the shipping expense. Customer Service makes all the difference in the world when you're in the field.

From the individual element testing, we found the Bay No. 3 was contributing a higher VSWR as compared to the other three bays on the 101.5 MHz translator. This bay was replaced with a same model spare, however this was not the primary cause for the 1.68:1 at the 101.5 MHz input port of the combiner.

Sweeping each of the combiner input ports revealed the problem: the 101.5 MHz filtered input port was bleeding

A Bext filter/combiner serves the three translators. Field adjustment was required to get proper performance due to minor adjustment changes that occurred during shipping.

Nautel VS300 transmitters are used for the translators.

back through to the broadband port. Apparently during the shipping process, the combiner was dropped, altering the internal components and altering the factory tuning.

With further assistance of Mr. Bononato, the combiner was "walked" back into an operational state by successfully monitoring the Nautel VS300 VSWR indications of the 101.5 translator, feeding one of the filtered inputs and the 97.1 MHz translator feeding the broadband input of the combiner.

Basically, the combiner filters are like a "T" network with several variables available to achieve the isolation and match required for the system. The adjustments were extremely small and several iterations were necessary to align this section of the combiner system properly. Ultimately, the VSWR at 97.1 MHz and 101.5 MHz was reduced sufficiently to allow full power output from all three Nautel VS300 transmitters.

Once all of the rigging, testing and tuning were completed, the three translators were put into service.

For off-air reception at the translator site, three Sangean HDT-1X receivers were chosen for the quality of audio, size and the price point. We used the ART IHF-to-balanced line output to convert the receiver output impedance and levels to then feed individual Omnia One audio processors. The composite output of the audio processors was the input feed to the Nautel VS300 transmitters. The overall quality of the retrans-

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H. HAWAII MEDIA

(continued from page 17)

mitted program sources was excellent.

While driving around the high-rise buildings in downtown Honolulu, the three translator signals remained strong, clean with infrequent multipath interference. The only time the translator signals became noisy was in the western regions of mountain range on Oahu. Because the translator site is at the peak of Mount Tantalus, the market coverage rivaled the Class C signals throughout the island of Oahu.

Sangean tuners show reception of the KORL HD channels.

ADD MULTIPLE HD

The unique aspect of this HD installation was the desire by the client for the HD1, -2, -3 and -4 channels to operate in the best possible stereo because the three FM translators on Mount Tantalus would be retransmitting the HD sub-channels. I have installed FM HD systems with an HD2 and occasionally an HD3 sub-channel, but this would be my first HD4 with all sub-channels operating in stereo. I'd been told by many engineers that it was not possible to have three sub-channels operating in stereo, only monaural.

The Nautel Importer Plus and Exporter Plus were configured to operate at the studios with a five-port input switch and my laptop PC. The Importer Plus was given an LCD monitor, keyboard and mouse. The IP addressing for these units were verified and the MAC address for the M50-HD exciter was programmed into the Exporter Plus.

This step is important. Because the connection between the Importer, Exporter and the M50-HD exciter is a UDP link, it is imperative that the MAC address of the M50 is programmed into the Exporter. The MAC address sticker was found on the back panel of the M50-HD. Just jot down the address and double check that you have it correct; otherwise the Exporter Plus will not communicate with the M50.

With the invaluable assistance of Nelson Bohorquez, Nautel customer service engineer in Quincy, Ill., the channel setup was effortless. Select the necessary encoding algorithm, Fm_Hybrid_MP3_SPS3, and then reduce the default bandwidth of the HD1 from 96 kbps to 48 kbps. To build additional stereo HD channels requires that bandwidth from the main HD1 signal be given to the other HD channels. To achieve stereo programming, the 48 kbps from HD1 was distributed across HD2, -3 and -4. The resulting bandwidths were HD1 at 48 kbps, HD2 at 32 kbps, HD3 at 24 kbps and HD4 at 24 kbps, respectfully.

For the audio inputs to the HD system, the Nautel Importer Plus was ordered with a four-digital-input audio card, which includes the digital input and output umbilical cable. Due to the size and weight, I secured the cable to the side of the studio equipment rack after the installation of the unit to keep any mechanical stress from the high-density multi-pin connector.

LOTTA STL

The next phase of the project was an upgrade for the STL system, necessitated by the new digital feeds. The studio facilities are on the fourth floor of the Pioneer Plaza in downtown Honolulu and use a T1 connection to the KPHI(AM) transmitter site as the first leg of a two-hop STL to the KORL(FM) transmitter site. For the T1 link, the Moseley SL-9003-T1 units provided

Because the translator site is located at the peak of Mount Tantalus, the market coverage rivaled the Class C signals throughout the island of Oahu.

the following signals: monaural KPHI(AM) program audio, the KORL(FM) stereo program audio, a bidirectional LAN, a unidirectional LAN, a back-haul single channel analog feed from the satellite receiver at the AM transmitter site and an RS-232 channel to transport the satellite closures to the KORL automation PC through the CircuitWerkes TransCon-16. The concept was to use the KORL(FM) HD2, -3 and -4 to feed the new translators in stereo.

The backup STL system is a Moseley Event 5800 system, which provides LAN connectivity to the FM transmitter site as well as the stereo program material for the backup audio processor at the transmitter site. Unfortunately, every time a U.S. Navy ship either arrives or departs Pearl Harbor, the surface navigation radar signals disrupt the 5.8 GHz STL frequency for as much as 30 minutes at a time. This is the reason the Event 5800 is the *backup* STL system.

For the replacement second hop, a Moseley StarLink SL-9003Q-2SLAN STL, receiver and transmitter and a Cresend Technologies 10 Watt boost amplifier provided the perfect solution.

Replacing the original Moseley PCL-606C STL transmitter with a new Moseley SL-9003Q-2SLAN and Cresend boost amplifier was accomplished without any loss of FM air time. The backup STL link fed the backup audio processor at the FM transmitter while the new Moseley SL-9003Q StarLink system was put online and tested successfully.

The ETH-03 output and the stereo analog audio output of the Moseley T1 Slave unit were given ID tags and patched to the proper inputs to the StarLink STL transmitter.

At the FM transmitter site, the StarLink STL receiver ETH-03 and AES outputs were routed to the Exgine and the AES inputs for the Nautel M50-HD "B" exciter using Cat-6 shielded cabling and shielded digital audio cable to the M50-HD AES input.

An STL rack at AM transmitter site shows existing Moseley T1 equipment and newly installed second hop for the main FM signal. The new Moseley also transports the Ethernet connection between the studio-mounted HD Importer and Exporter to the HD exciter at the FM transmitter.

HD SYSTEM TEST: DOES IT ACTUALLY WORK?

As a system test, the Importer Plus and Exporter Plus and five-port input switch were taken to the KORL(FM) transmitter site and the Exgine link to the M50-HD was connected to the switch.

The Nautel V5 transmitter was configured for FM+HD and put on the air with the exciter to verify proper HD digital carrier modulation and transmitter stability. The RF spectrum was checked using my Sencore PSA-1505 spectrum analyzer with the markers set to identify the HD mask; the transmitting system performed flawlessly! For this test, there was no audio present on the HD subcarriers.

After monitoring the performance of the transmitting system for an hour, the Nautel V5 transmitter was restored to analog only and the Nautel Importer and Exporter Plus units and five-port switch were taken back to the studio, installed into the STL equipment rack and the five-port switch connected to the ETH-03 port of the Studio StarLink T1 chassis to feed the UDP data stream.

THE FINAL CONNECTIONS

Broadcast Tools ADC-1 units were chosen to convert the three HD sub-channel program feeds from analog to digital. Following the documentation for the ADC-1 units, all the analog program sources were calibrated for a +4 dBm level and the digital output levels were calibrated for 0 dBfs. The importer digital inputs were connected to the appropriate ADC-1 outputs and the three BSI automation systems were brought online.

With another visit to the main FM transmitter, the Nautel V5 was again converted to FM+HD and put on the air. Using my portable Insignia HD receiver, I detected all four of the HD subcarriers in stereo with

(continued on page 20)

Why does Axia outsell every other IP console? We've got connections.

Did you know that there are over 2,500 Axia consoles on the air? That's more than all other AoIP consoles — combined. Is it because our ads are so irresistible? Our marketing guys think so... but, no. It's because broadcasters know that a network's value increases with the number of devices that talk to it. And nobody connects to more IP-Audio devices than Axia.

With this huge installed base of broadcast studios around the world, we've attracted dozens of partner companies, all offering Livewire™-compatible products. A device with a Livewire port is instantly available to any other device on the network. So, if you're shopping for IP consoles, be sure you ask: "How many partners do you have?" Because a network that only plays with itself isn't very well-connected... is it?

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AXIA

H. HAWAII MEDIA

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the client's three new formats. Using the Sencore PSA-1505 spectrum analyzer, spectrum occupancy was once again checked and verified to be within the FCC HD mask.

The only detail remaining was to fine-tune the analog delay. Once back at the studio, this process only took about five minutes to configure a seamless blending from analog to digital with matched levels.

As a side note, when I listened to the other HD broadcasts in the Honolulu market I found little to no analog delay had been incorporated on some of the signals. My Insignia receiver would detect an HD signal and, instead of blend, the digital audio would simply jump back and replay the last several seconds of the analog signal.

With the KORL(FM) HD signal in full operation, the final step was the FM translator site and retuning the three Sangean receivers. The Nautel VS300 FM transmitters were shut down, and then each receiver tuning preset was memorized for the appropriate HD sub-channel assignment and verified before being put on air.

WHO NEEDS COOLING?

Within several hours of successful operation the first system problem cropped up. The three translator signals began repeating the main channel 101.1 analog audio, not the HD subcarriers as assigned. Resetting the three receivers with the preset button to the proper HD subcarriers cleared the problem. About 10 days after returning to home, Mr. Hochman notified me that the Sangean receivers were routinely switching back to analog mode, requiring him to drive to the site several times a day to reset them for HD subcarriers.

The problem turned out to be due to room heat causing the receiver to unlock from the HD signal. Mr. Hochman met an air conditioning technician at the site to pull out a dead air conditioner, temporarily hang plastic sheeting to keep any direct moisture from the equipment and install a box fan to keep the air circulating until the replacement air conditioner is installed.

Although the Sangean is a great-sounding FM receiver, I contacted my SCMS sales representative, Bob Mayben, and asked what other HD receivers are on the market that would stay locked onto the assigned HD channel, power up only to the assigned HD channel, have a small size and it sure would be great to have balanced left and right analog outputs. This seemed to be a tall order and we were expecting a matching price tag *if* such a product was available.

Within an hour, Mr. Mayben called me back with great news. Knowing just how important the reliability needed to be for the H. Hawaii Media translator site, Mr. Mayben contacted Lukas Hurwitz, sales and marketing manager for Inovonics, and arranged for three of their new INO 632-00 HD receivers to be shipped immediately to Honolulu.

The operating features provided by the Inovonics INO 632-00 are impressive. If you need a good HD receiver with plenty of features, you must look into this product.

HD PROVIDES UNIQUE FORMAT CHOICES

In closing, I want to point out something about Mr. George Hochman and H. Hawaii Media that gives me a real and lasting belief in radio entertainment: Mr. Hochman perceived a need within his market place. A need that was ignored by the other "big players" in Honolulu: the value and power of communicating to the ethnic listeners of Honolulu with entertainment they want, need and would appreciate.

Through Mr. Hochman's innovative use of HD, financial investments, research, licensing struggles and a virtual worldwide search for just the right caliber of talent for his two ethnic formats, he began building for the future — a future he saw for Honolulu, for Korean and Japanese citizens and visitors alike, by providing the latest Korean and Japanese popular music with announcers speaking in their language as well as English.

Even though I didn't understand the language, I was entertained by these new musical formats, and to me this is what truly works in radio: entertainment that cuts across language and cultural boundaries without any apparent effort on the part of the broadcaster.

Mr. Hochman demonstrated the imagination, willingness and determination to step away from the "cookie cutter" formats provided by people who really have no clear idea of what listeners want, need or feel comfortable with in their listening markets. Some may say that he didn't have any choice because the "big players" owned all the ratings with their "cookie cutter" formats, and that may be so; however, necessity is the mother of invention. Mr. Hochman has the imagination to challenge the "Goliaths," and the final proof of performance is in the bottom line as well as the devotion of the listeners and advertisers who will support and encourage the entertainment they want for themselves.

Stephen Rutherford is owner of Rutherford Resources in Bullhead City, Ariz.

Got a facility story to tell? Drop us your story idea at rwee@nbmedia.com.

CORRECTION

Due to an editing error with Cris Alexander's story "Use Antenna Modeling to Evaluate Potential Reradiators," in the Aug. 22 issue, the captions on Figs. 5 and 6 were reversed. We also printed the wrong graphic for Fig. 1. The correct figure is shown below.

Fig. 1: Array geometry moment method model for three-tower in-line array.

DRM

(continued from page 22)

access, wireless networks and 4G mobile communications. The combination of high data capacity, high spectral efficiency and resilience to interference as a result of multi-path effects makes it ideal for the high data applications that are becoming common in today's communications scene.

This signalling and detection technique incorporates kinematic filtering and signal multiplexing, aptly named "kineplex." In the early 1950s Collins Radio foresaw the need to transmit data over relatively narrow channels. The 16-tone Tactical Digital Information Link, TADIL-A, is used by the U.S. Navy to share radar tracking data, which makes a classic buzz sound like an alligator happily making little 'gators. It's one of the more distinctive sounds on shortwave. The receiving end of the link used a bank of 15 extremely-sensitive electromechanical resonators, housed in an equipment cabinet six feet high by three feet wide by two feet deep.

Receivers have come a long way since then.

NEW MODELS

DRM receivers evolved from front-end down-converters feeding personal computers. Later, digital signal processor manufacturers started producing chip sets containing both the DSP and the RF digitizer. Standalone receivers used these chip sets or modules without the PC.

Initially, many firms tried to "seed" the shortwave market by offering "add-on" accessories to use the down-converter of existing higher-end receivers and adding a complex processor at the IF, as shown in Fig. 4. The bandwidth of a DRM signal varies from 9 kHz to 20 kHz, and the number of carriers used in the COFDM-modulation is relatively small (a maximum of 460 at the highest bandwidth vs. lowest carrier spacing options). These features motivated a real-time software implementation of a DRM-receiver on a conventional personal computer using the sound card as the input and output device. A long-, medium- and shortwave front end with an intermediate frequency (IF) between 5 kHz and 15 kHz is used to receive the DRM signal. This addressed the technically-adept, but didn't apply to the villager starving for entertainment.

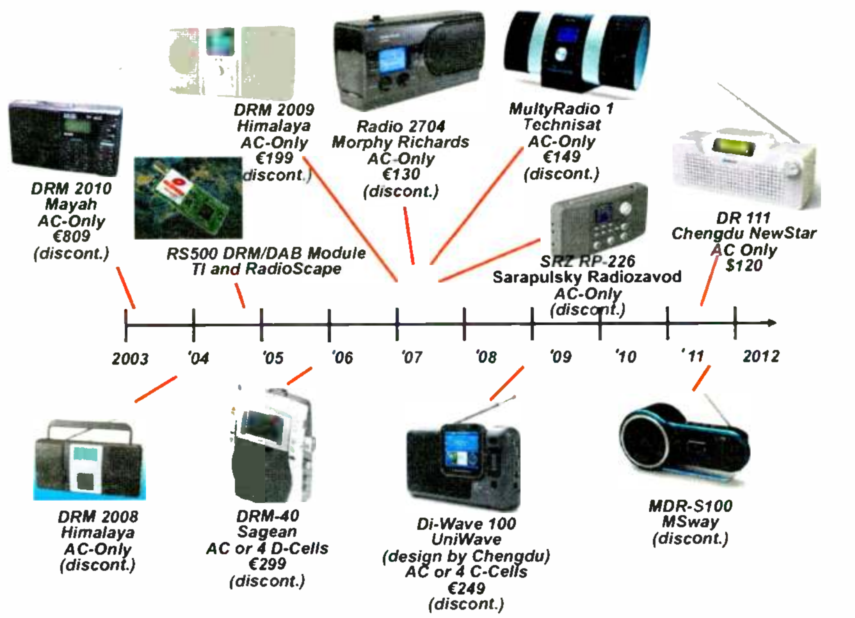

Fig. 3: The evolution of DRM 'standalone' shortwave portable receivers has left many artifacts behind.

One of the more-interesting DRM modules was the Digital World Traveller, Fig. 5, by Coding Technologies introduced in 2004. This handy little module was connected to the USB port of a PC or Notebook. Priced at $260, the device came with software capable of receiving DRM, FM and AM radio programs without any additional power supply or battery.

Fig. 5: The Coding Technologies Digital World Traveller was a convenient shortwave accessory for the traveler with a PC on the go.

Around 2007, a few manufacturers started selling standalone receivers (Himalaya Electronics, Technisat, Morphy Richards, Starwaves, UniWave, Sarapulsky Radiozavod). Most of the receivers were based upon the discontinued Radioscape RS500 module. The standalone models relied on household electricity and thus were not portable, as seen in Fig. 3. We do see a steady price reduction headed for the magic $100 goal (in production quantities, and not including taxes, V.A.T. or shipping).

PROCESSING GIANTS

DSPs can be found in most of our day-to-day consumer devices, including mobile handsets, digital cameras, navigation devices, TVs, DVD players and game consoles. They are ubiquitous in multimedia, telecommunications and networking applications. These products use a variety of hardware approaches to implement DSP, ranging from the use of off-the-shelf microprocessors to field-programmable gate arrays (FPGAs) to custom integrated circuits.

Programmable "DSP processors," a class of microprocessors optimized for DSP, are a popular solution for several reasons. In comparison to fixed-function solutions, they have the advantage of potentially being reprogrammed in the field, allowing product upgrades or fixes. They are often more cost-effective (and less risky) than custom hardware, particularly for low-volume applications, where the development cost of custom ICs may be prohibitive.

The lowest complexity processors are "hard-wired" or dedicated processors such as FPGAs, lacking the flexibility to be reprogrammed should the DRM specification change. This was the nemesis for Digital Audio Broadcasting (DAB, Eureka-47), where dedicated chip sets specifically developed for DAB were termed worthless when the standard was updated, leaving receiver manufacturers reluctant to enter an immature arena.

Cutting-edge technology can allow a DSP to have lower power than the equivalent ASIC, due to the ASIC using older technology. Also, the DSP can usually be controlled to minimize clock speeds when the processing load allows. Alternatively, the power in an ASIC is minimized by ensuring that the signal-processing operations are dimensioned correctly, particularly in terms of the bit widths being processed.

FUTURE OF DRM?

The success of DRM is dependent on a combination of satisfying a need and technology arriving with the ability to fit that need. In order to provide better quality, a more complex carrier signal and additional processing was required. The need was to provide a better service, but the very enabling technologies also provided alternate forms of entertainment and information. Within the last decade's window of opportunity DRM took one step forward and other mass media took several steps forward.

In a developed country, the speed of development favors the Internet because of rapid acceptance of service. The hardware is inexpensive, the bandwidth is expanding and the social media networks have been providing the individualized services that people demand. Given a limited budget for a household in an undeveloped country, they will try to maximize their "bang for the buck."

With the advent of the Internet and SmartPhones and social media networks, all bets are off. We have recently seen the value of social media networks in social uprisings in the Middle East and Africa. People can get the same shortwave information over social media with the uncensored spontaneity of amateurs.

Without a viable (cost and battery-conscious) receiver, DRM has a hazy future.

Ernie Franke is a broadcast consultant in St. Petersburg, Fla. He earned a master's degree in electrical engineering and has been the chief engineer at several broadcast stations.

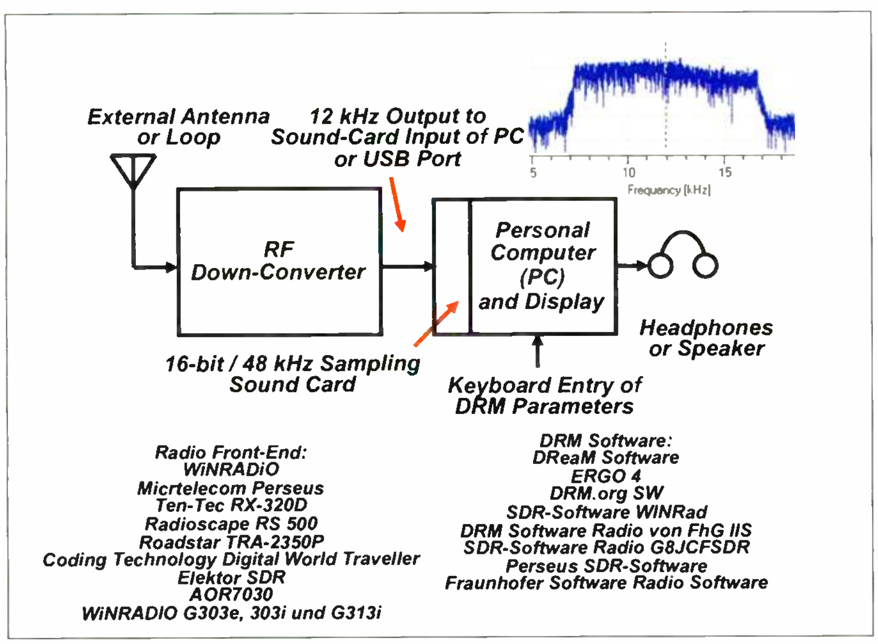

Fig. 4: The DRM software radio uses an RF down-converter ahead of the sound card in a standard PC.

Will Digital Radio Mondiale Save Shortwave Broadcasting?

High-technology system may not head off Internet advances

BY ERNIE FRANKE

Once touted as the "Savior of Shortwave," Digital Radio Mondiale has not lived up to its hype. Proposed in 1988, with early field-testing in 2000, inaugural broadcasting in 2001 and its official rollout in 2003, DRM has had a lackluster career over the last decade.

With the allure of FM-quality audio and fade-free operation, it had appeared that DRM might revive the shortwave community. Unfortunately, it has been overcome by other events, some technical and some social. The main weakness has been alternate sources of information and entertainment, fueled by the very technology that gave DRM hope.

Additionally, in areas of the world without ubiquitous social media, DRM has yet to realize receivers at a moderate cost with adequate battery life. The very processing technology that allows improved operation using the more complex DRM waveform costs more and consumes more power than the standard AM receiver. A quick look at standalone DRM receivers over the past decade shows almost a dozen companies entering the market, only to retreat when the promise didn't materialize.

A DEMAND AND A HOPE

If you mention "shortwave," the average person pictures a wiry-looking "ham radio operator" in a basement or attic. But in fact, most of the less-developed world listens to shortwave. Outside the U.S., shortwave has been, and probably will continue to be, a serious contender for disseminating entertainment and information.

The rise of the Internet has influenced many broadcasters to cease their shortwave transmissions in favor of broadcasting over the World Wide Web. When BBC World Service discontinued service to Europe, North America, Australasia and the Caribbean, it generated many protests. The shifting of resources from shortwave to Internet and television by the Broadcasting Board of Governors, which oversees U.S. international broadcasting, further reduced broadcasting hours in the English language. With recent budget slashings of 70 to 80 percent, resulting in announcements of closing large stations such as the Radio Netherlands Bonaire and Radio Canada Sackville sites, increased pressure has been placed on shortwave to perform. Although most of the prominent broadcasters continue to scale back their analog shortwave transmissions or completely terminate them, shortwave is still common and active in developing regions, such as parts of Africa and South America.

Examining both the location of DRM stations and target areas, Fig. 1 shows that most DRM shortwave stations and target areas are in Europe. Until an inexpensive, battery-conscious receiver is available, continents such as South America and Africa won't be viable target areas.

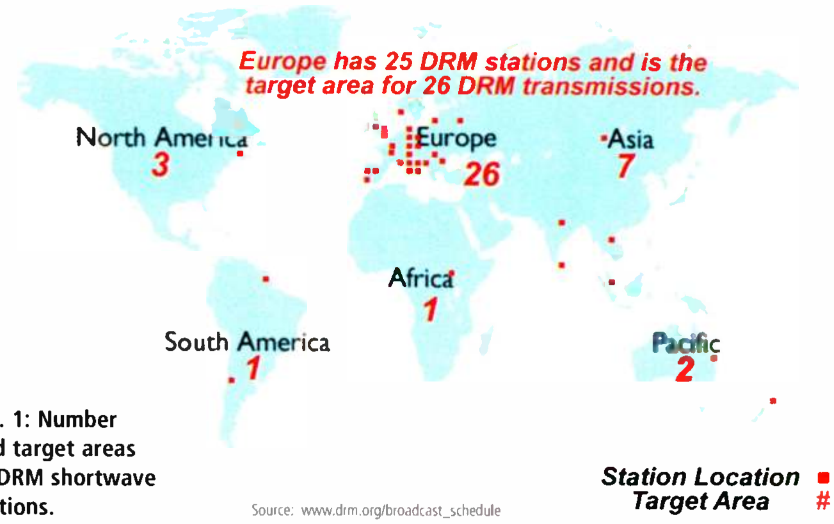

Fig. 1: Number and target areas of DRM shortwave stations.

Adil Mina, VP of business development for Continental Electronics, chairs the DRM USA Group. He has written on its website that "the receiver that all of us are looking for is still the small receiver, the inexpensive receiver that will have a good battery life. That's what most people are looking for. It's the one that should be like your BlackBerry, your telephone, that can sit for two days, three days, without you having to go back and charge it."

IN PURSUIT OF QUALITY

The main requirement of DRM development was to ensure that far greater audio quality could be achieved whilst keeping the transmissions in a form to operate alongside existing AM transmissions. This meant having the ability for the transmissions to occupy a variety of different bandwidths dependent upon the location and frequencies in use.

There are two main elements to the DRM waveform: audio coding and RF modulation. Along came several leaps in technology to compress CD audio into a manageable size. Improved computer technology also provided the necessary processing speed to adopt a complex waveform. However, with higher processing speed comes increased cost and battery consumption.

AUDIO CODING AND COMPRESSION

DRM's audio compression system employs two main techniques. The first is called Advanced Audio Coding. The brain does not perceive all the sounds that are heard by the ear. A strong sound on one frequency, for instance, will mask out others close in frequency that may be weaker. AAC analyzes the audio spectrum in sections and only encodes those sounds that will be perceived. However AAC on its own does not provide sufficient compression of the data to enable the transmissions to be contained within narrow shortwave bandwidths.

To provide the additional data compression, a scheme known as Spectral Band Replication is employed. This analyzes the sounds in the highest octave, which are normally from sounds such as percussion instruments or those that are harmonically related to other sounds lower in frequency. SBR analyzes them and sends data to the receiver that will enable them to be reconstituted later.

COFDM

The DRM transmitted signal uses a form of modulation known as Coded Orthogonal Frequency Division Multiplexing. It is resilient to many common forms of interference and fading. Its main drawback has been that it requires a significant level of signal processing to extract the data from the carriers and reassemble it in the correct fashion. Signal processing ICs are now sufficiently powerful and are at a reasonable cost to make the use of this form of modulation viable.

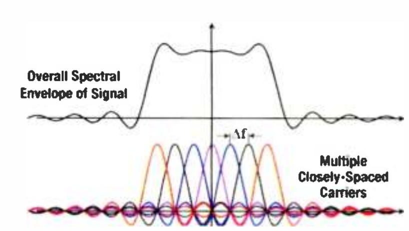

Fig. 2: The COFDM Spectrum of a DRM signal

COFDM uses a large number of closely-spaced carriers that are modulated at a low rate data. Each carrier is modulated with Quadrature Amplitude Modulation using a selectable error coding. Normally, closely-spaced signals would be expected to interfere with each other, but by making the signals orthogonal to each another, there is no mutual interference. This is achieved by having the carrier spacing equal to the reciprocal of the symbol period. The data to be transmitted is split across all the carriers. By using error correction techniques, if some of the carriers are lost due to multi-path fading effects, then the data can be reconstructed.

COFDM has gained a significant presence in the wireless market place. It is now popular with wideband digital communication, digital television and audio broadcasting, DSL broadband Internet

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