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Currents Online

Selected headlines from the past month.

Silliman Receives NAB's Engineering Honor

Thomas B. Silliman, president of Electronics Research Incorporated (ERI), and retired ABC senior advisor of science and technology Antoon (Tony) Uyttendaele will be presented with the 2008 NAB Engineering Achievement Awards for radio and TV respectively during the Technology Luncheon at the NAB Show.

Silliman

Georgia Tech Tests Wireless Emergency Alert System for Visually Impaired

Results indicate that 94 percent of blind and visually impaired test subjects found the tested system to be a significant improvement over their current methods of receiving emergency alerts.

Two Alabama Cities Adopt Alert FM

Selma radio stations WALX and WJAM, and Birmingham radio stations WBHJ, WBHK, WBPT, WNCB and WZZK, join stations in Mobile, AL, and Florence, AL, as some of the first in the state to deploy a local platform for Alert FM.

Broadcast Electronics Adds Burgess in Sales

Burgess will oversee sales for BE Audiovault and The Radio Experience products for the southern U.S. and Alaska and several key broadcast accounts.

Loeffler Joins Mayah to Open North American Office

Daniel Loeffler will represent the German company as it opens a North American office.

Arbitron, Nielsen Terminate Project Apollo

The two companies had been working on the proposed single-source, national research service since early 2005.

APT Prepares to Launch New Brand Identity

A new logo and a new corporate identity will be the focus at the NAB Show this year. The move is designed to reflect the company's growth in size and maturity.

Site Features

Take the FASTtrack with You

The exclusive *Radio* magazine FASTtrack organizes the NAB Show exhibitors into categories so you can find everyone fast. The application runs on any handheld device and also includes the Broadcast Engineering Conference listings.

Engineer's Notebook

Everyone has his coleection of tips and tricks. So do we. They're posted in the Engineer's Notebook, and they're just what you were looking for.

E-mail Newsletters

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A day in the life

What's you typical day at the NAB Show?

Everyone's agenda is different during the convention. I am often asked what my day is like at the convention. It's probably safe to say that it's probably not like yours. Compared to mine, your schedule is probably rather loose.

I know you have certain sessions that you want to attend, and you may have some meetings with exhibitors scheduled. I also know that some companies hold engineering team meetings at the convention. Still, I doubt that your schedule has something scheduled every 15 or 30 minutes.

Mine does.

This year, I'll be in Las Vegas from Friday to Friday. The first two full days, Saturday and Sunday, are a mix of events. While not as hectic as the weekdays, I'll be moving between several SBE meetings, checking in on the SBE Ennes Workshop when I have a moment, dropping in on manufacturer seminars, preparing materials for use during the week, and just making sure that everything is ready to go.

And that's just Saturday.

Sunday is similar, but I start with a six-hour (at least) SBE board of directors meeting.

Once the convention floor opens, the clock starts ticking. After waking up Monday morning, it's almost non-stop motion until I go to bed. Most of my time is spent on the convention floor where I am visiting as many exhibitors as I can to learn about the new technology and new products. Much of my time is scheduled for specific appointments, and during the time that is not scheduled I move from booth to booth.

When I'm not on the exhibit floor, I'm at events. On Monday morning I am moderating the Broadcast Engineering Conference session called Communicating with Management. On Tuesday afternoon I'll help present the SBE membership meeting. On Wednesday I'll attend the Technology Luncheon and then meet with the Pick Hits panelists to collect their choices for the annual Pick Hits of the convention.

Thursday is my light day when it comes to events, but that just means I have more time on the convention floor.

In the evenings, the busy schedule continues. Sure, there are parties to attend, but I still have work to do. I usually have a business dinner to attend as well.

When I return to my hotel room – usually no earlier than 10 p.m. – I would like nothing more than to kick back and fall asleep. But I can't. There is still work to be done. During the day, I have collected a great deal of information and material, and I have to sort through it while it's fresh in my mind. Some of this information is used for Currents, our daily online news update.

Then there are the photos. As we have done since 2005, *Radio* magazine will produce a photo blog of the convention. The *Radio* magazine staff, contributors and reporters take photos during the convention, and we prepare and post a large number every evening. We don't post everything. We select the best of the batch each evening to give you a visual taste of what happened each day.

And then finally I check e-mail and voice mail so I can stay in touch with business.

I know you're no stranger to long hours and a heavy workload. I've built studios and transmitters sites, too. But while the convention for many attendees is something of a break from the routine, the convention *is* the routine for me. I'm not complaining, but by Thursday afternoon, I'm ready for it to be over.

And then in December, we start planning for the next year's convention again.

Chris Scherer

What's your opinion? Send it to radio@RadioMagOnline.com

*Above: Rays broadcasters **Andy Freed** (left) and **Dave Wills** (right) interview Rays' star third base prospect **Evan Langoria** on the "The Hot Stove Radio Show."*

*Top: **Larry McCabe**, Tampa Bay Rays Senior Director of Broadcasting and **Rich Herrera**, broadcaster and Director of Radio Operations are shown on the field during spring training.*

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Fans of the Tampa Bay Rays baseball team are intimately familiar with Dave Wills and Andy Freed, play-by-play announcers and hosts of "The Hot Stove Radio Show." Offering the inside track on all things Rays, the show kicked off its 2008 season with the "Countdown to Opening Day" series. While at a remote from a well-known sports bar, ACCESS showed its true worth. Two minutes before the broadcast, the ISDN line that was supposed to be used for the broadcast failed to connect. Luckily, they had the ACCESS running on Wi-Fi provided by the restaurant. The broadcast got on the air and was flawless for the entire one hour show.

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Content delivery options

By Kevin McNamara, CNE

In many ways, radio has not changed since the 1940s. Yes, the technology has become more sophisticated, the quality of the audio is better (well, depending on whom you ask) and we have transitioned to a fully digital studio and delivery system. But radio still does the same thing it has done since its inception: broadcast in a single direction. This worked fine for radio's first 50 years, but the advent of the Internet, satellite and various wireless platforms provide a growing threat to the once mighty radio station.

As a stand-alone medium, traditional broadcasting is subject to a multitude of restrictions as compared with other Internet-driven delivery platforms. Its primary limitations include coverage area, available bandwidth and interactive capabilities. While these are significant issues, they are not impossible to overcome using some alternative delivery methods now available.

Even new forms of audio media use the same one-way transmission scheme that radio has used since its inception.

Another interesting point to consider, most of the multimedia content available to the general public is also broadcast only in a single direction, hence the terms podcast, webcast and Bluecast. The list goes on forever. Internet radio stations also broadcast in a single direction. They are really not much different, with the exception that many of the new media radio broadcasters offer users a wider range of services including video, data and on-demand capabilities.

Many stations are already streaming broadcasts on one of the many internet radio portals and on their own websites. While the current state of technology permits any person connected to the internet to broadcast audio and video, you need to be aware of the various delivery options available and some of their benefits.

Choices of delivery platforms

Just 20 years ago terrestrial broadcasting was the primary platform for the delivery of multimedia content. Times were simpler then, the biggest issues we dealt with was choosing the format to reach the most 25-54-year-olds and making sure the processors were adjusted properly for minimal listener fatigue. The Internet was arguably the reason for the evolution of the new media delivery, but still required hard connection to a network; however, in recent time, new wireless services have un-tethered access and now a variety of handheld devices and laptop PCs can access streaming media in real-time or store for use at a later time.

The FCC has recently auctioned spectrum that will allow carriers to create seamless local, regional or national wireless networks supporting higher speed voice, video, radio and data services. The handsets for these services promise to fulfill all business, personal and entertainment needs in a single device.

While the licensees for existing and emerging wireless services have or are in the process of cutting deals with content providers for custom entertainment offerings, these devices also have Internet access, which makes any online streaming media available.

Whether you are currently streaming audio or are planning to make the move it is necessary to have a basic understanding of the options for delivering the content.

Streaming multimedia

The first thing to understand is the distinction between streaming and non-streaming multimedia. Streaming describes the method of delivery rather than the specific coding method. Streaming is the process of sending an encoded datastream over a network for reception (decoding) at one or many network connected devices, basically

a broadcast, i.e. Internet radio. Non-streaming delivery is content stored on a server for delivery when requested by a user, i.e. podcast, webcast. These are also called linear or non-linear delivery methods.

It is easy to understand that less bandwidth will be required to send one stream to many users simultaneously, as opposed to sending multiple unique streams to one (or many) users. In order for the multimedia files to get to the intended users there are four different types of routing schemes that can be utilized depending on factors such as bandwidth limitations, location of users and type of streaming content (single or multiple streams). There is a significant financial impact to choosing the wrong delivery method due to the cost of bandwidth; this must be closely evaluated with your provider in order to maintain maximum delivery efficiency with the lowest cost.

- Unicast – Provides streaming delivery from a single source to a single destination. This is perhaps the least economical delivery method. Since multiple users will get different streams essentially, the bandwidth required for a single stream is multiplied by the amount of concurrent users. This is also called a one-to-one relationship.
- Multicast – There are several different flavors of multicasting, but the most common is called IP multicasting. In this arrangement, a single group address is established (such as aaa.bbb.ccc.ddd). The destination nodes initiate contact with the group address which is completely opposite of traditional TCP/IP delivery. A routing tree is established when each of the destination devices establishes the connection. Using this routing tree concept is an efficient means to establish a streaming network but can be expensive when compared to other means. This is a one-to-many relationship.
- Anycast – This is a variation of the multicast scheme, however the connections are initiated from the source address to the destinations. The major difference is that only one connection is established at any given time based on the most efficient routing. What makes this efficient is that the source address is propagated through several servers, the closest connection is established, and then the process is repeated until communication is established with all connections. Since Anycast is always establishing, dropping and re-establishing connections, this arrangement does not work as well with connection-oriented protocols due to the additional time required to establish the connection every time. This is also considered a one-to-many relationship.
- Broadcast – This is pretty much as you would expect; data initiated over the network can be received by any device on the network. This can be an inefficient method of delivery from a pure flooding-the-network-with-traffic perspective. This technique is no longer the preferred method but this is a true one-to-many relationship.

As most things technical, these explanations form a good starting point to evaluation your particular needs. I recommend that you perform your own research and have a complete understanding of the various formats, protocols and delivery methods.

For terrestrial radio to survive it needs to go well beyond being a single-dimensional platform. Using these technologies to complement the traditional broadcast operation is essential to take broadcasting to the next level.

McNamara is president of Applied Wireless, Cape Coral, FL.

Localism proceeding portends re-regulation of radio

By Harry Martin

In January the FCC released its notice of proposed rulemaking (NPRM) announcing tentative conclusions and proposed rules concerning localism. The retroactive nature of many of the Commission's proposals has engendered a strong response by the broadcast industry. Comments on the proposals are due April 28. Many observers have suggested that if the proposed rules are adopted, the FCC will be turning back the clock nearly three decades, with licensees having to start complying again with burdensome record-keeping, ascertainment, studio location and staffing requirements that were dispensed with years ago.

According to the Commission, the proposed rules are designed to address the perception that broadcasters may not be addressing the needs and interests of their communities sufficiently. Over the past several years, the Commission has solicited comments from the public and engaged in localism hearings at venues all over the country. From comments received through that process, the Commission has determined that "many stations do not engage in the necessary public dialogue as to community needs and interests and that members of the public are not fully aware of the local issue-responsive programming that their local stations have aired."

Dateline

June 2 is the deadline for submission of biennial ownership reports by radio stations in Michigan and Ohio.

On June 2, radio stations with more than 10 full-time employees that are located in Michigan and Ohio must electronically file their Broadcast EEO Mid-Term Reports (Form 397) with the FCC.

Also on or before June 2, radio stations licensed in the following jurisdictions must place their annual EEO Reports in their pubic files: Arizona, D.C., Idaho, Maryland, Michigan, New Mexico, Nevada, Ohio, Utah, Virginia, West Virginia and Wyoming.

The following are some of the highlights of the FCC's proposals to address this determination:

- The Commission has tentatively concluded that each licensee should be required to establish a permanent community advisory board that would assist in ascertaining local issues and provide ideas for responsive programming.
- Stations would be required to have staffing at their facilities 24 hours per day.
- Stations would be required to maintain their main studios within their communities of license. Currently a main studio can be located anywhere within 25 miles of the community of license.
- The Commission has tentatively concluded that it will reintroduce renewal application processing guidelines incorporating a specified minimum percentage of programming aimed at addressing local issues. Licensees meeting the requisite percentages would have their renewals processed by the Media Bureau, while those falling short would have their renewals considered by the full Commission.
- The Commission is considering the adoption of rules to foster improved communication between licensees and their communities, including requirements for the following: ad hoc viewer surveys via telephone or Internet; town hall meetings to help prioritize issues to be covered through news and informational programming; participation by station management on community boards, councils and commissions; and the establishment of dedicated telephone numbers, websites and email addresses, publicized during programming, to facilitate community dialogue.
- The Commission is also seeking comment on the prevalence of voice-tracking and whether anything can and/or should be done to limit that practice. Also in the content area, the Commission is seeking comment on whether it should require licensees to maintain and make available data regarding the airing of local music.

While many of these proposals clearly will not be adopted, the fact that some of them are being considered at all is worrisome. Keep in mind that the FCC's localism proposals will not surface for a vote until a new FCC chairman, most likely a Democrat, is in office.

Martin is a past president of the Federal Communications Bar Association and a member of Fletcher, Heald & Hildreth, Arlington, VA. E-mail martin@fhhlaw.com.

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On-air Te

Radio broadcasters rely heavily on their telephones. Whether used for news, interviews, contests, requests or talk radio, the ability to easily manage phones on-air is a long-standing requirement, and that's unlikely to change. As technical director of Comrex, I've had a long-time understanding of the state-of-the-art in this domain and that was only enhanced by Comrex's acquisition of the Gentner line of telephone interface equipment in 2002. I'll describe here the current paradigm in telco interface and introduce where this stuff is heading – and it's heading there fast.

Hybrids aren't just electric cars

The essential piece of studio telephone integration is the telephone hybrid. As shown in Fig. 1 (page 16), the two-wire nature of the local telephone loop necessitates that the conversation back and forth must take place over the same circuit. Fig. 1A shows this as being analogous to two people speaking into a single tube. The downside of this arrangement is that each speaker will hear himself echoed in the tube louder than he'll hear the far end speaker.

elephone nterfaces

By Tom Hartnett

Fig. 1B is closer to reality, showing the telephone network as separating the send and receive channels. But in the case of analog phone lines these are reconverted to a single two-way circuit for delivery to the customer. In both the 1A and 1B scenarios, a hybrid must be used to convert the studio-side, two-wire circuit back to four-wire for use on-air. The output of a hybrid will consist of only caller audio, with all the host audio digitally subtracted from this output. It can be assumed with modern digital hybrid circuits that any remaining host audio is more than 20dB below that of the caller. Lack of a proper hybrid arrangement will result in studio host audio coloration since the hybrid feed-through will add with the main host microphone in the console and produce an effect often described as talking into a barrel. It's important to note that the main function of these systems is not to alter the send audio, remove garbage audio introduced by poor quality telephones or even far-end echo, but simply to keep the host audio unmolested by telephone feed-through.

Fig. 1C shows the best possible scenario, a four-wire connection at the studio side that keeps a completely independent path in each direction. In this case the actual hybrid function (two-to-four wire conversion) is not required to provide studio send-receive isolation. This is typical of systems that utilize ISDN, T1 or voice over IP (VoIP) interfaces at the studio (more on that later).

On-air Telephone Interfaces

But the term hybrid has come to mean much more than this conversion. Telephone feeds suffer from noise and hum, as well as widely varying caller levels (especially these days due to use of poor quality cell phone microphones). So it's essential that a hybrid (whether it utilizes a two or four wire interface) contain line filtering and AGC to combat these evils. Also, a talk-show host may want to dominate a conversation, so some kind of ducking function is required, detecting host voice activity and reducing caller audio level to a more submissive level.

Call management systems

Taking the concept further, once you move beyond one or two phone lines, you'll need the call management capabilities of a multi-line (or talk show) system. This typically consists of a mainframe that connects to the telephone lines, and one or more control surfaces to answer, park, screen and put calls on-air. In addition, studios taking callers will need some kind of screening capability to talk to callers in advance and inform on-air talent of their intentions. A block diagram of such a system is shown in Fig. 2.

Modern trends in these systems revolve around enhancing remote control capability. Higher-end talk show systems will include the ability to put these capabilities on a Web page, and this will usually combine the remote control and screening functions. The Comrex STAC talk show system's Web-based screening page is shown in Fig. 3, which is delivered by a Web server built into the mainframe of the unit. By use of a Web browser, calls can be screened and routed from any location that has Internet access, including remote broadcasts. Also, this removes the requirement for the client computer to have any specialized software installed to view and control the system. Multiple locations can monitor the caller queue and screening information as they listen to the show.

Additional features offered by some screening software may include reflection of caller ID information and logging screening information into a database.

Interface options

Many options are available to deliver multiple on-air phone lines. BRHSDN lines are certainly useable as voice lines, and have the advantage of pre-separated send and receive paths. Telos Systems has pioneered use of these for talk show systems with much success. But in my view the long-term outlook for BRHSDN service is uncertain, as fewer and fewer non-broadcast applications require it.

As the number of phone lines brought into a facility increases, the cost-benefit tips toward a multiplexed digital delivery such as T1 or PRHSDN. For maximum flexibility with use on the widest range of telco interface equipment, individual lines from the T1 may be converted back to analog POTS lines in the channel bank. While effective, this approach isn't entirely elegant, since you are intentionally introducing an additional digital-to-analog as well as a four-to-two wire conversion that must be reversed by the hybrid within the talk show system. The market for direct-interface T1/PRI broadcast telephone integration is dominated by the higher-end Telos 2101.

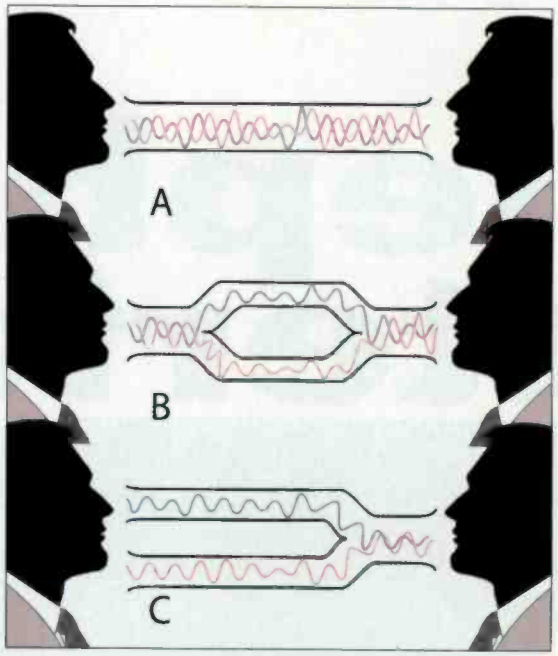


Fig. 1. Analogies of the various methods of providing a communications link.

VoIP is the future

The final option for telephone interface isn't delivered by a phone line at all. The concept of VoIP (Voice over IP) is gaining popularity quickly, and in some instances broadcasters aren't having much choice in the conversion.

The generic term VoIP can have two meanings. First, it can refer to traditional analog lines brought to a station's demarcation point, then converted to IP telephony for use over a PBX system. In this instance there usually isn't much benefit for the broadcaster to make this conversion – it's better to grab the broadcast lines at the demarc and route them to the studio interface equipment in the traditional way. This isn't always possible in an environment like a college campus, where an entire Centrex-style system is likely to be replaced by a VoIP network, and access to legacy lines is unreasonable.

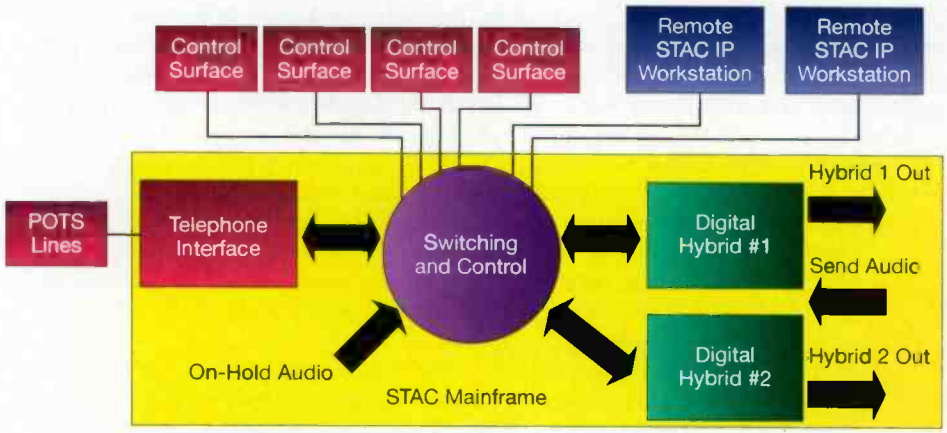
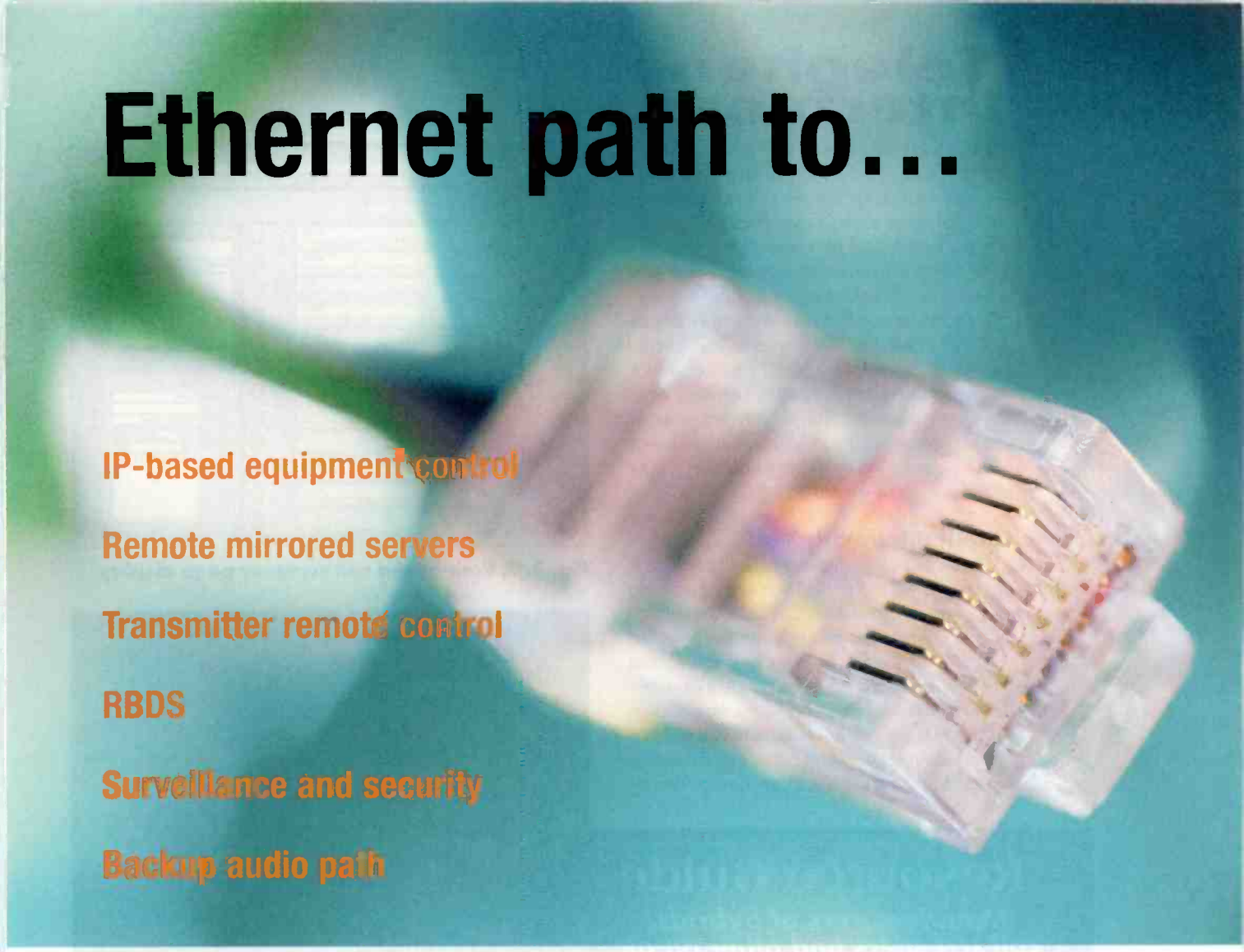


Fig. 2. A block diagram of the Comrex STAC, a talk show phone system.

Ethernet path to...



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On-air Telephone Interfaces

The second meaning of VoIP is direct access to a VoIP service provider over an Internet trunk. Services such as Vonage and Broad Voice can offer significant savings over traditional telcos, offering full-featured phone service at less than $10/month in some cases, especially if the lines are used primarily for incoming calls.

The good news is that after years of incompatible VoIP standards, both types of VoIP circuits are converging to follow the SIP protocol, which removes most incompatibilities between hardware and providers. This means that whether you're programming your SIP-compatible device as a PBX extension or a direct interface to your provider, setup is the same.

Challenges with VoIP

While it has cost advantages, VoIP interface is not always without peril. If you are subscribing to a low-cost provider via DSL or cable modem Internet connection, that link will be provided without QoS (quality of service), meaning that other Internet traffic on the local LAN or within the Internet at large can cause the transmission to drop out. This risk is usually removed within a PBX environment with traffic shaping, as well as when subscribing to the VoIP service provided by your DSL or cable provider (often at higher prices).

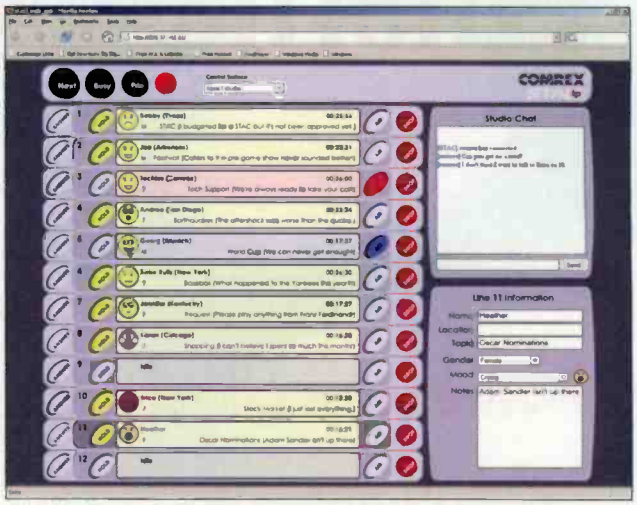

Fig. 3. A Web-based call screener page display.

The Telos 2101 can access BRI-ISDN and PRI-ISDN lines for on-air use.

VoIP phone lines usually offer a choice of audio coding. For on-air lines, it's best to stick with G.711, which is the same mildly-compressed audio codec used with traditional telephony. But that algorithm requires 64kb/s plus IP overhead, and multiple channels can fill an Internet access trunk pretty quickly. The other most popular choice for audio coding is the G.729 family of algorithms, which reduces the data stream by a factor of eight. While you may notice a slight quality reduction with G.729, this codec usually isn't the limiting factor on a poor phone call, given the horrible quality of most cell phones. And the significant bandwidth savings can provide for higher stability on contended Internet connections.

VoIP integration bridging the gap

VoIP lines can be converted to traditional analog POTS lines using inexpensive ATA (analog terminal adapter) hardware bridged into traditional broadcast hybrids. But as is the case with T1 and ISDN, you are adding analog-digital and four-two wire conversions that must be reversed in the hybrid. An additional risk present in these conversions – one that isn't as relevant in traditional telephony – is that

Resource Guide

Manufacturers of hybrids, call screeners and other on-air telephone products

AEQ
800-728-0536
www.aeqbroadcast.com

AVT Audio Video Technologies
+49 91152710
www.avt-nbg.de

Broadcast Bionics
+44 1444 473999
www.phonebox.com

Broadcast Tools
877-250-5575
www.broadcasttools.com

Circuitwerkes
352-335-6555
www.circuitwerkes.com

Comrex
800-237-1776
www.comrex.com

Condron Broadcast Engineering
602-381-5733
www.thecallscreener.com

JK Audio
800-552-8346
www.jkaudio.com

KLZ Innovations
800-334-9640
www.klz.com

Radio Systems
856-467-8000
www.radiosystems.com

Sonifex
207-773-2424
www.sonifex.co.uk

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the delay associated with VoIP can allow a significant and irritating echo to return to the caller if the ATA is not doing a good job of echo cancellation.

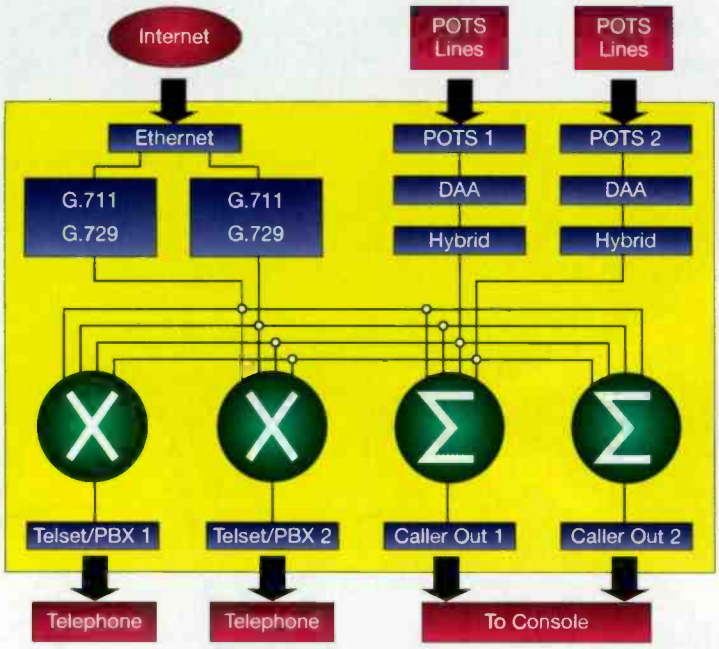

Fig. 4. A block diagram of the Comrex DH42, a phone system that can access POTS and VoIP services.

Fig. 4 shows a block diagram of the Comrex DH42, to be introduced at the 2008 NAB Show. The product allows interworking between up to two POTS lines and two VoIP lines in a studio environment. The system maintains send-receive separation on its VoIP lines, and acts as a digital hybrid on its POTS lines. Other than that, it treats all these lines equally, providing AGC and filtering to maintain call quality. All four lines may selectively be put on-air, conferenced or routed to a downstream device like a PBX. Using such a system, broadcasters can reap the benefits of VoIP technology while keeping a foothold in familiar POTS territory.

Preparing for the future

Given the onslaught of changes in technology and the need to do more, cheaper and faster, it's likely that POTS and ISDN lines will cease to become cost effective or easy to get from the telcos. Long-term, VoIP technology is poised to become the only game in town for voice telecommunications. In fact, we have had a significant increase in requests for a professional digital hybrid audio interface for SIP-based VoIP connections, which would seem to validate this observation. But being able to get your callers on the air is still the bottom line – regardless of the circuit. Giving broadcasters the tools to make that happen ensures that technology won't hinder that ultimate goal.

Hartnett is technical director of Comrex, Devens, MA.

from the company that brought you apt-X® coding technology...

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WorldNet Oslo

NAB Booth N8811

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The WorldNet Oslo offers both linear PCM and Enhanced apt-X coding options. Enhanced apt-X will deliver the same audio quality as linear with under 2ms delay and at a fraction of the data rate. Other options include MPEG L2, J.57 and J.41 companding. With four channels of audio per plug-in module, up to seven audio modules per unit, and a choice of over 20 different audio modules, each WorldNet Oslo has the capacity of up to 28 mono channels / 14 stereo pairs.

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Throw your terminal screwdriver in the trash can!

No Dip Switch settings here - configuration and control of the WorldNet Oslo is straight-forward and simple thanks to APT's powerful and intuitive Codec Management System (CMS). Offering extensive real-time management of multiple codec units, the CMS enables alarm monitoring, logging and performance monitoring as well as configurable user and audio profiles.

apt

soundconnections

For more information, call APT on 800 955 APTX or 617-923-2260 or check out www.aptx.com

NAMM SHOW NEW PRODUCTS

by Erin Shipps, associate editor

Rugged optical connectors
Neutrik
Booth N9029

IP65 Rated Opticalcon: This connector has been upgraded for dust-tight and water-jet protection. The upgrade extends the outdoor capabilities of the Opticalcon, making it ideal for mobile trucks and other outdoor broadcast settings. It is especially useful when weatherproof extensions are required. To achieve a water-resistant IP65 connection, additional seals and gaskets have been employed on the chassis connector and cable ends. For earlier versions of Opticalcon cable connection systems, an upgrade kit is offered so it can be combined with the upgraded NO2-4FDW chassis connector or NAO2-4S75W coupler. The new cable ends are 100 percent backward compatible and work with all existing chassis installations.

732-901-9488; www.neutrik.com; info@neutrikusa.com

Aluminun and steel panels
APW Enclosures
Booth SU5613

Filler panels: Stantron's aluminum and steel filler panels are rack-mounts, designed to block open spaces and provide a continuous and clean look to the front of the rack. They come in a wide range of styles and sizes and are compatible with Stantron's E-Rack and Presentation Racks. Stantron manufactures both varieties of panels in solid, perforated, vertical or horizontal-slotted styles, as well as solid flat designs. The aluminum filler panels come in a variety of finishes to complement existing equipment and rack configurations. Stantron's steel filler panels are available in four standard colors.

714-634-7300; www.apw.com
sales.us@apw.com

Baffles, panels and tiles
Pinta Acoustic/Sonex
Booth N6814

Sonex Clean: These products are designed for environments that require noise control with durable, washable and cleanable materials. The baffles, panels and ceiling tiles are fully encapsulated in FR taffeta vinyl for efficient cleaning and long, durable life. They offer excellent sound absorption to reduce reverberation or echo. The Sonex Clean line is made from Pinta Acoustic's Willtec melamine foam core, which is ASTM E84 Class 1 fire-rated. Products are available in 17 standard colors. The baffles and panels are 24"×48"×2" and the ceiling tiles are available in 24"×24"×2" or 24"×48"×2".

800-662-0032; www.pinta-acoustic.com
sales@pinta-acoustic.com

Audio mixers
Avlex
Booth N5232

Bardl P Series: Available in 8- and 12-channel configurations, the Bardl P Series offers features including general sound reinforcement for fixed and portable use, as well as integration into desktop recording environments. The series provides USB and S/PDIF digital connectivity. The integrated USB capability facilitates bi-directional audio communication with software-based recording applications while the S/PDIF port enables a CD recorder or other similar equipment to serve as a mastering deck. Two of the same model mixers can be cascaded for increased channel capacity via a 9-pin D-Sub connector. The P Series mixers also provide switchable mic/line inputs. Primary input channels have three-band shelving EQ while a contour control is provided for Groups 1, 2 and the main L-R stereo bus. For monitoring, the mixers provide separate headphone level control and direct output control for recording.

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web: www.musicamusa.com

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Compact console
Audioarts Engineering
Booth N7612

Air 1: The Air 1 Console is a compact console that retains the features and interconnect system of much larger consoles. The low profile frame, only 2 1/2" high, comes equipped with rugged tabletop mount with direct access TRS rear connectors. The unit features dual program busses, a cue that can automatically interrupt the main monitor when activated, a switchable meter, monitor source selection, dual monitor outs, long-throw faders, dual mic preamps, solid-state illumination, a built-in headphone amp and an on-air indicator.

252-638-7000; www.wheatstone.com
sales@wheatstone.com

Antenna
Jampro Antennas
Booth C2607

JHPC Penetrator: The JHPC-HD antenna is the medium-power version of the Penetrator antenna. This model has been designed for use with HD Radio broadcast. Rated at 50kW maximum input, each bay consists of a Penetrator-style radiating element with a 3 1/8" shunt feed line. Each JHPC is factory tuned to any frequency in the FM Band II (87.5-108MHz) range on a tower structure that best simulates the customer's actual tower. Multiple frequency design is also available. The antenna is constructed of the highest quality marine brass and copper. A hot-dipped, galvanized-steel mounting bracket for utmost grounding supports each bay. Standard round leg mounting brackets for uniform face towers are included with each antenna.

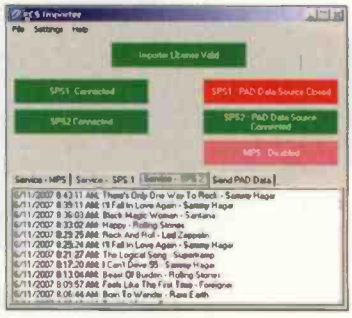
916-383-1177; www.jampro.com
jampro@jampro.com

Vertically polarized FM antenna
Propagation Systems Inc
Booth C2324

PSIFLV: For broadcasters searching for a low-power antenna that is omni-directional and vertically polarized, the PSIFLV is constructed of marine brass and copper. It accepts deicers or radomes for environmental protection. Each element can handle input power of up to 500W and provides excellent bandwidth and low VSWR <1.1:1. This antenna is factory tuned and mounting hardware is included.

814-472-5540
www.psibroadcast.com; sales@psibroadcast.com

Interactive digital importer
RCS
Booth N5917



HD Importer: RCS HD Importer supports the recently launched HD Radio application of Itunes Tagging. The RCS HD Importer enables advanced radio capabilities, like multicasting and datacasting. As a stand-alone module, it can work with any automation system and integrates seamlessly with Nexgen. The Importer is made up of several core applications that need to start in a certain order. It aggressively manages those applications and their configuration settings, keeping them running at all times.

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RemoteAmp Blue allows IFB monitoring through a cell phone equipped with Bluetooth Wireless Technology. This is a listen-only device designed for voice IFB or full-bandwidth stereo music listening. The line input jacks and separate volume controls allow wired operation in parallel with the Bluetooth connection.

RemoteAmp Two provides a wired, listen-only connection for mono IFB or full bandwidth stereo music listening. Separate volume controls for the XLR and 3.5 mm line input jacks allow a simple mix of mono and stereo sources.

Each has a powerful ½ watt stereo headphone amplifier that will cut through any crowd noise. **BluePack** and **RemoteAmp Blue** also pair to Bluetooth-equipped sound cards and music players in full-bandwidth stereo A2DP mode.

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Wireless mic system
Lectrosonics
Booth N5223

UTPR20: Combining the capsule from Heil Sound's PR20 dynamic microphone with the Lectrosonics UT Series Digital Hybrid Wireless transmitters, this system provides compandor-free audio for wide dynamic range and neutral frequency response. The UTPR20 uses Heil's grill basket and capsule identifier ring features a gold-colored battery access ring to match the unit's overall color scheme. The transmitter features 100mW RF power, 256 synthesized frequencies and full compatibility with Lectrosonics Digital Hybrid Wireless and analog receivers.

800-821-1121
www.lectrosonics.com; sales@lectrosonics.com

Audio analyzer
Sencore Electronics
Booths N1122 and SU12108

SP495 Sound Pro-EX: The SP495 Sound Pro-Extreme is an update to the SP495 Sound Pro. Improvements include a 1/30th octave FFT analyzer, doubling the ETG resolution, an early decay time (EDT) measurement, noise criterion curves overlay on the RTA, improved speech intelligibility STIPA with memory to store test results, improved multi-band decay test performance with memory to store test results, the addition of noise criteria computation per-band, 140V and 200V computation to the impedance meter and LEQ measurements to the TDA. New functions include the auto home theater (Auto HT), auto time delay, macro, card memory logger, and clear memories/macro functions.

800-SENCORE; www.sencore.com
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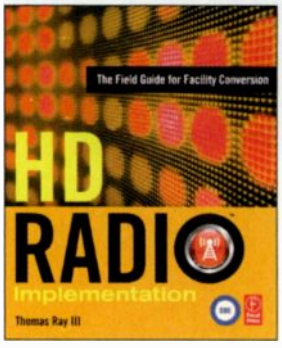
Digital radio installation resource
Elsevier/Focal Press



Booth L27 (SBE)

HD Radio Implementation: Published in cooperation with the Society of Broadcast Engineers, this book offers a hands-on approach to HD Radio implementation. It covers the elements of the radio station signal chain that need to be upgraded or modified to better accommodate the needs of the transmission system.

781-221-2212; www.elsevier.com
usbkinfo@elsevier.com

Portable remote control
Burk Technology
Booth N6920

ARC Plus Mobile Web Connectivity: Burk Technology has added mobile Web connectivity to the ARC Plus broadcast facility control system. The ARC Plus now allows remote access via Web-enabled cell phones and PDAs, including Treos, Iphones, Pocket PCs and more. Virtually any device with Web access can now monitor transmitter readings and allow remote operators to review alarms, issue commands and run macros without a PC.

800-255-8090; www.burk.com
sales@burk.com

User-programmable studio control panels
Axia Audio
Booth N7620

Smart Switch: Smart Switch panels interface with Axia's Pathfinder PC router control software to provide on-air hosts, guests and producers with one-touch activation of router salvos, studio switching operations, audio equipment control, custom intercom functions and more. These panels are 1RU devices available in either 9- or 17-button configurations. They contain back-lit, dynamic LCD displays, and are designed for convenient placement next to host or guest mic positions. Smart Switch panels connect to Axia networks via Ethernet.

216-241-7225; www.axiaaudio.com; inquiry@axiaaudio.com

HD Radio audio codec
APT
Booth N8811

Worldcast Horizon HD: The Worldcast Horizon HD is a duplex stereo codec that enables broadcasters to deliver FM and HD content from studio to transmitter site. With both a T1 and ethernet interface, broadcasters can utilize existing T1 links for the FM transport and send their HD content as a UDP stream embedded in the T1 link, eliminating the need for additional bandwidth and cost. At the remote/transmitter link, the HD content is presented back on an IP port. As a fully duplex device, WorldCast Horizon HD allows off-air monitoring or an independent channel to backhaul RPU feeds and satellite down-linked audio. An RS-232 port is available for PAD and contact closures for remote control.

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Bluepack: JK Audio combines Bluetooth Wireless Technology with professional audio electronics in a rugged new belt-pack design. Bluepack allows live man-on-the-street interviews through a cell phone equipped with Bluetooth wireless technology. Bluepack pairs to a cell phone like a Bluetooth wireless headset. A 3.5mm stereo line input jack allows recordings to be mixed into the broadcast. The 3.5mm stereo line output jack provides a full bandwidth microphone signal on the left channel and Bluetooth audio on the right channel. The stereo headphone output gives a mix of the XLR microphone input, 3.5mm input and Bluetooth audio. The 0.5W stereo headphone amplifier will cut through any crowd noise. Bluepack also pairs to Bluetooth equipped sound cards and music players in full bandwidth stereo A2DP mode.

800-552-8346; www.jkaudio.com; info@jkaudio.com

Software-based Intercom

Clear-Com Systems

Booth C5908

Concert: Based on voice-over-Internet Protocol (VoIP) technology, Clear-Com Concert's audio features deliver crystal-clear sound and non-blocking communications over a standard Local Area Network or Internet between local and remote users. This eliminates the need for picking up a phone or connecting to a conference bridge. The interface facilitates ad-hoc communications and makes the user the conductor with easy drag-and-drop functionality. The Presence Awareness feature monitors participants' availability. The system, scalable to many hundreds of users, is designed with proprietary protocols and error recovery algorithms for mission-critical applications that include live broadcast productions. The system uses AES 128-bit encryption to provide security for protecting privacy of transmissions. Supporting a variety of codecs including wideband and ultra wideband, Clear-Com Concert provides realistic audio.

510-496-6666; www.clearcom.com
sales@clearcom.com

AM reference receiver and modulation monitor

Inovonics

Booth N5829

Model 525: The 525 is a frequency-agile, wideband AM-broadcast receiver that utilizes a highly linear phase-locked detector to provide accurate off-air measurements of AM carrier modulation. An important feature of this new monitor is the ability to resolve the amplitude-modulation component of the station's carrier during IBOC Hybrid Digital broadcast operation. Positive and negative carrier modulation is shown simultaneously on a high-resolution LCD display. This can be switched to provide a read-out of received signal strength and asynchronous noise as well, two parameters that can influence the modulation reading. Measurement response is pancake-flat to 10kHz, although a menu-controlled low-pass filter in the audio-monitor output provides a cutoff that can be programmed between 10kHz and 2kHz in 1kHz steps. This allows the user to preview the sonic compromises imposed by pre-transmission audio filtering and to simulate the response of consumer radios.

800-733-0552; www.inovon.com; info@inovon.com

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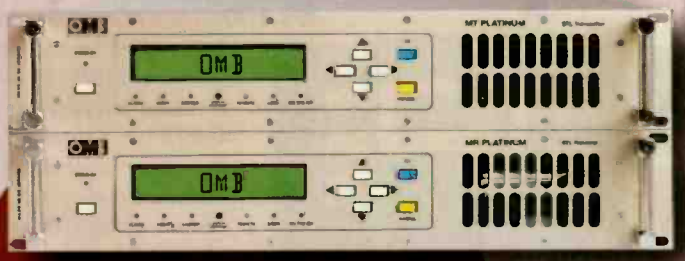
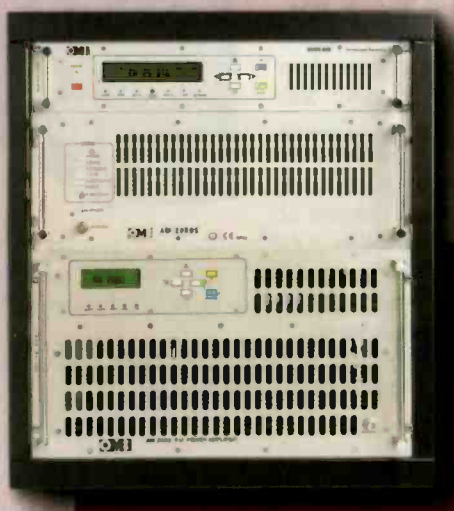


MT/MR PLATINUM >1GHz

is a high-performance Studio-to-Transmitter Link. It is made up of the 5W MT transmitter externally synthesized in 0MHz sub-bands with a step of 100KHz, and the MR double conversion receiver, that is externally synthesized, too. The MT is microprocessor controlled, and includes LCD display for the visualization of the most relevant transmission parameters (frequency (6-digit), forward and reflected power, modulation level), balanced Mono, Stereo (MPX). The MR receiver has the same visualization system as the transmitter. It includes balanced Mono and Stereo (MPX) outputs. Furthermore, the MT/MR Platinum STL includes a jumper in order to get a proper operation with digital signals.



EM 2000

is a 2000W FM transmitter made up of the EM 25 DIG exciter (or EM 20/30 exciter) and the AM 2000 FM amplifier. AM 2000 includes eight 300W high-efficiency MOSFET technology amplifying modules, fed by 2 independent switching power supplies, which are made to withstand the working conditions. The amplifiying modules work independently thanks to a power combining structure that provides high isolation between them.

EM 10000

is a 10000W FM transmitter made up of the EM 250 COMPACT DIG exciter and three control units which combine the power of six AM 2000 FM amplifiers. AM 2000 includes eight 300W high-efficiency MOSFET technology amplifying modules, fed by 2 independent switching power supplies, which are made to withstand the working conditions. The amplifying modules works independently thanks to a power combining structure that provides high isolation between them.

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2.4GHz transmitter
Zaxcom
Booth N4533

IFB 100: This 2.4GHz IFB transmitter is designed for use with the company's TRX900 series of wireless microphone systems. The IFB100 transmits a single RF carrier containing timecode, IFB audio, and remote control signals to any number of TRX900 wireless systems, giving users the ability to record a group of channels and play them back as a virtual multitrack recording. The unit directly accepts SMPTE timecode and balanced audio at line level. The IFB100's 125mW RF power output provides a range of up to 300 feet for transmission of IFB audio and up to 1,000 feet for timecode transmission and remote control signals. Its 2.4GHz frequency can be used license free in almost any country.

973-835-5000; www.zaxcom.com; info@zaxcom.com

Stand-alone disc duplication
Vinpower Digital
Booth SL 7709

Xerox-branded tower duplicators: The Xerox-branded standalone tower duplicators offer the ability to transfer image files directly from a PC to the duplicator's internal hard drive, saving users the time and effort usually required to produce a master disc for each project. With a range of SKUs from 1:1 up to 1:15 writer drives, each operates independently of a PC. Several models in the line also provide users with a 256MB buffer memory for more stable and reliable high-speed duplication with a greatly reduced occurrence of buffer underrun errors.

626-282-3300; www.vinpowerdigital.com
info@vinpowerdigital.com

Powered monitor
Adam Professional Audio
Booth N8120

A5: This powered monitor uses the same technology as existing Adam monitors. Technically, the A5 is a smaller version of the A7, and can be used either in stereo or to fill out a 5.1 surround system. Powered by 2x25W on-board amplifiers, the A5 combines Adam's accelerated ribbon technology folded ribbon tweeter with a 5" woofer constructed of a carbon fiber and Rohacell sandwich. The front of the A5 sports dual ports for low frequency response down to 55Hz, metal grills for added durability, as well as power and gain controls. The rear panel includes balanced XLR jacks, unbalanced RCA jacks and Stereolink, which connects speakers with input and output jacks, allowing the user to control the overall volume of the system from any one speaker's gain control.

818-991-3800; www.adam-audio.com; usa@adam-audio.com

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Middle Atlantic Products
Booth SU10205

WMRK: WMRK Series configured server enclosures are manufactured with features that simplify cable, thermal and power management when mounting servers or other data-specific components, and save installers time and energy when ordering, installing and servicing components. The enclosures are designed to house servers from all manufacturers. They are also designed to facilitate passive thermal management with perforated front and split rear doors and a Decora temperature-monitoring module for local monitoring; as well as electrical knockouts on rear plates, removable split bottom rear panel, integral gland grommeted cable pass-throughs and a 42-space vertical lacing strip.

973-839-1011; www.middleatlantic.com
sales@middleatlantic.com

Surround headphones
Beyerdynamic
Booth N7917

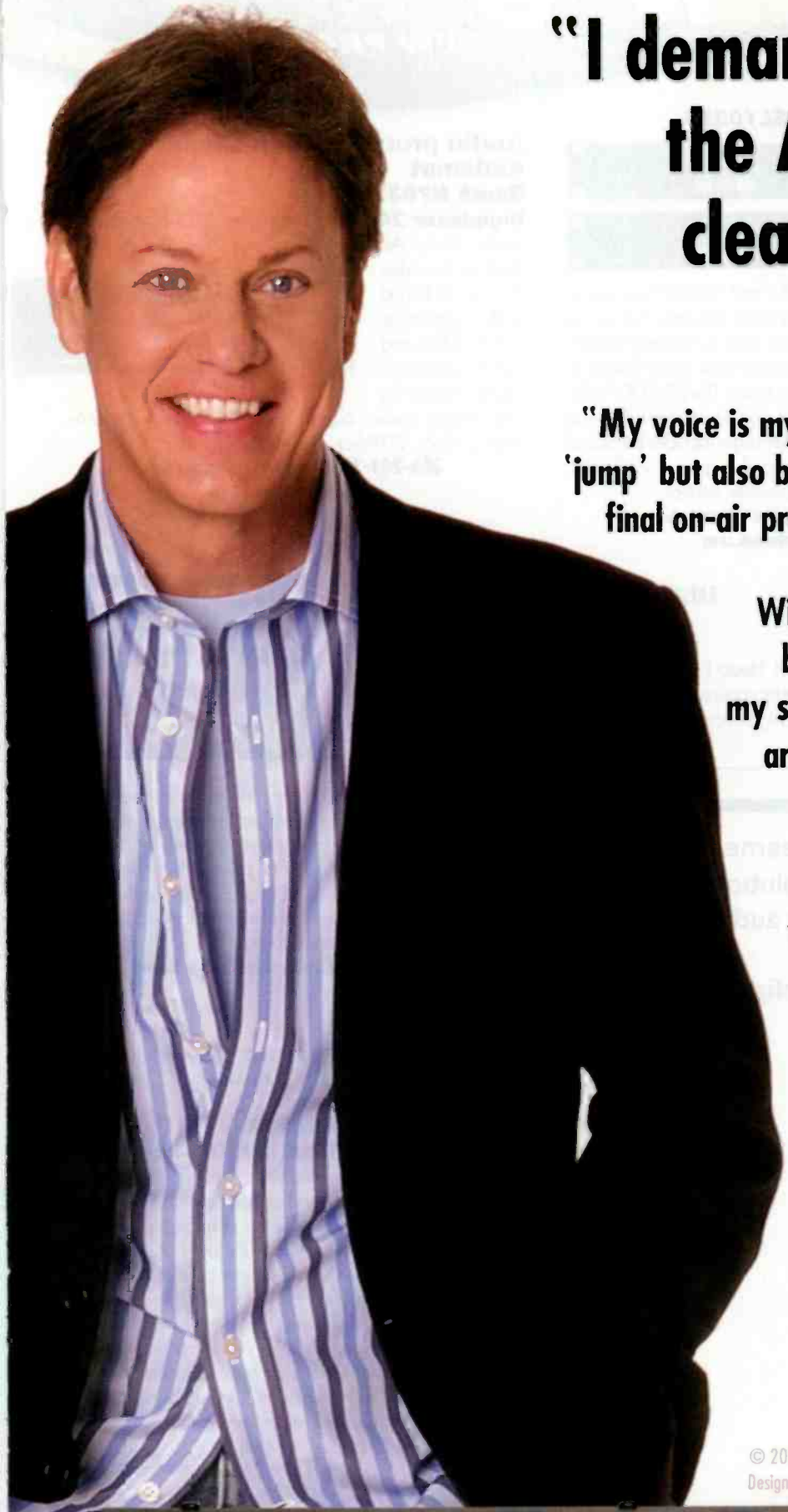
DT 770 PRO HT: Beyerdynamic unveiled the Headzone PRO, a 5.1 portable surround monitoring system, at NAB2007. To provide additional sound isolation in noisy remote environments, the Headzone system has been extended to include the closed DT 770 PRO HT headphones. Thanks to Binaural Environment Modeling technology and patented ultrasonic headtracking, Headzone recreates a 5.1 surround monitor setup utilizing conventional stereo headphones. To further create the feel of a traditional 5.1 setup, the Headzone uses an ultrasonic headtracker to detect the position of the listener's head and adapts the source audio playback accordingly.

800-293-4463; www.beyerdynamic-usa.com
info@beyerdynamic-usa.com

Studio automation platform
Broadcast Electronics
Booth N7007

Audio Vault X: Audio Vault X is a new studio platform with more capacity and flexibility for adding and personalizing format channels on the Internet or over the air. New capabilities enable broadcasters to do more with less. Audio Vault X handles complex segues, such as inserting donut tracks in the middle of a song, station ID or weathercast. It can even do double donuts – that is, drop in a voice or other segment in the middle of a song or jingle twice. Plus, talent can manipulate fades, audio levels and other characteristics of segues. Personalized studio mashups combine studio tools and applications from more than one source into a single workspace. The Audio Vault X multi-thread engine enables talent in separate studios to work collectively off the same log, in real-time. BE's Audio Vault X is a complete digital media system for live-assist or satellite operation. The new platform offers on-air studio operation as well as a full suite of scheduling and production modules for multi-track editing, music rotation and ad insertion.

217-224-9600; www.bdcast.com; bdcast@bdcast.com



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Solid-state recorders
Tascam
Booths SL9623 and SL10328

SS-R1, SS-CDR1: The SS-R1 and SS-CDR1 are a pair of single-rack-space Compact Flash recorders. Designed for applications that previously used a cassette or Mini Disc recorder, the SS-R1 records in WAV or MP3 formats to achieve extended recording times. The SS-CDR1 adds a slot-loading CD recorder for additional recording and audio transfer functions. Each recorder includes balanced and unbalanced audio inputs and outputs, RS-232 and parallel control and a wired remote control.

323-726-0303; www.tascam.com
tascamlit@tascam.com

Audio processor and exciter
Audemat
Booth N7932

Digiplexer 246: The Digiplexer 246 processor is similar to the Audemat Digiplexer 2/4, but the 3RU device provides 2- or 4-band audio processor with an RBDS and stereo encoder, audio back-up and remote control capabilities. The unit is available with a 10W, 20W or 100W exciter.

305-249-3110; www.audemat.com
contact@audemat.com

High output headphone box
Aphex Systems
Booth N5617

Head Pod: The Model 454 Head Pod is a four-output headphone distribution pod with four independent stereo power amplifiers. Working with any headphone type, the Head Pod provides high headroom to avoid clipping even at high sound levels. Designed for ultra low distortion and wide frequency response capabilities, the Model 454 also allows for longer listening sessions without listener fatigue. Individual volume controls are provided for each of the four headphone outputs, made with durable metal jacks. A master volume control on the Head Pod allows for matching of source levels. It connects to any analog source, with a rear panel selector switch to choose between balanced sources (discrete left and right) and unbalanced sources.

818-767-2929; www.aphex.com; sales@aphex.com

Microphone capsule heads
Neumann
Booth N8207

KK 131, KK 143, KK 145: Neumann has doubled the number of modular capsule head options for the KM D miniature digital series of microphones with the introduction of the new KK 131 (free field-equalized omni-directional), KK 143 (wide cardioid) and KK 145 (cardioid with low frequency roll-off), which are all derived from the analog KM 100 and KM 180 series. The three new modular capsule heads join three existing options for the KM D microphone series, which locates Neumann A-to-D converter technology in the microphone immediately next to the capsule. Like these latest additions, the existing KK 183 (diffuse field-equalized omni-directional), KK 184 (cardioid) and KK 185 (hypercardioid) capsule heads were also developed from the KM 100 and KM 180 series of miniature analog microphones. The KM 100 series AK capsule heads are not interchangeable with KM D series capsule heads.

860-434-5220; www.neumannusa.com; neumlit@neumannusa.com

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www.arrakis-systems.com

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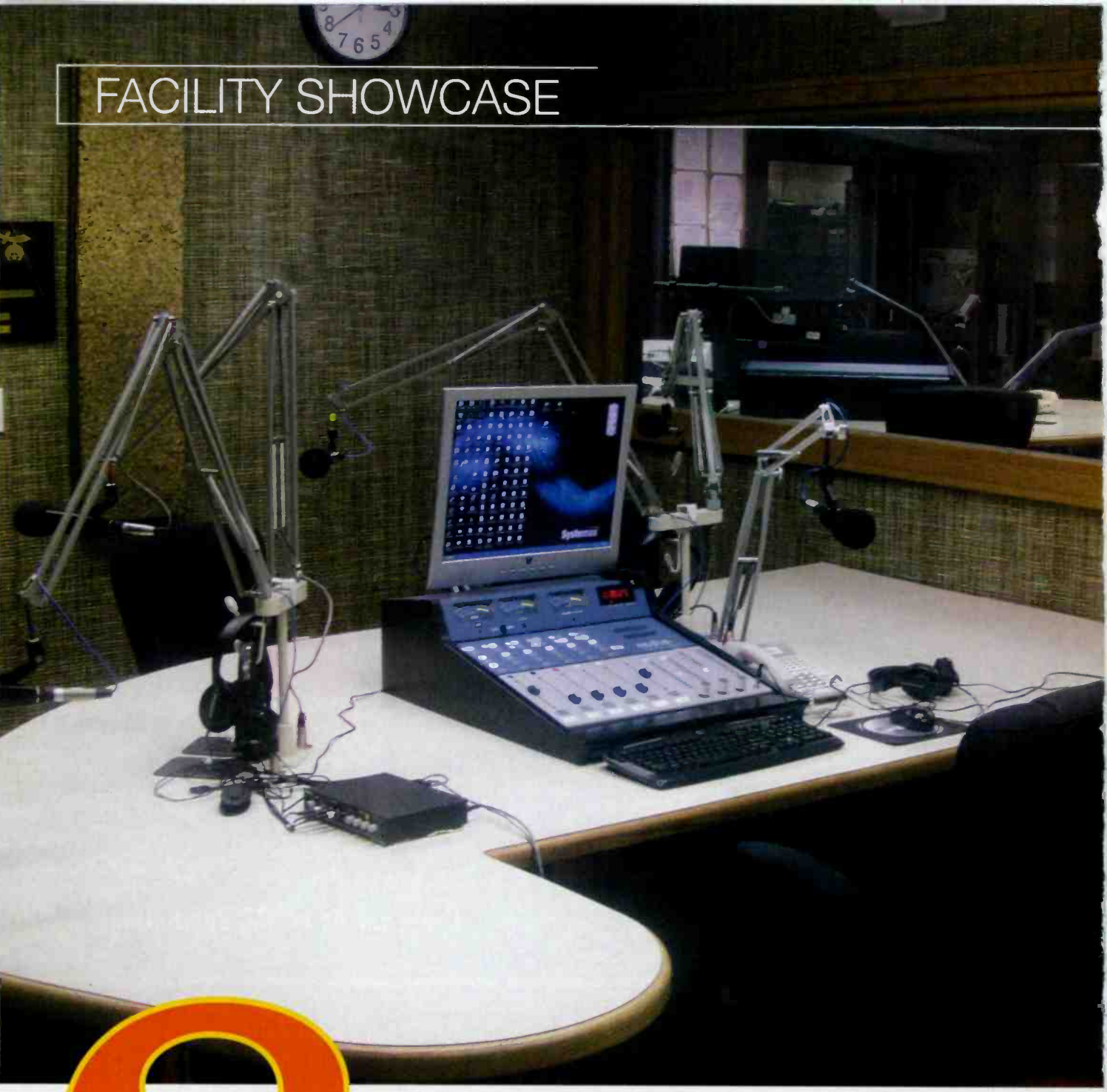


Streamlined Station

By Dave Lyons

WHKP-AM updates 20-year-old equipment

WHKP is a locally owned and operated, stand-alone, heritage, AM station serving Hendersonville, NC, since 1946. WHKP has always been innovative, especially when it comes to marketing the station. In addition to our broadcast services, we also stream our audio content over the Web and on a local cable access channel. We print a local news and advertising magazine, called *Peaks*, and feature a large, full-color LED sign in front of the station for advertising and station promotion.

When I first assumed the chief engineer duties at WHKP, I inherited a facility that, while beautiful and functional, was aging as far as broadcast equipment. The station had been using the DOS-based RDS/Phantom automation system, since the mid-1990s, for traffic and play-out of commercial material. The station featured mostly live programming from 5 a.m. to 6 p.m. with a satellite-delivered music service at night and on weekends. However, during live programming, the announcers were playing music from PCs, CDs and even records. The control room's Harris console was at least 20 years old, and the production room had a 10-year-old LPB board – both with no spare parts available.

After expressing my concerns to Art Cooley, president and general manager, it was decided that we would undertake a general upgrade of our broadcast facilities, with an eye toward eventually broadcasting in HD Radio. We then toured some area radio stations that had recently upgraded their facilities with new equipment and automation systems. After careful study we eventually decided on Google's Maestro system for our automation and Radio Systems for our consoles. The Maestro system consists of two audio servers, one assigned to control, the other to production. Since our old automation system also served as a traffic system, we replaced it with Natural Soft's Natural Log 8 and also purchased their music scheduling program, Natural Music 5.

In addition to the control and production rooms, we had a conference/studio room with a table and microphones (used primarily for group interviews) and a desk with a small mixer and equipment (capable of limited production, mostly for news gathering). We decided to upgrade this room as well, giving it full production capability. Our audio editing software was upgraded to Adobe Audition.

The control room features a 12-channel, Radio Systems Millenium Digital console, while the conference and production rooms would each have the six-channel versions. The existing patch panels were replaced with Radio Systems Studio Hub+. The control telephone system was upgraded with a new Telos 1x6 digital hybrid system.

We also opted for new furniture in the control and conference rooms. We wanted a stylish, modern-looking effect, that wouldn't break the bank. We were fortunate to find a local cabinet maker who took our ideas and came up with a very sturdy, nice-looking, functional design. The two rooms feature identical furniture.

Again, with a plan to eventually implement HD Radio, we replaced our 20-year-old Harris SX-1A transmitter with a new, digital-ready Harris DAX-1R. The SX-1A was moved to standby duty.

The turntable's slide-out drawer is mounted in the equipment rack built into the studio cabinet.

Getting started

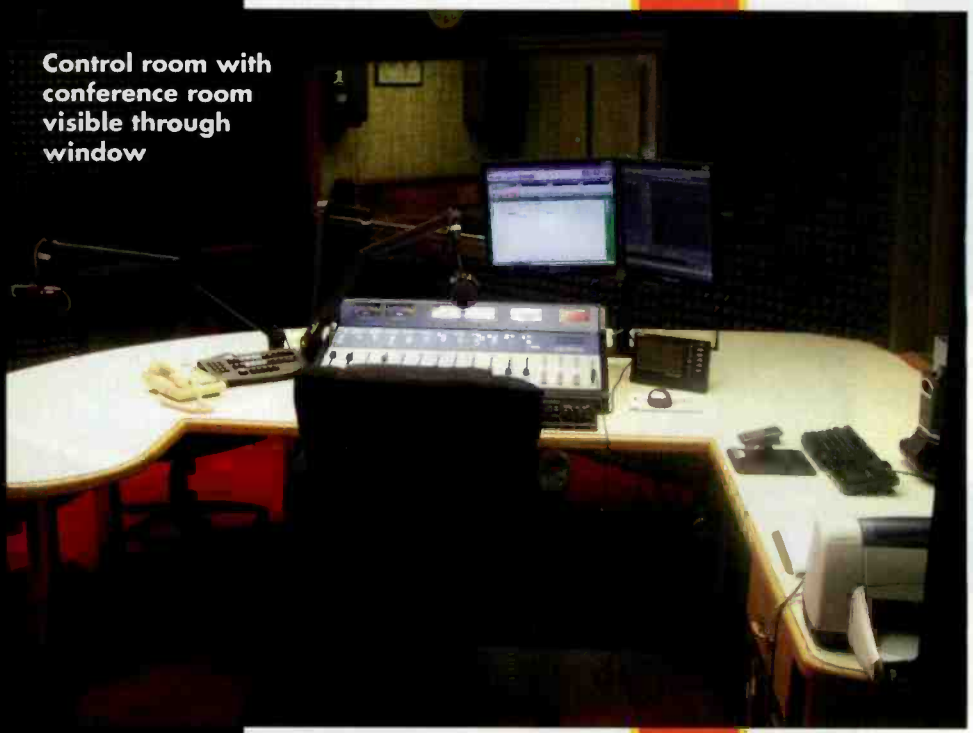
The greatest challenge we faced was completely gutting our existing studios while keeping the station on the air. Our plan was to rebuild the studio/conference room first (giving us some production capability), then rebuild the production room, and finally, move control to the newly rebuilt production room, while the control room was being rebuilt. The control room was completely stripped of every piece of equipment, cabinet, shelf and carpet. Two existing equipment racks were also moved out of the room to a glass-doored closet in the control room, creating less clutter and more room, while still giving the announcers a view to the metering.

Another issue was that our existing LAN was built with CAT-3 wiring and was not capable of handling the traffic of our new Maestro system. Instead of completely rewiring our network, we ran a new CAT-5 network strictly dedicated to the automation system. This would be a closed network with no Internet access to assure no chance of virus infections or hackers.

All existing wiring between all three studios was pulled out and replaced with 25-pair CAT-5 cabling, utilizing punch-down blocks. In addition to the three studios, a central engineering station, consisting of two racks, is tied to all three rooms. The two racks contain the Maestro servers and associated equipment, as well as the five satellite receivers we use for programming content. With this configuration, we are able to patch any audio feed, to any room, using either our Broadcast Tools audio switchers or RJ-45 patch

Production room

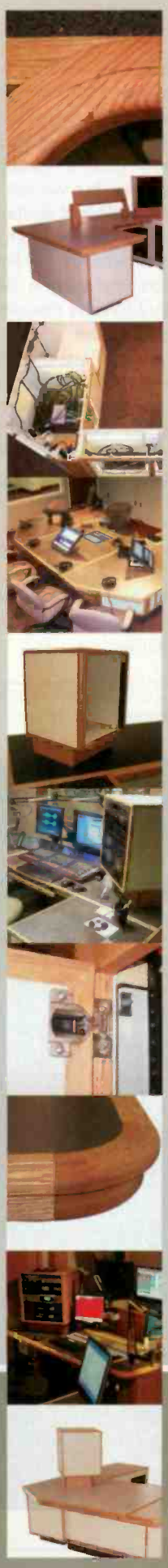
Control room with lobby visible through window



Control room with conference room visible through window

Equipment List

Adobe Audition
Aircorp Pro Announcer
Arcview dual monitor arms
Armstrong Xlink
Audioscience 1024
Auralex acoustic foam
Belar AMMA-2
Belkin 8-Port Gigabit Switch
Belkin Wireless G Router
Broadcast Tools SS 16.4, ACS 8.2 Plus
Cyber Power UR500 UPS
DBX 166A, DBX 286A
Dell PCs
EVO LCD Monitor Arms
Google Maestro
Harris DAX-1R
JK Audio That-2
Middle Atlantic racks
Natural Soft Natural Log 8, Natural Music 5
Ortronics RJ-45 Patch Panels
Radio Systems Millenium, Studio Hub+
RDL RUDA4D
Sennheiser MD421U
Siemon SIE-S66M150
Telos 1x6
Yellowtec Mika



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panels in the central engineering racks. This gives us the ability to put any of our three studios on the air with some simple patching and/or routing.

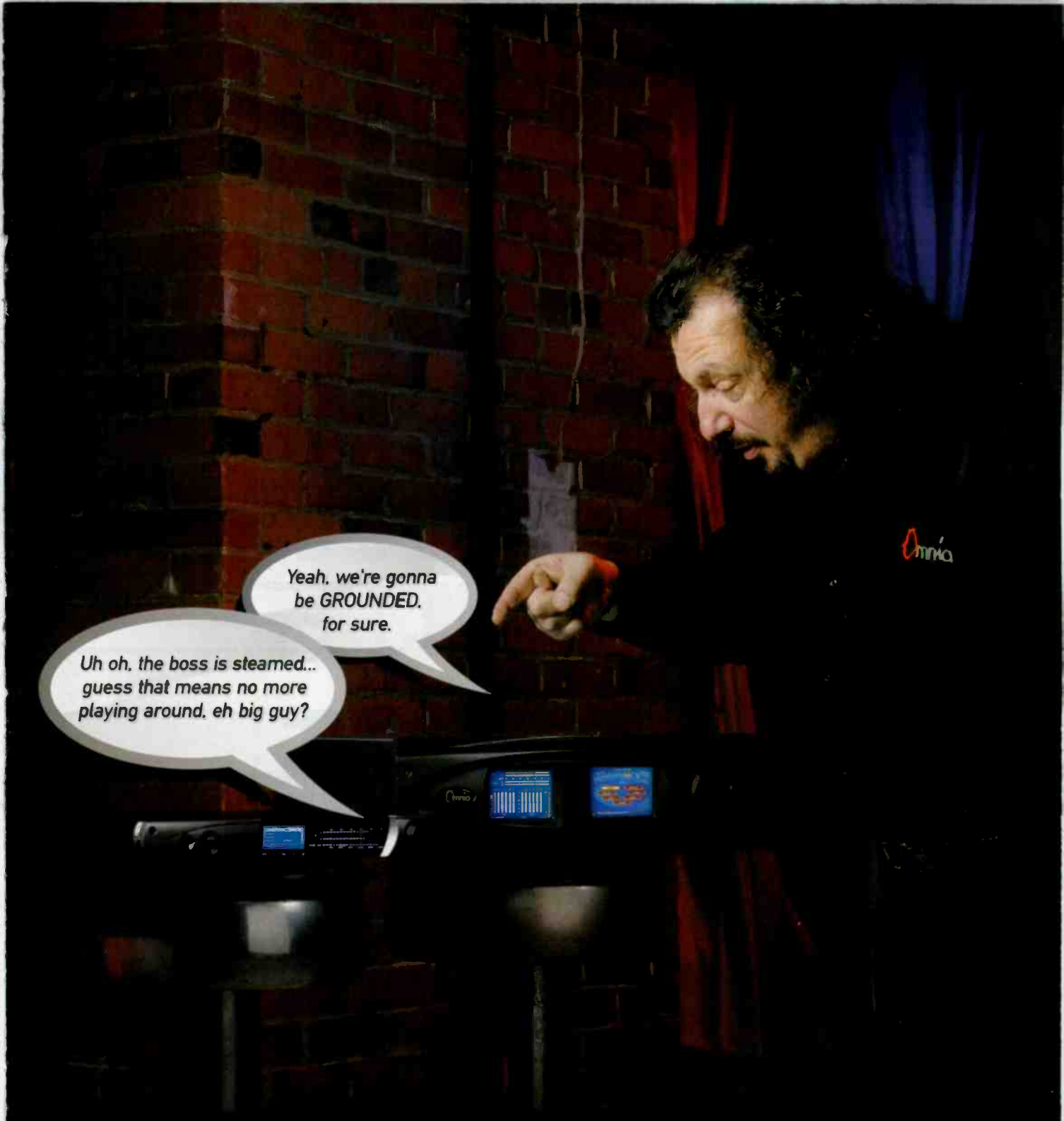
Our GM wanted a clean, uncluttered work area, with as little surface-mounted equipment as possible. So we put all rack-mountable equipment in racks that were built into the studio furniture cabinets. This included a turntable encased in a slideout drawer and mounted in the cabinet equipment rack. The cabinets in the control and conference rooms are open on one end to allow easier access for wiring and repairs. And the desk areas of the furniture are completely open underneath, giving easy access to wiring. In the rooms with new furniture you will only see the console, telephone, mic and monitor stands, and keyboards on the desktops. The control and production rooms were also fitted with Auralex foam panels for sound dampening.

Equipment racks in central engineering, containing Maestro Automation on the left and satellite equipment on the right.

We started the project in June 2007 and went on the air from our new control room on Nov. 21, 2007. While the studio work is mostly complete, we are still working on our new transmission equipment. The new DAX-1R transmitter is operating, but we are still installing additional equipment that will be needed when we decide to make our HD Radio transition, including a Belar AMMA-2 Wizard modulation monitor/analyzer.

While WHKP has embraced the future of broadcasting, we haven't forgotten our past. In our station lobby, we have quite an impressive collection of antique radios, broadcast equipment, records, pictures and news articles, mostly donated by area residents. Now in our 62nd year of broadcasting, we're an AM radio station looking forward to tomorrow.

Lyons is chief engineer of WHKP-AM, Hendersonville, NC.



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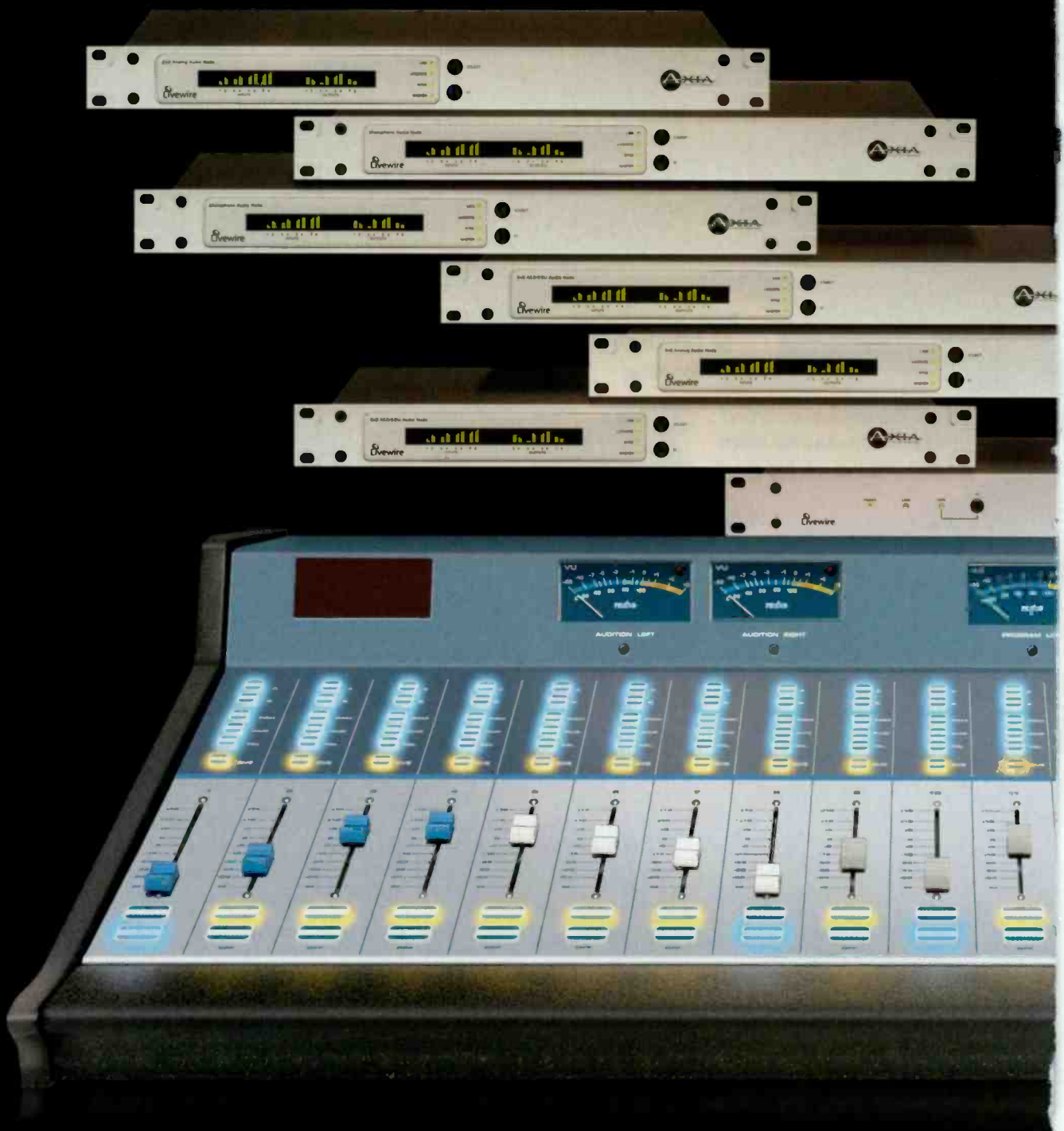

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Tips, tricks, hints and more

By John Landry, CSRE

Vehicle maintenance

If you have a remote van or truck with a pneumatic mast, now is the time to do some routine maintenance before the warm weather and remote season starts up. Maintenance includes a general cleaning of the sliding sections with a non-abrasive cleaner (such as lacquer thinner or alcohol) and lubrication. The mast should be slowly extended a section at a time, inspected and wiped with the cleaner. Care should be exercised not to scratch the surface of the tubing sections. Then put the recommended lubricant (or light machine oil) into the lubrication holes with the pressure removed. Oil should never be rubbed onto the outside surfaces, as this will attract dirt, eventually causing binding and expensive repairs. The mast should be stored with the drain valve open on the bottom to prevent condensation inside the tubing. Ice inside the tubing can cause complete failure and ruin a mast. A canvas cover is available to protect the mast during down times, which will lead to much longer service life.

Spring is the ideal time to check each vehicle's mast.

Check for leaks

Another chore for the spring at the transmitter site is to check the transmission lines for leaks. A quick and cheap way to look for leaks is to spray a joint (or suspected leak site) with foaming window cleaner. Any leak will cause the foam to bubble, making it obvious where the leak is. If your system uses a dehydrator on the air intake, the desiccant jar should be emptied and changed. Desiccant pellets can be redried in an oven and saved for reuse. Store them in a tight-sealing container such as a paint can. Any compressor pumps should be checked for noise and wear. If it needs to be replaced, a quick and inexpensive replacement is available from McMaster-Carr: part number 8280K26 oil-less air compressor. If you don't already have a spare on the shelf, now is probably a good time to get one.

It's easy to forget about an automatic dehydrator, until a problem arises.

A dead bug's life

The coming of spring and warm weather also means the return of birds and bees. While birds aren't a big deal, bees and wasps seem to love transmitter sites! Who hasn't surprised at least one nest of them when opening a transmitter building or an ATU doghouse door? And since stings are no fun, take time now to make your site wasp-proof. Buy a case of wasp spray and make sure a can is always near the door. Spray the perimeter of the shack door each visit during early Spring. If you have windows or insulator bowls on your shack, spray the outside and inside edges of them. Wasps and hornets have a keen sense of smell and don't like the smell of the wasp killer. Frequent spraying keeps the smell strong enough that they'll find another spot to build. If you have time, get some caulk and make things airtight. Any obvious holes from old feed lines or conduit should be filled. Good Stuff, a plastic expanding foam, will temporarily fill smaller holes and cracks. Make as many visits to the site as you can before the warm weather sets in, and if you do see a nest under construction return at dusk to spray it.

Landry is an audio maintenance engineer at CBS Radio/ Westwood One, New York.

Do you have a tech tip? Send it to us at radio@RadioMagOnline.com

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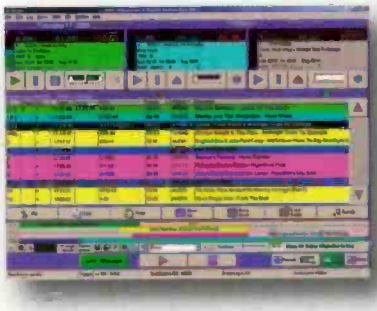
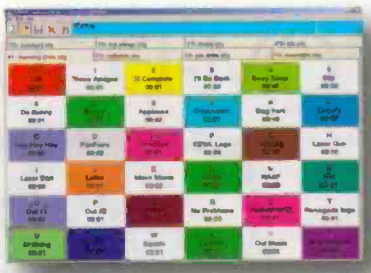
Radio Automation



Simian - radio automation and digital play out system

Instant Audio



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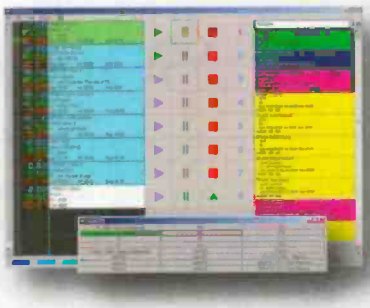

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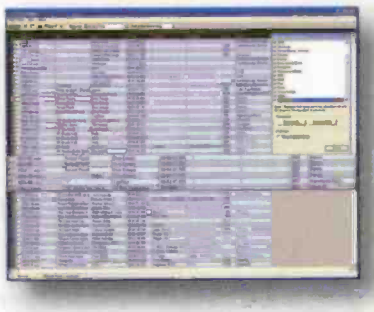

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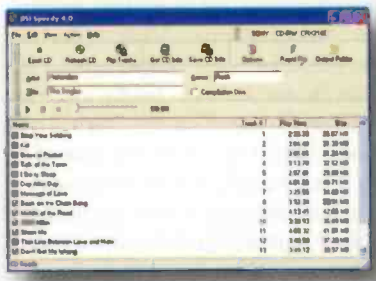

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BSI

Sony PCM-D50

By Scott Trask

I was looking for a recording device that didn't need a lot of setup, had built in microphones, was capable of recording in various formats and was easy enough for even an intern to operate. We are routinely visited by new talent who sing for the staff, and the programming department wanted an easy way to record the performance. We also needed a field recorder for live events and interviews.

We chose the Sony PCM-D50, which is capable of recording WAV files and MP3 playback. The device itself has two built-in microphones; an external handheld microphone can also be used, and there are connections to attach a line-level device.

Around the device

The PCM is roughly 6" by 3" and 1 1/4" thick. The case is made of aluminum, so it will withstand use on the road. The first items you notice are the two built-in microphones positioned at the top of the device. Below the microphones are the peak level LEDs for the left and right channels, then a display window just above the controls: menu, FF, pause, stop, record, play, divide, light, display and A-B keys. Each side has a turn knob level control. The one on the left adjusts the headphone level and the right adjusts the record level. Along with the record level on the right side is the power on/off switch, the line-in (analog and optical) and a Memory Stick slot. On the left side are the line-out (analog and optical), headphone jack, input select switch (mic or line), a mic attenuation switch (0dB or 20dB), USB connection, hold and digital pitch control (DPC), 6Vdc input jack and a remote jack.

The display shows time information, recording or playing status, level meter, memory stick indicator, folder and track numbers, recording mode, amount of memory remaining, power levels, and if the limiter, low cut filter and sync record are set to on.

Performance at a glance

Wide range of sampling rates

Record buffer

4GB internal memory

Memory Stick expansion slot

Built-in mics

USB Interface

The recorder can run on four AA batteries or a 6Vdc power supply, such as the included wall wart. The batteries are housed in a battery carriage that slides into the battery compartment. The battery life is approximately 20 to 26 hours of playback with standard batteries and 25 to 27 hours with rechargeables. The battery life for recording depends on the recording mode: The lower the sampling frequency and bit-rate the longer the battery life. As always, battery life depends on the operating conditions.

The recorder also includes four batteries, a USB cable and a copy of Sound Forge Audio Studio LE.

Recording functions

The unit can record using the built-in or external microphones. Files can be transferred via USB. The optional Memory Stick can naturally be removed and read with a card reader. The primary use of the built-in cardioid microphones is for recording musical performances. There are two positions for the microphones: 90 degrees (X-Y position) and 120 degrees (Wide Stereo position). Set the recorder six to nine feet from the sound source. The unit has a camera tripod thread mount on the bottom to simplify setup. The X-Y configuration works well for close miking to a small group. When recording a large group, such as a choir, set the microphones in the wide stereo position.

When an external microphone is attached, the unit displays the Plug-in Power menu. If plug-on power is needed, select on. The built-in microphones are automatically bypassed when an external microphone is used. The PCM-D50 can record from a line-level source using the 3.5mm line-in or optical input. Audio can be played directly from the unit via the 3.5mm line-out or optical output. Other features include digital pitch control, super bit mapping, and A-B playback repeat. Optional accessories include

a remote commander, tripod stand, microphone windscreen and the XLR-1 XLR mic adapter.

The default sampling rate and bit-rate is 44.1kHz/16-bits, but there are eight settings to choice from: 22.05/16, 44.1/16, 44.1/24, 48/16, 48/24, 96/16 and 96/24. This is easily done through the menu functions. A low-cut filter (LCF) of 75Hz or 150Hz can also be set in the menu. A limiter can be engaged for auto level control. The limiter recovery time can be set for 150ms, 1 second or 1 minute.

A prerecord buffer can be engaged to include five seconds of audio before the record button is actually pushed. This came in handy when I recorded some bands, because I did not always know when they were going to start. The extra audio at the beginning was a big help in editing.

Sony

P 800-686-SONY

W www.sony.com/professional

Using it

I found the device to be user friendly. The manual gives good instructions on how to set up and use the device. The display window is a good help, as it displays a lot of information to ensure the recorder is set up the way you want to record or playback. The multi-function I/O was a great help, and directly connecting to a DAT or MD player worked flawlessly. One of our on-air personalities used the unit to finally transfer his archived audio from DAT for editing and storage.

I used the recorder to record several musicians thanks to the Atlanta Institute of Music. I used every sample rate, LCF on and off, and mic attenuator on and off. The audio we recorded turned out very good. It was surprising that even audio recorded at the lowest sample rate was not bad. If Aerosmith was to appear on stage at a local establishment and I pressed the record button and then saw that it was set to 22.05/16, I would have no problem airing that audio. I felt the recordings in the wide stereo position were the best quality, as it picked up not only the musicians' audio but also the ambiance of the auditorium. It especially picked the high frequencies very well; listening to the cymbals just roll off was pleasant to my ears. I was a little disappointed in the X-Y microphone configuration, as I did not get the good stereo separation I was expecting.

If you plan on using the recorder for individual interviews, I recommend using a handheld microphone. We conducted several handheld interviews using the built-in microphones at 90, 120 degrees and pointing straight out, at about 4" to 6" inches from the person speaking. While the overall audio quality was good, we experienced some phasing and overdrive issues with the close proximity to the microphones. You could use just one channel for a mono track and eliminate the phasing issue, but the mics are still very sensitive to any plosives. Overall, I'm impressed with the PCM-D50. It's a good piece of equipment.

Trask is director of engineering, Lincoln Financial Media of Georgia, Atlanta.

Editor's note: Field Reports are an exclusive *Radio* magazine feature for radio broadcasters. Each report is prepared by well-qualified staff at a radio station, production facility or consulting company.

These reports are performed by the industry, for the industry. Manufacturer support is limited to providing loan equipment and to aiding the author if requested.

It is the responsibility of *Radio* magazine to publish the results of any device tested, positive or negative. No report should be considered an endorsement or disapproval by *Radio* magazine.

JK Audio Remote Mix 4

By Chris Wygal

When a radio station sets up for remote broadcasts, at least one friendly passerby (usually the know-it-all type) will inevitably stop and ask, "How are you getting back to the station?" While hybrid and codec technology may be troublesome to explain in layman's terms, the passerby's question is one that motivates radio engineers. "How do we get the audio from here to there?" Today, there are more and more unique ways to do it.

Ma Bell is a longtime friend of radio. She has afforded us equalized and dial-up lines over copper for decades. Even when we try to squeeze out every possible ounce of bandwidth using new-fangled codecs, she keeps us connected. But much to the broadcaster's delight, times are changing, and Ma Bell has made room for "Ma Cell." With the Remote Mix 4, JK Audio has introduced 21st century cell phone and Bluetooth technology to the remote broadcast clique. In this age of wireless-ness, it's a timely introduction.

Getting on the air

The Remote Mix 4 is a four-channel field mixer combined with a telephone hybrid. Setup of the unit is quick and easy, which is especially attractive to sportscasters. The Remote Mix 4 footprint is compact (taking up minimal space on press row), and the front-panel controls are straightforward and easy to use. The unit is powered by a 9Vdc wall wart power supply, or two 9V batteries, which can last as long as 10 hours. When the batteries are used in conjunction with Bluetooth, the Remote Mix 4 can be completely wireless. The inputs and outputs on the back panel are clearly labeled.

The Remote Mix 4 is designed with sportscasters in mind, who usually wear headsets. Each microphone adjustment is adjacent to its corresponding headphone adjustment. Gain controls for each microphone input allow for trimming the microphone levels, and each headphone has a control for adjusting volumes independently. A red clip indicator LED is present on each microphone channel. The four headphone controls each have their own toggle switches that route either the program mix, or the phone return feed to the headsets. If for example the play-by-play announcer is the only one who needs to hear the station break, he can switch his headset to return. The other three headsets, if switched to mix, will only hear the microphone channels. When a headphone is switched to return, some of the microphone mix can still be heard as well.

A master level control is situated under an LED meter that indicates program output between -30dB and +3dB. A cue level control adjusts the input level from an external audio source, such as a radio receiver, or a wireless interface, such as a cell phone. The cue feeds the headphones. Also on the front are the Bluetooth (more later) and the bass boost switches. Bass boost adds more lows to the program mix before sending the audio to the phone hybrid. A standard DTMF keypad and redial button are situated on the front panel, along with a hang up, dial/talk switch that seizes the phone line when making or receiving a phone call. A small red LED flashes to indicate an incoming call. The power switch for the Remote Mix 4 is on the front panel as well.

On the back

The back panel of the Remote Mix 4 is home to the program master output XLR plus a 3.5mm phone jack. The balanced XLR can feed a house PA or IFB system. The 3.5mm phone jack (TRS) carries the program mix on the left, and either of the receive or cue mixes on the right. This is handy for connecting to a small portable recording device. An RJ-11 jack accepts a standard phone line for typical landline usage. A switchable universal handset interface allows for the connection of the Remote Mix 4 to the coiled handset cable on a telephone where analog phone lines are not available. The Remote Mix 4 essentially becomes the earpiece and mouthpiece of the telephone.

Performance at a glance

- Bluetooth technology
- Interface with standard cell phone
- True telephone hybrid
- Bass boost for warmer sound
- 9V battery operation
- 48V phantom power on each channel
- Small and compact
- Cue input for monitoring external audio source
- Quick and easy setup

A 1/4" jack is selectable between either an external cue input (from a radio receiver, for example) or the wireless phone interface. When using the supplied 2.5mm to 1/4" cable, a cell phone with a 2.5mm headset jack can be plugged into the Remote Mix 4 in place of a standard phone line. The other party is heard through the returns on the headphones. The Remote Mix 4 becomes the headset of the connected cell phone, and the user is no longer tethered to a landline connection.

The four microphone and headphone jacks are on the back panel as well. A switch on each microphone input provides individual phantom power. Inputs three and four are switchable between microphone and line inputs. Input 4 is equipped with a stereo 3.5mm input jack, where the stereo signal is summed to mono (perfect for the output of a laptop).

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The big news

Finally, the really exciting Bluetooth feature is explained! The Bluetooth feature works much like the wireless cell phone interface, but eliminates the need for extra cables when using a Bluetooth-enabled cell phone. To do this, first select the option on the Bluetooth device to set up a connection. Then hold

the Bluetooth button on the Remote Mix 4 for five seconds. The Remote Mix 4 will locate any available Bluetooth-enabled devices within range. In order to connect to a Bluetooth device the first time, the device must be in pairing mode. Once the connection is established, the Remote Mix 4 will become a Bluetooth device on the cell phone. Bluetooth connectivity offers 20kHz bandwidth. With the Bluetooth technology available, less and less wiring and cabling are needed when using the Remote Mix 4 in the field.

The Remote Mix 4 is compact, simple and hands free – three terms that describe today's technology trends. JK Audio has kept up with the times and when the radio station staff needs to set up in the field quickly and easily, the Remote Mix 4 is keeping up in our wireless and hands-free world. Hopefully, Ma Bell doesn't mind sparing some airtime for all of these new wireless innovations!

Wygal is the programmer, engineer and Web designer for WRVL in Lynchburg, VA.

Editor's note: Field Reports are an exclusive *Radio* magazine feature for radio broadcasters. Each report is prepared by well-qualified staff at a radio station, production facility or consulting company.

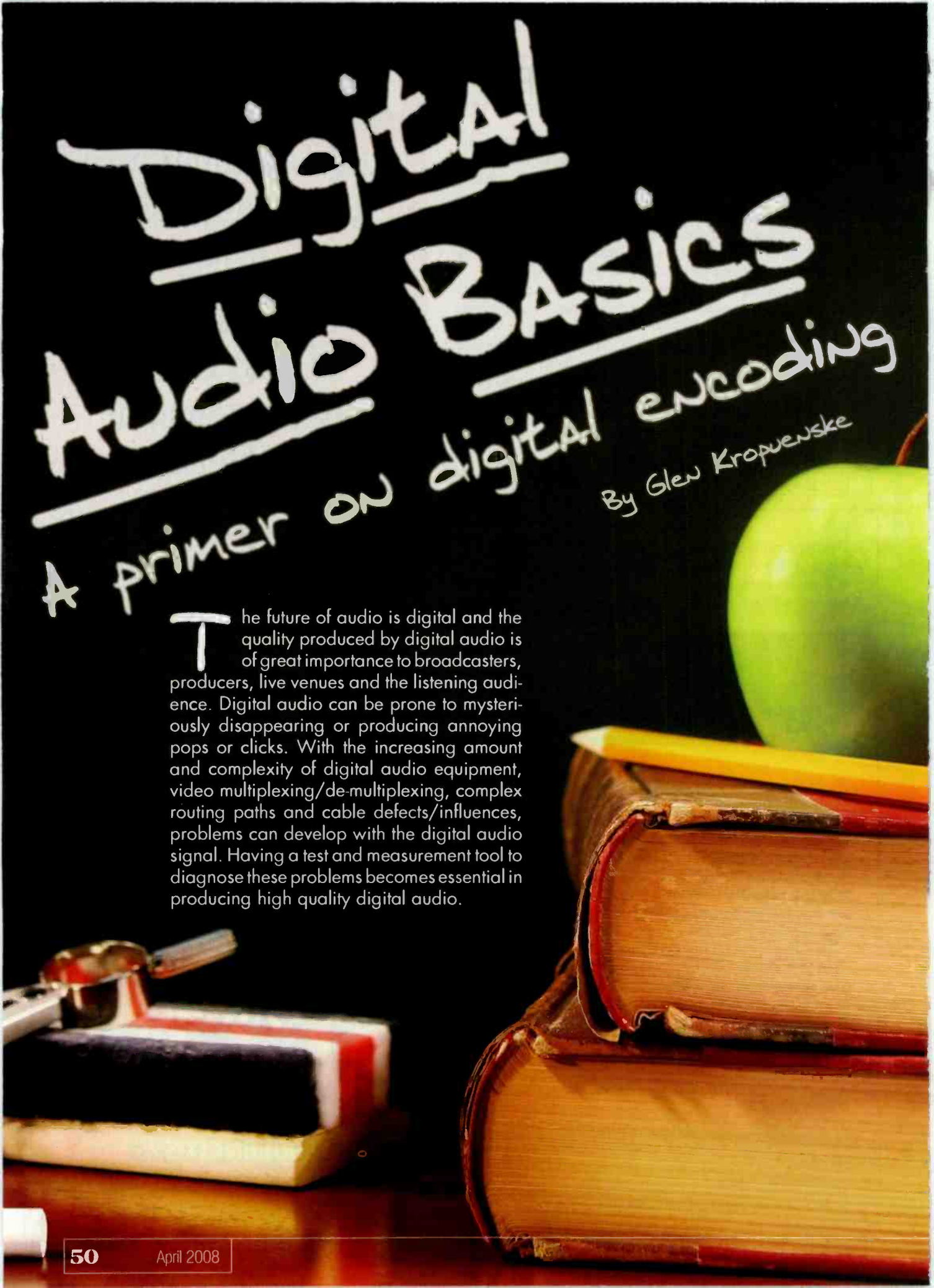
These reports are performed by the industry, for the industry. Manufacturer support is limited to providing loan equipment and to aiding the author if requested.

It is the responsibility of *Radio* magazine to publish the results of any device tested, positive or negative. No report should be considered an endorsement or disapproval by *Radio* magazine.

Digital Audio Basics

A primer on digital encoding

By Glen Kropuenske



The future of audio is digital and the quality produced by digital audio is of great importance to broadcasters, producers, live venues and the listening audience. Digital audio can be prone to mysteriously disappearing or producing annoying pops or clicks. With the increasing amount and complexity of digital audio equipment, video multiplexing/de-multiplexing, complex routing paths and cable defects/influences, problems can develop with the digital audio signal. Having a test and measurement tool to diagnose these problems becomes essential in producing high quality digital audio.

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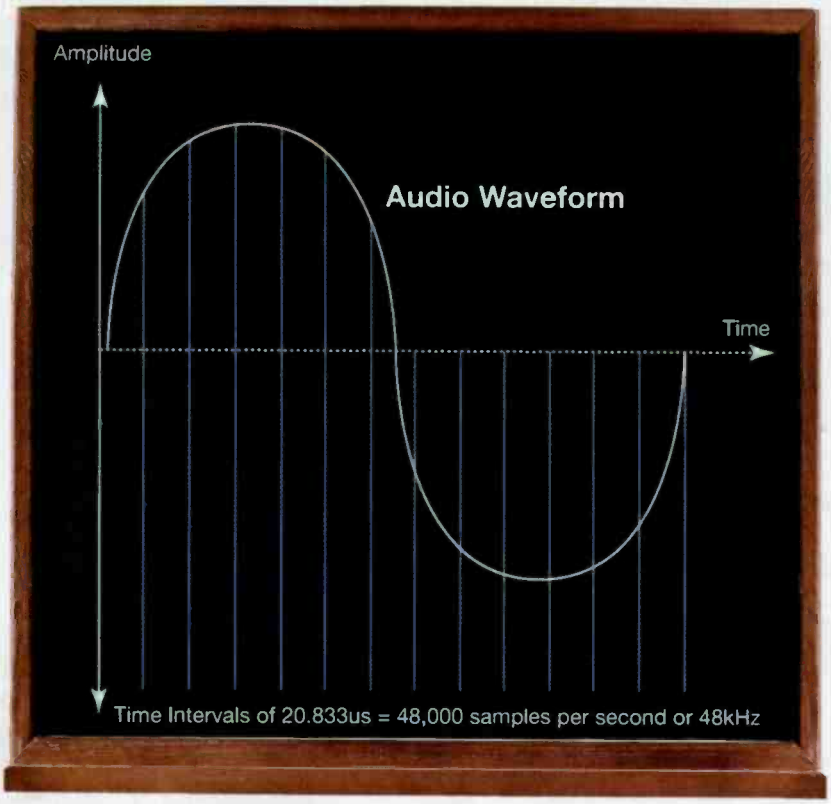


Fig. 1. Sampling an analog waveform at a specific rate.

Digital sampling to binary bits

The analog audio signal is converted to digital using a sampling process involving a digital-to-analog converter (D/A). To convert an analog audio signal from continuous changes vs. time to discrete digital values vs. time, the level of the signal is measured at certain intervals in time. This process is known as digital sampling. Sampling is like taking snapshots of the signal level as it increases and decreases over time. The sampled voltage levels are converted to digital binary values. The captured sequential digital binary value samples represent the audio levels as the audio varies in time.

Sampling happens at equally separated intervals. The number of intervals in which the values of the signal level are captured (snapshots) in one second is the sampling rate (see Fig. 1). The sampling rate frequency is the reciprocal of a sampling interval (time). The measurement unit of the sampling rate is the hertz (Hz) or number of samples per second.

The conversion process samples voltage levels of the changing audio at a much faster rate than the audio changes frequency. The Nyquist Theorem states the sampling rate must

be equal or greater than two times the highest audio frequency component. Therefore, to recreate audio at 20kHz, the minimum sampling rate must be higher than 40kHz.

Quantization

Fixing a digital value to the audio level at each sample interval is called quantization. The amplitude range of the audio waveform is divided into level steps. For example, there are 16 discrete binary values to specify the amplitude or level using a 4-bit binary system. The 16 values are divided in half (almost), with seven binary values to indicate positive voltage levels and eight values to indicate negative voltage levels.

The 4-bit quantization provides an example of how audio levels can be converted to digital 4-bit words representing the audio signal. However, the binary range is not symmetrical for positive and negative values and it has insufficient values for adequate low audio level reproduction. It should be noted that with digital quantization, the audio level must not exceed the quantization range or largest digital code word (1111), or digital clipping occurs. Fig. 2 shows the quantization levels on a sampled waveform.

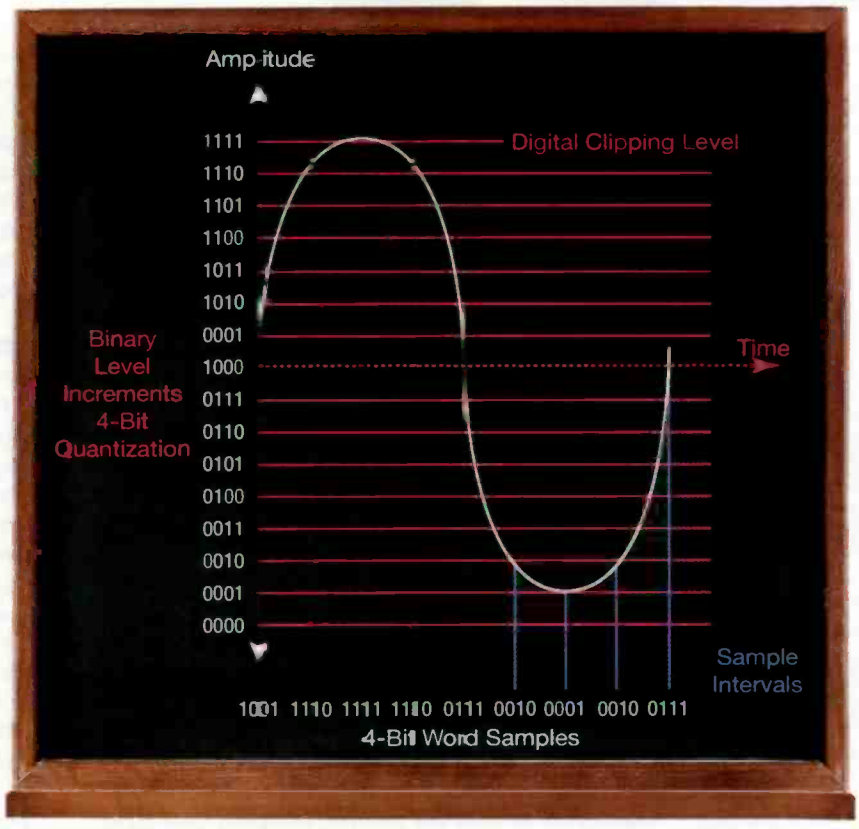


Fig. 2. Assigning amplitude values to a sampled waveform.

A quantized binary value may be encoded in a form adapted to the sampled signal and system requirements to provide overall system improvements. The most-used and adopted coding system is the Pulse Code Modulation (PCM) encoding. PCM linearly quantifies all quantizing intervals by means of a fixed scale over the signal amplitude range. PCM makes use of a two's complement system to distinguish positive and negative binary coded values (see Fig. 3). The analog-to-digital (A/D) conversion resolution accuracy is determined by the number of quantizing levels possible. The bit word length determines the number of quantizing level steps or amplitudes (resolution) that can be achieved. For example, an 18-bit digital code word provides 262,144 increments for coding analog signal amplitudes.

The number of bits used to form the PCM digital words (bytes) used to represent each of the sampled audio levels can vary, but typically range from 8 to 24 bits. The more bits used to represent the amplitude, the greater the dynamic range that can be represented. Each bit provides approximately 6dB of range. An 8-bit digital audio signal has a 48dB dynamic range (quiet to loud audio range) while 16-bit digital audio provides 96dB of dynamic range. A 24-bit digital audio word length provides 144dB of dynamic range.

PCM digital audio is often sampled at 44.1kHz or 48kHz, although other rates are possible. Sampling rates up to 192kHz can easily be found in some equipment.

In a digital audio system, the maximum audio level corresponds to 0dBFS (dB full scale) which is assigned the largest digital code word. Manufacturers have adopted the familiar 0VU level equal to +8dBm as a standard operating level (SOL). This level corresponds to -20dBFS, in which the digital values are well below the largest digital code word value. This provides 20dB of range for audio peaks to go above +8dBm before digital clipping occurs.

Putting the bits together

To retain the audio as a digital signal, each digital sample word or quantized value is sequentially output and moved along the transmission line or cable. This produces a stream of digital bits, word by word or sample by sample. It is important to realize that for each sample interval an 8-bit to 24-bit digital word length is created. These are serially assembled and output from a A/D converter. In reality, the number of bits per second (b/s) is equal to the sample frequency, multiplied by the digital word length.

The amount of data created by digital audio is quite large. Sixteen bits per sample at 44.1kHz creates 705,600 bits per second. To convert bits to bytes, divide by eight (8 bits = 1 byte). A 24-bit digital audio sample with a 96kHz sample rate produces a bit-rate of 2,304,000 bits per second. If two-channel audio is produced and assembled together onto a common transmission line, the data rate doubles.

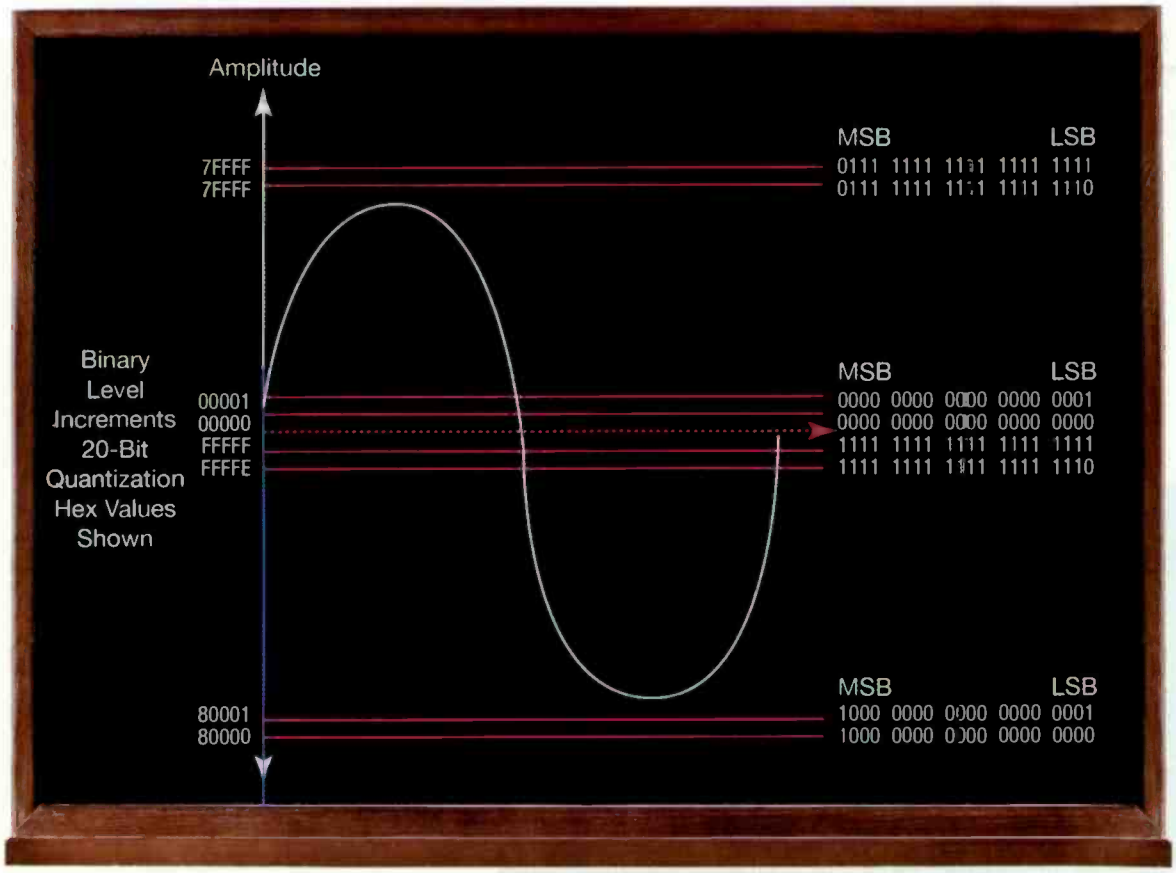


Fig. 3. The two's complement numbering system is used to differentiate positive and negative binary encoded values.

Without modification, a PCM digital audio stream can be difficult to receive reliably. If all the bits are set to ones or zeros for a period of time, the signal is essentially a dc voltage. A dc signal cannot be passed reliably by some circuits, and a prolonged dc level causes a shift or offset on the digital transmission line. A lack of digital transitions on the line would also make locking a receive clock or recognizing sync transitions difficult for the receiver.

To resolve this potential problem, the PCM digital data is encoded using another scheme called bi-phase mark coding (BPM). Bi-phase coding ensures that a dc shift doesn't occur on the line, by maintaining a dc balance. A dc balance means the on-time (highs) must equal the off-time (lows) resulting in a mean or average of zero volts. Bi-phase coding ensures balance by producing continuous and balanced transitions on the data line.

With bi-phase coding, each bit as a time slot begins with a transition and ends with a transition. If the data bit is a 1, a transition occurs in the middle of the time slot, in addition to the transitions at the beginning and end of the time slot. A data 0 has only the transitions at the beginning and end of the time slot and does not have a transition in the middle. This insures a transition and voltage balance no matter if there is a logic string of zeros or ones. Fig. 4 illustrates this. With regular transitions, the signal is a balanced ac signal in which a receiver can easily recover the clock rate.

Note that with bi-phase coding, the clock frequency is two times the audio data bit rate. Every audio bit is represented as two logical states when bi-phase coded. Each audio bit is divided into two time intervals or cells per data bit.

Blocking it together

The audio data words are assembled and transmitted serially. Some form of organization is needed so the receiver can reassemble and identify the assorted bits of information in the data stream. Organization involves assembling the data into blocks. Each block consists of 192 frames of audio. Each of the 192 frames can be divided into two sub-frames for two-channel audio. Each frame is produced at the digital audio sampling rate. In a 48kHz audio sampling rate, each frame is 20.833µs with each frame lasting 4ms.

Each frame can carry two audio channels. In a two-channel mode, the samples from both channels are transmitted in consecutive sub-frames. Channel 1 is in sub-frame A and channel 2 is in sub-frame B. In stereo mode, the interface is used to transmit stereo audio with both channels simultaneously sampled. The left audio is in the A channel sub-frame and the right audio is contained in the B sub-frame. Fig. 5 (page 59) shows the AES-3 data structure.

In addition to the digital audio word data bits, each sub-frame contains additional data. Each sub-frame consists of 32 bits, which includes 20 or 24 bits of audio word data bits and 8 bits of additional data. Each sub-frame includes bits for preamble or sync data, auxiliary data, audio data word bits, validity (V), user (U), Channel status (C) and Parity (P) data bits. Fig. 6 (page 60) shows the sub-frame structure.

Considering that each sub-frame consists of

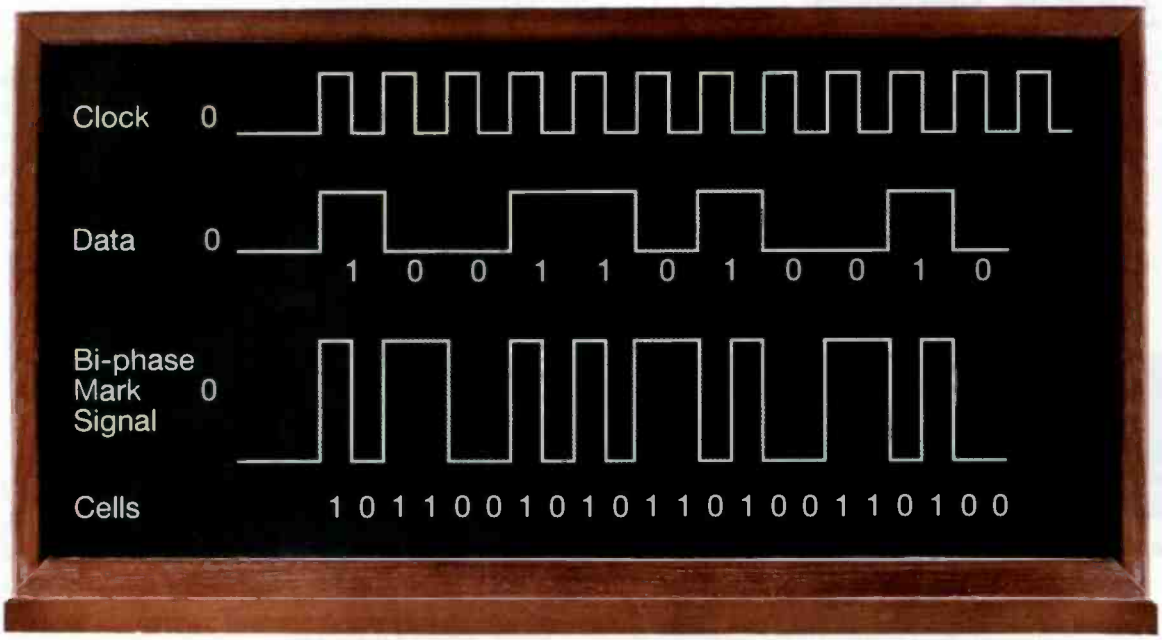


Fig. 4. Bi-phase coding ensures balance by producing continuous and balanced transitions on the data line.

32 x 2-bit, occurring in 20.833µs (FS = 48kHz), the bit-rate increases to 1,536,024 x 2 = 3,072,048b/s.

For each sample, two 32-bit words are transmitted, which results in a bit-rate of 2.8224Mb/s at 44.1kHz sampling rate or 3.072 Mb/s at 48kHz sampling rate.

Preamble or sync bits

The first four bits of each sub-frame consists of four preamble bits. The preamble bits may be called sync words, as they identify the start of a new audio block and each sub-frame. A Z sync bit arrangement marks the

start of the first frame in the 192-frame block. The sync word Y indicates the start of every B sub-frame. The sync word X indicates the start of all remaining frames. The bit patterns are shown in Fig. 7 (page 60).

The preamble has a distinctive data pattern that actually is not in compliance with the bi-phase coding rules. The first bi-phase coding violation occurs during the initial portion of the preamble marking the start of a frame. The initial portion lacks a normal bi-phase transition. The remaining preamble transitions identify the word type as Z, Y or X. This purposeful violation of the bi-phase coding rule allows a digital audio receiver to identify that start of the audio blocks and sub-frames. Table 1 shows the details of the preamble.

Preamble	Cell Order (last cell 0)	Cell Order (last cell 1)
Z	11101000	00010111
X	11100010	00011101
Y	11100100	00011011

Table 1. Bi-phase clocked cell structure.

It should be noted that the bi-phase rule breaking is by design and its effects do not cause any problems. Also by design, the bi-phase coding may cause each bit of the preamble or each of the 32-bit in the sub-frame to be opposite in phase.

The four bits following the preamble may be used as part of the main digital audio word or can be used for an additional audio signal known as auxiliary audio data. The use of auxiliary audio is rare. However, one application is for voice control communication. If the auxiliary bits are used for a special application, the following audio data word length is limited to 20 bits. The auxiliary data bits, if not used for a special application use, may be used to add bits to the audio data word length, extending the word length to 24 bits. Note the location of the auxiliary data bits in Fig. 6.

Audio sample word bits

The audio data word contains the digital audio bits defining the quantization values. The sequencing of values through the 192 sub-frames in each block is the digital audio itself. After one frame ends, another begins in the serial steam of digital audio bits.

The data word length can be 8 to 24 bits. If the auxiliary bits are defined for use for the digital audio data word, a 24-bit word length for maximum resolution and dynamic range is available. If the auxiliary bits are specified for special use, the maximum word length is 20 bits.

The audio data word is put into the sub-frame with the LSB (least significant bit) at the left while the MSB (most significant bit) is at the right in the pictured data stream.

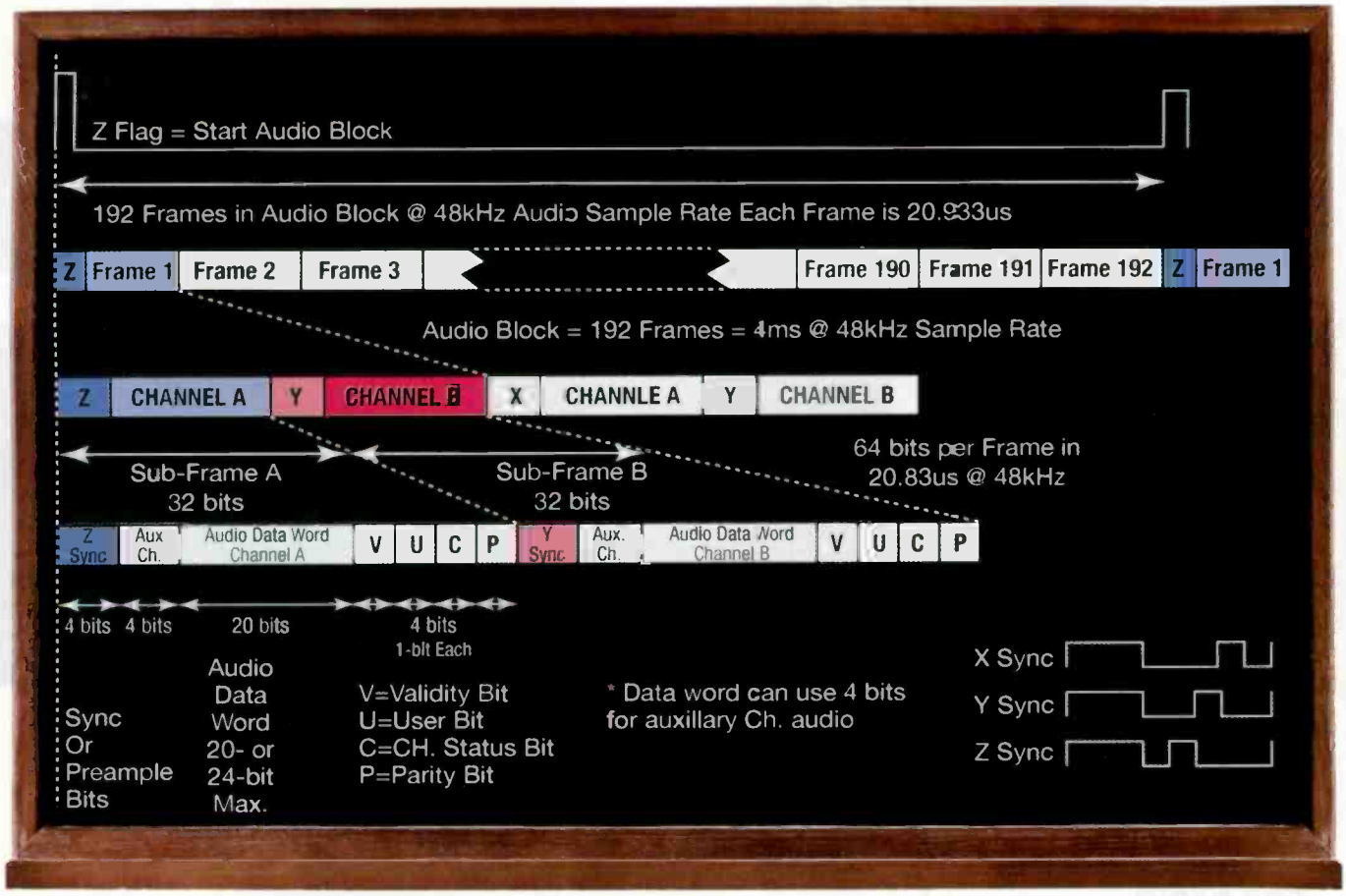


Fig. 5. The AES-3 data structure.

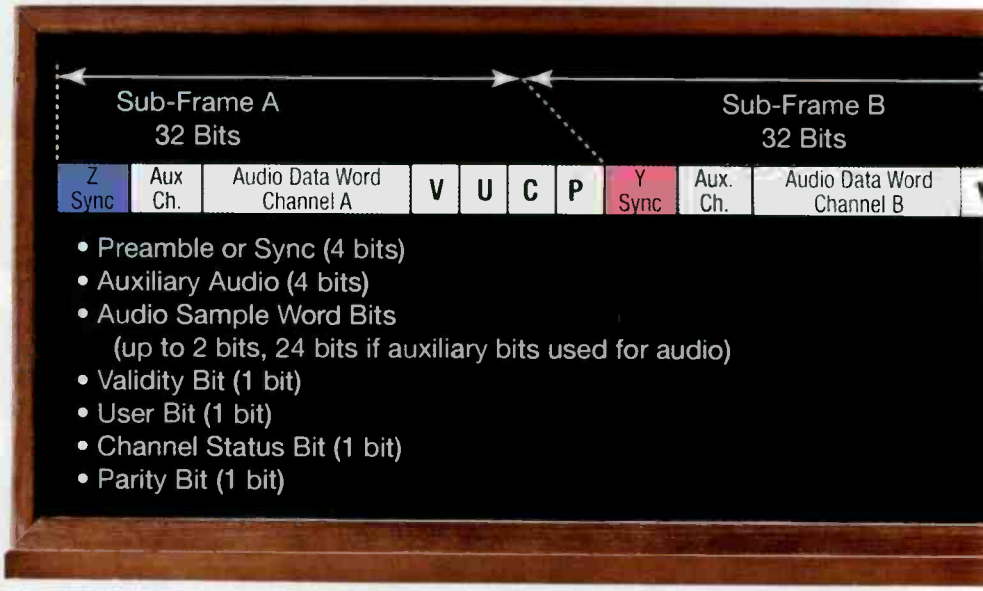


Fig. 6. The structure of the data sub-frames

While this may seem backward as illustrated on paper, as we are programmed to read from left to right, the data is clocked out of the D/A converter in the order of LSB to MSB, which is the order of the digital bit stream vs. time.

The validity bit (V) is set to zero if the audio sample word data is correct and suitable for D/A conversion. The validity bit was intended to signal the receiver to qualify it as being suitable for conversion. Several audio standards specify this bit to be set (to 1) when carrying data-compressed digital audio that cannot be converted to audio with a linear PCM receiver. This would cause the receiver to mute, preventing a burst of high-level noise output before the channel status data can be read and interpreted to indicate the digital signal is not PCM audio.

There has been some variation or confusion to the use of the validity bit. Certainly when the audio word data is modified with compression schemes, the validity bit must be set. However, some applications have set the validity bit if an error was found and concealed. As a result, some manufacturers have chosen to have the receiver not generate or verify the sample word validity.

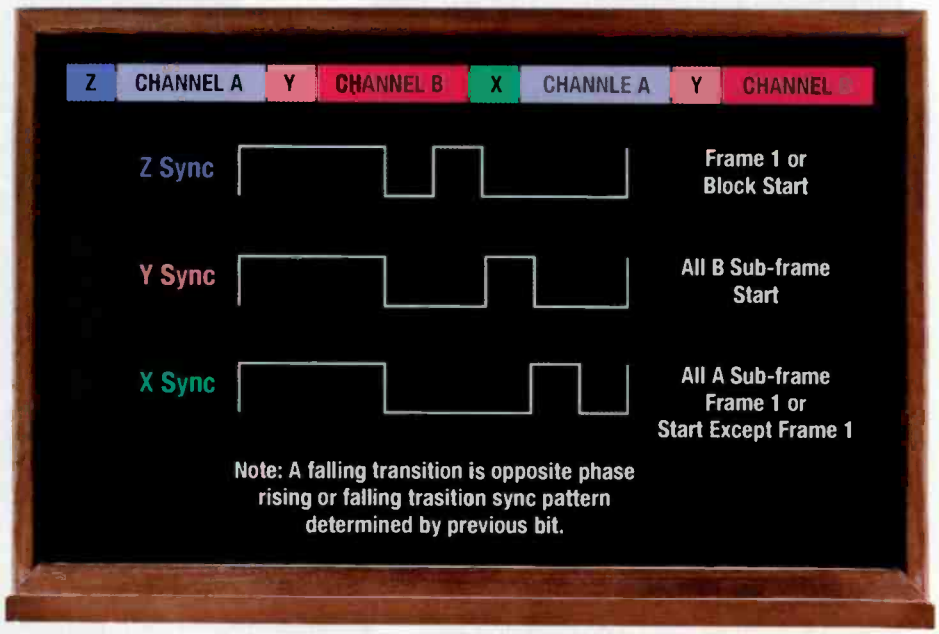


Fig. 7. Detail of the sub-frame preamble

The user bit (U) in each sub-frame occurs after the validity bit (V) and prior to the channel status bit (C). The user bits accumulated from each of the 192 sub-frames for that channel provide specific user information. The information is typically application-specific information regarding the program related information or instructions for preservation of the data.

The channel status bit (C) occurs after the user bit (U) and prior to the parity bit (P) in each subframe. The channel status bits accumulated from each of the 192 frames provides information regarding the audio in each channel. Since the information is usually identical in each channel, a receiver may elect to read only one of the channel's status bits.

Interpretation of the 192 channel status bits can best be understood by grouping the recovered bits from each sub-frame sequentially into 8-bit words (bytes), accumulating into 24 rows or bytes. In this manner, information regarding the channel's audio is indicated to the receiver.

The parity bit (P) is used to keep the data bits in the sub-frame (bits 4 to 31) at even parity. Even parity ensures that the total number of ones in the sub-frame data is always an even number. The parity bit permits the detection of an odd number of bits or bit transitions (bi-phase coding ones) indicating a bit error.

Even parity means that there is always an even number of mid-bit transitions (ones) in the data area of the sub-frame. Consequently, under normal bi-phase coding, the polarity of the first bit in every preamble or sync word is the same polarity transition. Also the second half transition of the parity bit is also the same as the previous sub-frame. This alone can be used to check for an error in the sub-frame data, ignoring the parity bit. In many cases this is more accurate, as two missed transitions would still result in even parity, and the error would be overlooked if only the parity bit were being used as an error indicator.

The parity bit is used to keep the data bits in the sub-frame at even parity.

Kropuenske is an application engineer at Sencore.

Viero Web services

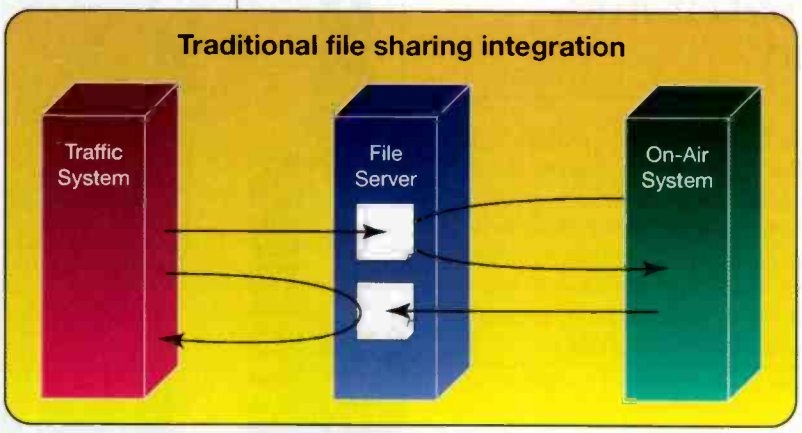
By Matthew Ferry

Your phone rings at 8:09 a.m. You keep typing to finish entering cart 794 before answering the call. Another ring and you are still filling in the fields: Advertiser Name, Spot Title, Spot Length. You save the record and grab the phone. It's traffic and they have a big late-add that has to run today. You make the trek to traffic's network of cubes and work it out, but then they want to go over a few make goods from the day before. You sigh as you set down your cup of coffee...

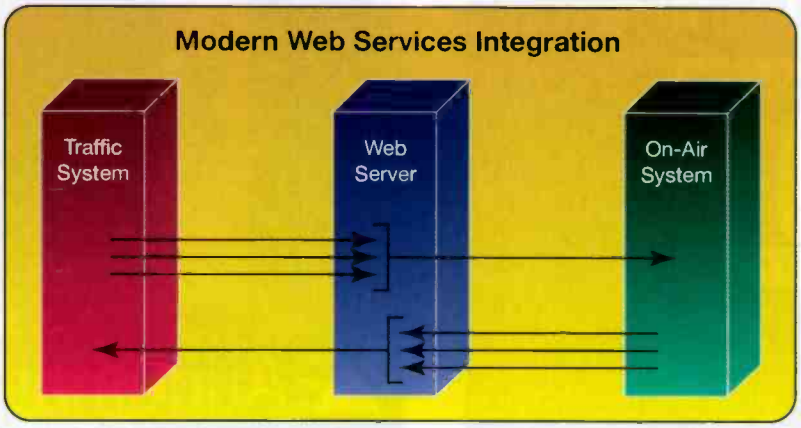
If this scenario sounds familiar then I'd like to share with you a trend that will make your work easier. Several years ago, a new technology called Web services began to make headway in software architecture. The technology leverages the same mechanisms used by Web browsers and Web servers to allow systems to more easily communicate and share data. Web services have proven extremely useful and are now considered to be mainstream software architecture.

They have also found their way into radio stations. The history of traffic and on-air software packages reflect how traffic and on-air departments are typically organized. These software packages have grown as separate silos and are usually provided by different vendors. This isn't bad – it allows vendors to specialize in areas in which they are good.

But these systems need to talk to each other. In the past, daily exchanges of flat files for logs and as played information were used. Modern software packages, such as Viero RMS (traffic and billing) and RCS Nexgen (on-air playback), now provide support for information exchange via Web services. RMS and Nexgen both have large installed bases and currently there are more than 1,000 radio stations where information is exchanged via Web services rather than traditional flat files. The use of this technology in radio is now well established.



In this method each system publishes a file to a file server. At a scheduled time, or on demand, each system looks for the other system's file and imports the information.



In this method, each system maintains a set of Web service interfaces on a Web server for the other system to call. As needed, each system sends messages to the other system's Web service interfaces.

The power of Web services

Web services are a powerful technology because of their openness and flexibility. Web service messaging is firewall friendly, easing communication between even geographically distributed systems. The open standards used in Web services help bridge communication between vendors using different development technologies. Data encoding and hierarchical structures allow any kind of data to be represented, including very complex representations of complete business records. And Web services offer a more dynamic method of interoperation than traditional flat files.

This approach enables a much richer and more dynamic interopration capability allowing each system to more actively participate in and support the other's function. This means smarter and more efficient software that allows users to get their work done faster. I began by highlighting some typical frustrations experienced by radio station staff: duplicate data

entry, late-adds and make goods. The good news is that these issues are being tackled. Below are the capabilities in Viero RMS that ease these issues. Currently, RCS Nexgen has full support for all of these features.

Features

Duplicate data entry. Viero RMS communicates in real-time with the on-air system to provide needed data, such as advertiser name, spot title, etc, eliminating double data entry. If anything is updated in the on-air system, such as a change in spot title, the correlating update is also made in the traffic system – automatically and in real time without human intervention. This is a time saver for both departments.

Late-adds. Support is built into the platform for electronically getting a spot over to the on-air system and into the schedule up to 15 minutes prior to air time. This capability is currently used by another product, Viero Transact, which uses Web services to manage national buys across a broadcast enterprise. The user interface components to leverage this capability within the traffic system have not been developed yet, but are targets for the near future.

Make goods. What if make goods could be handled automatically? Make goods are a manual chore for traffic staff. In the world of Web services the as played information is sent to the traffic system within seconds after a spot airs, or doesn't air. This feature, known as Near Real Time Verification is currently in production. It allows traffic to take immediate action, and enables a broader time frame to work reconciliation. In the future this feature will trigger the traffic system's scheduling engine to automatically reschedule the spot. This will be tied in with the late add capability to get the spot back on the air the same day.

The bridge between the systems is a Web server capable of running Web services built by the different vendors. In most cases that will mean a Windows-based Web server running Microsoft's Internet Information Server (IIS). IIS actually hosts the Web services and passes the messages back and forth. This does not need to be a very expensive machine because the real processing still happens in each system, and the Web server is simply moving messages back and forth between the systems. In the past there would typically be a file server in-between that both systems would write files to and read files from. The new model uses a Web server instead.

Over time, Web services, related technologies and the platforms they enable will fundamentally change software used in radio for the better. We are in the very early stages of seeing what these technologies will do for radio but as time passes more opportunities will be seen and more vendors will begin to participate as market pressures increase demand for efficiency. Radio will continue to seek new ways of enhancing revenue and as new channels for advertising continue to expand, such as additional streaming and HD Radio offerings, workloads for station staff will likewise increase. The software vendors who recognize this and add value in the form of increased efficiencies will be valued by broadcasters. This is an exciting time for our industry and a great time to be developing improved solutions for broadcasters.

Ferry is senior VP of software development, LAN International.

by Erin Shipps, associate editor

Digital audio signal generator
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Digirator DR2: A reference-grade digital audio signal generator with transformer balanced AES3, S/PDIF and ADAT outputs, this unit produces a comprehensive set of audio test signals. The DR2 also supports surround sound test sequences for verification and adjustment of professional Dolby Digital, Dolby E and DTS installations. The internal low jitter clock generator can be synchronized to AES3, DARS, word clock and video signals. Measurement of channel transparency, channel propagation delay and sample frequency is supported.

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UPS
Falcon Electric

SSG series: The SSG Series industrial-grade UPS Plus is available in models from 1.5 to 3kVA. This regenerative on-line UPS is an advanced, robust and rugged system that provides a higher level of power protection. The SSG Series is capable of withstanding elevated temperatures of up to 131 degrees Fahrenheit. It also provides an improved level of battery monitoring and replacement notification, as well as user-friendly, hot-swappable battery packs. Slide-out battery packs are easily replaced through the UPS front panel, without having to remove the UPS from the equipment rack, eliminating the handling of individual batteries and the associated wiring hassles. The SSG Series also gives users the ability to program two independent output load segments. This feature is designed to extend the battery runtime for a specified pair of receptacles for a particular connected device - such as a controller - while cutting off the power to the remaining receptacles after a utility loss or low battery condition.

800-842-6940; www.falconups.com
Info@FalconUPS.com

IP audio codec
Mayah Communications

Ganymed 1102: This codec is based on the Centauri product family. The focus is to cover applications in the field of audio-via-IP. The Ganymed 1102 is an IP audio codec encoding and decoding high quality audio signals into various coding formats and streaming towards other Mayah Ganymed, Centauri or audio player tools, such as Windows Media, Real Player, Winamp and others.

+49 811 55 16 0; www.mayah.com; info@mayah.com

Sample rate converter
Sonifex

RB-SC2: This 1RU device produces AES/EBU, S/PDIF and Toslink optical level digital audio outputs. The sample rate of the outputs can be set by an internal clock or from various external synchronizer sources. The unit also supports higher sample frequency rates up to and including 176.4kHz and 192kHz. It has two independent sample rate converter circuits and two optional video synchronizing boards that set the output sample rate to 48kHz. A special X-Lock mode allows the unit to function as a full bi-directional sample rate converter with the output of sample rate converter one syncing the input of sample rate converter two and vice versa. A serial RS-232 port is included so that the RB-SC2 settings can be controlled remotely. The front-panel LED indicators alter automatically when using RS-232 commands.

207-773-2424; www.independentaudio.com; info@independentaudio.com

NEW PRODUCTS

Microphone holder
Audix

Cab Grabber: The Cab Grabber enables sound engineers to place a microphone on a guitar amplifier or cabinet without the use of a mic stand. Constructed of durable aluminum tubing, the Cab Grabber is lightweight but very capable of holding microphones, including large diaphragm condenser mics weighing up to 16oz. Designed to operate with most guitar combo amps and cabinets on the market, the Cab Grabber will accommodate cabinets ranging from 8 to 14" in depth. Held on to the cabinet by a tension fit design, it will work equally well on cabinets with a flat surface or a recessed edge. The tubular arm, which is threaded to hold any standard microphone clip, will rotate 180 degrees and can be locked into position with a thumb screw. This, plus the ability to place the Cab Grabber at any location on either the side of the cabinet or the top of the cabinet, will allow the user to place the microphone in the exact location they desire.

800-966-8261; www.audixusa.com
info@audixusa.com

Network audio decoder
Barix Technology

Extreamer 1000: The Extreamer 1000 combines Instreamer and Exstreamer technology into a single device, adding several new features including a host of inputs and outputs. Eight contact closures allow broadcasters to transport data information with the audio to trigger and control local announcements, such as station identification, an EAS signal or even local studio cut-ins to a national feed.

866-815-0866; www.barix.com
info@barix.com

Secure digital flash recorder
Samson Technologies

Zoom H2: This portable recorder has four mic capsules onboard for 360-degree recording. With four mic capsules in a W-X/Y configuration, the audio is decoded instantly, bringing these four signals together for stereo imaging. For maximum flexibility, users can record from the front of the H2 in a 90-degree pickup pattern or the rear of the H2 in a 120-degree pickup pattern at up to 96kHz/24-bit resolution as a WAV file, or as an MP3 file at bitrates up to 320kb/s. Additionally, a 360-degree pickup pattern can be recorded at up to 48kHz/24-bit resolution to convert recordings to 5.1 surround.

631-784-2200; www.samsontech.com
info@samsontech.com

2.4GHz signal generator
Kaltman Creations

Air Horn: Working independently or in conjunction with Air Sleuth Pro is the Air Horn. Air Horn is a stable 2.4GHz ISM band signal generator that can be selectively set to generate RF signals for any number of the 11 Wi-fi channels. It has three transmission modes: Single Channel, Traverse all Channels and Peak. Air Horn can be used to assist in antenna placement, RF shield effectiveness, wireless network stress testing, etc. Air Horn is also sold as a software-based application.

678-714-2000; www.kaltmancreationsllc.com
sales@kaltmancreationsllc.com

Portable mixer
Allen & Heath

Zed-14: This stereo mixer provides 13 independent sources to the mix, 10 independent outputs, two pre-fade and two post-fade aux sends, a USB send and return for PC or Mac recording, playback and effects, a unique dual stereo input capability, and advanced monitoring facilities. It has six mono channels with a responsive three-band, swept mid-channel EQ, and four stereo channels with a two-band EQ, in 100mm faders. In addition to the twin TRS jack inputs, flexible dual input options include stereo return on RCA, 2TRK on RCA, and USB interface. Larger versions of the console are available.

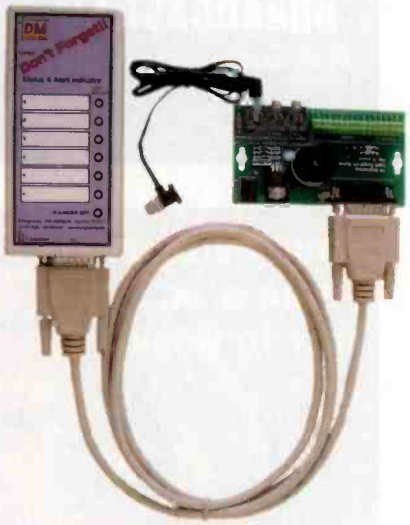
800-431-2609; www.allen-heath.com/US

250W FM transmitter
Tugicom RF Design

EXC250: The new case, heavy-duty fans and heatsink are specially designed for 24/7 operation even at severe conditions. The unit features a large LCD display, switching power supply for 110V/220V operation, soft start, digital temperature sensor, current sensor, backup protection for SWR, nine-pole low pass filter to meet FCC spectral rules, RF monitor and 45dB stereo separation.

+972-9-8615533
www.tugicom.com
info@tugicom.com

Status reminder
DM Engineering

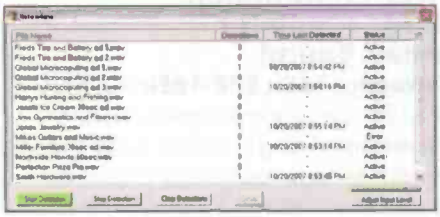


Don't Forget: This unit monitors tasks such as switching the local-remote switch or shutting the door on the aux transmitter, and reminds the user if the action is forgotten before he leaves the site. Don't Forget features six jumper selectable inputs for either N.O. or N.C. conditions, three opto-probe LED and visible light sensor inputs, LEDs for each monitored input. One opto-probe may be used to monitor a selected LED or light source for applications such as local-remote switches, over-modulation LEDs or any other light source desired. Don't Forget can also be used as a fault-alerting system and is easily interfaced with a remote control.

800-249-0487
www.dmengineering.com
info@dmengineering.com

NEW PRODUCTS

Audio monitoring software
Fifty Thousand Watt Software



Listen4me: This software program can listen for ads, music, jingles promos or any other recordings. The program can listen to audio from any source. Listen4me is extremely simple to use. Give it a sound file sample of any recording it should listen for. Listen4me listens to the sample recording to learn what it sounds like, and then listens to the audio stream to detect plays of the recording. Listen4me can keep a log of all occurrences of detected recordings, including recording name, date and time. The log file can be imported into a spreadsheet. Listen4me can also save a sound file copy of the detected recording, for later review and verification. In addition, it can keep a snippet of the preceding sound, so the user can determine what was playing right before the detected recording. Listen4me can listen for up to 20 recordings at once.

763-390-4046; www.50kws.com
sales@50kws.com

Music library
Loopmasters

Scratch Loops 1: The latest download collection features more than 150 fresh scratch loops from New York's DJ Shiftee, the youngest ever DMC Mix champion at the age of 17. Throughout this diverse range of scratches is a range of samples of musical loops, bass lines, vocals and some rhythmic combinations. The Scratch Loops 1 Pack includes 158 samples in Acidized WAV and Rex2 formats that have been organized so they will work easily with Stylus RMX, too.

+44 845 094 1460
www.loopmasters.com
info@loopmasters.com

Serial remote control
Broadcast Tools

SRC-16: Connecting through an RS-232, RS-422 or RS-485 serial port with baud rates up to 38400, the SRC-16 can notify a PC software program that any of 16 optically isolated inputs has been opened or closed and allows the software to control sixteen SPDT, 1A relays. The user is supplied with three pre-programmed format selections. The "PIP" GPI format is used on most of the company's audio switchers, and the AT-1616 format is used on one of the company's serial GPIO interfaces and the short ASCII format. Also, two units can be operated in a standalone mode (master/slave mode) to form a relay extension cord, with sixteen channels of control in each direction. The SRC-16 may be expanded to 64 inputs by 64 relays. Optional external Ethernet and USB capabilities may be added. The SRC-16 is supplied with LED indicators to display input and relay status. Three front-panel LEDs display power, serial transmit and receive data.

877-250-5575; www.broadcasttools.com
bti@broadcasttools.com

Power conditioner, voltage regulator
Furman Sound

IT-Reference 16Ei, SPR-16Ei: The IT-Reference 16Ei offers discrete symmetrical power to cancel hum-inducing noise from audio and video, power factor technology to provide extra current on-demand for power-starved amplifiers and four discrete dual-screen power banks to eliminate inter-component interference and noise. The SPR-16Ei power voltage regulator provides an ultra-stable 230Vac supply from excessively low or high ac voltage sources. It is also outfitted with Furman's proprietary Series Multi-Stage Protection Plus (SMP+) technology, which combines three filtering and protection circuits: Linear Filtering Technology (LIFT), Series Multi-Stage Protection (SMP), and Extreme Voltage Shutdown (EVS).

707-763-1010; www.furmansound.com
info@furmansound.com

Headphone amplifier
Behringer

Powerplay Pro-XL HA4700: This headphone amp offers four high-power amplifiers that can drive loads down to 8ohms and offers improved sound performance and new signal routing and mixing functions. The Powerplay Pro HA4700 houses four independent amplifier sections for up to four stereo mixes with individual balancing, mono/stereo and left/right mute switches, balance and volume controls, plus numerous master features, in a single rack space. The unit features three headphone output connectors for each of the four channels with one on the front and two on the rear panel, thus offering connections for up to 12 headphone sets. Each input/aux section can be separately set for individual sound adjustment. Paralleled main outputs allow cascading of several headphone amplifiers, while an accurate eight-digit LED output meters for each channel enables precise level control. An adjustable direct in on the front panel provides connection of any additional stereo signal source.

877-672-0816; www.behringer.com; support@behringer.de

LED steady burning obstruction light
Farlight

L-810: This omni-directional, red, steady burning obstruction light is designed for marking tall structures. Its light source is from three Luxeon Star LEDs, it consumes 4W, uses 120/240Vac or 12Vac/dc, weighs 4lbs (single), 7.5lbs (dual) and has an estimated lifetime of 70,000+ hours.

310-830-0181; www.farlight.com
moreinfo@farlight.com

UPGRADES and UPDATES

Lawo has released a version 2.8 upgrade for the company's Zirkon on-air radio console. The upgrade features a number of enhancements to the system control and configuration software. (www.lawo.de)...***OMT Technologies*** has unveiled the fifth generation release of Imediatouch, which includes 20 new features and user interface enhancements for touchscreen and drag 'n drop functionality. (www.omt.net)...***Prism Sound*** will show new software for its Dscope Series III audio analyzer at the NAB Show. Version 2.1 software upgrades functionality of the Dscope Series III, including generation and analysis using Windows sound devices, including multi-channel device support and 192kHz sample rate support on AES3 and S/PDIF digital audio interfaces. (www.prismsound.com) ■

Audio interface
Tapco
Link.Firewire 4x6: The bus-powered Link.FireWire 4x6 audio interface is a portable recording interface for musicians, composers and producers. It features a four-input/six-output format, low-latency 24-bit/96kHz operation, dual Mackie mic preamps and includes Mackie Tracktion Music Production Software.

425-487-4333; www.tapcoworld.com; sales@tapcoworld.com

Micro router
Harris
16x2: For use with the Netwave console, the Micro Router comes with a Netwave Dual Selector Upgrade Kit. This kit transforms the standard dual fader module into an integrated Micro Router controller. Two 10-character alphanumeric displays provide the means to display a pair of source selection legends. This Micro Router Package provides up to 16 remote sources to any Dual fader module, allowing individual selection for each fader. It even allows the selection of the local Netwave analog input, for a total of 17 possible selections on each fader.

800-622-0022; www.broadcast.harris.com; broadcast@harris.com

Linear PCM Recorder
Olympus

LS-10: The LS-10 Linear PCM Recorder features 24-bit/96kHz PCM recording; WAV, MP3 and WMA recording and playback; 2GB built-in memory; an SD removable media card slot; built-in speakers; a remote jack; a recording battery life of 10 hours; and an aluminum body.

888-553-4448; www.olympusamerica.com

USB drive security
Corsair

Flash Padlock: This allows users to have a hardware-secured lock to protect their data on a USB drive. This prevents any unauthorized access. Users can program in a PIN, much like they do for an ATM, to lock/unlock their data. An easy-to-use keypad in conjunction with lock/unlock indicator lights makes the Flash Padlock intuitive to use. Lastly, it is fully plug-and-play, and requires no software or drivers to work properly.

888-222-4346; www.corsair.com
info@corsairmemory.com

Cleaning wipes
Caig Laboratories

Deoxit: Deoxit Wipes are solvent and VOC free, RoHS compliant, come pre-saturated, individually packaged and are available in quantities of 25, 50 or 1,000. Deoxit Gold Wipes are available as well for use on gold or other precious metal plated connections. Deoxit is a unique fast-acting deoxidizing solution that cleans, enhances, lubricates and protects electrical conductivity on all metal connectors and contacts. Deoxit also acts as a lubricant and assists with difficult or tight connections. The entire Deoxit line of products is safe to use on plastic parts. Deoxit provides a long lasting protective coating that seals out oxygen and protects connections so they remain as good as new and significantly prolongs the life of the product being treated.

858-486-8388; www.caig.com
info@caig.com

CLASSIFIEDS

FOR SALE

HELP WANTED

CHIEF ENGINEER, RADIO
Southern California

Radio Broadcast Engineer for multiple AM/FM stations in So. California. Duties include scheduled/ emergency maintenance of all technical equipment in studio, transmitter facilities/towers and all IT equipment and networks; ensuring compliance with FCC rules and regulations; managing personnel; recommending equipment purchases/ budgets and maintaining the high quality and technical integrity of the stations. Requires 5-10 years broadcast engineer exp., technical knowledge of electronics including RF, studio maintenance and troubleshooting exp., in-depth knowledge of Scott Studios broadcast automation equipment, and strong IT skills with knowledge of networks. Must be able to test and maintain equipment proficiently, troubleshoot technical problems, able to work constructively under pressure and handle numerous time-sensitive issues on a daily basis. SBE Certification is a plus. Please send resumes to **corphr.mjk@gmail.com.**

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Contributor Pro-file

Meet the professionals who write for *Radio* magazine.
This month:
Facility Showcase, page 36.

Dave Lyons
Chief Engineer
WHKP-AM
Hendersonville, NC

Lyons has 20 years of radio broadcasting experience, having held positions in programming, research, on-air and engineering. His career has taken him to stations in Arizona, North Carolina, Alabama, Mississippi and Florida.

He previously worked as a project manager for Control Solutions.

He is also the owner of Wavtek Productions, an audio and video production company.

Radio
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Written by radio professionals
Written for radio professionals

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Armstrong Transmitter Corp	69	315-673-1269	www.armstrongtx.com
Arrakis Systems	35, 51	970-224-2248	www.arrakis-systems.com
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This index is a service to readers. Every effort is made to ensure accuracy, but *Radio* magazine cannot assume responsibility for errors or omissions.

by Erin Shipps, associate editor

Do you remember?

Around the office we've recently been reminiscing about oddly shaped mics like those used on TV by Bob Barker and Gene Rayburn. This ad from *Broadcast Engineering* in January 1969 shows that because of its shape, this Electro-Voice RE55 took some convincing: The ad says, "The more you listen, the better it looks." The RE55 was cased in steel with a satin nickel finish, had a wide-range, omnidirectional dynamic and featured an exclusive Acoustalloy diaphragm that could provide "undistorted output in sound fields so intense as to cause ear damage." But, the ad also states that the mic's long design is merely a convenience factor. Its diameter of ¾" fits nicely in the hand and is great for interviews. So maybe there isn't such huge significance to it's shape, but mics like this sure made the Barkers and Rayburns of our time look good.

Do you have interesting radio ads from the past? Tell us about it at radio@RadioMagOnline.com.

Sample and Hold

In-Vehicle Technologies

Billions of Dollars - Factory to Dealer Sales

According to the Consumer Electronics Association, sales of in-vehicle consumer electronics will grow at a rate of 13 percent in 2008 to more than $12.8 billion. The study also reports that the typical American spends nearly 17 hours a week in his vehicle. That's a lot of technology usage! Here is a look at the growth trends over the past five years.

Source: U.S. Consumer Electronics Sales and Forecast, Consumer Electronics Association, January 2008

Note: 2007 data is estimated, 2008 data is projected

That was then

The man in the middle of this photo is Lord Kelvin, an Irish mathematical physicist and engineer born in 1907. He is most famous for inventing the Kelvin scale of temperature measurement, but later in his career he also invented and engineered the electrical telegraph. This photo is one of many featured in Inecom Entertainment Company's DVD release, *Westinghouse*, due out April 8. The video focuses on the life and legacy of George Westinghouse and his partnership with Nikola Tesla. Westinghouse was a pioneer in broadcasting and founded KDKA, the first commercial broadcast radio station in the country.

Photo from Westinghouse (2008) courtesy of Inecom Entertainment Company

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"By far the best processor I've ever used."

"It achieved greater loudness with a smoother sound right out of the box."

"Your GUI is so well designed I didn't even need to read the manual to get started."

"Love the box!!! The sound of the station is vastly improved...it's loud, wide and clear."

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