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Radio World®
ENGINEERING EXTRA
August 24, 2005

DESIGNER INTERVIEW

The Evolution Of Transmitter Design

Geoff Mendenhall Talks About the Latest in Radio Transmission Systems

By Michael LeClair

With the advent of HD Radio, the industry is in the middle of a rapid evolution in the design of transmitters and transmission equipment. In this month's Designer Interview we speak with Geoff Mendenhall, vice president of research and development at Harris Broadcast Communications, about his perspective on these changes.

There are few engineers in a better position to talk about broadcast transmitter design and technology. Mendenhall is a registered Professional Engineer in the states of Illinois and Ohio, a member of the Association of Federal Communications Consulting Engineers and a senior member of IEEE. He has authored more than 30 technical papers as well as the chapter on FM transmitters for the seventh, eighth, ninth and soon-to-be-released 10th edition of the NAB Engineering Handbook.

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Perceptual Tests of Ibiquity's HD Radio Coder at Multiple Bit Rates

Study Suggests Use of Multiple Audio Channels on HD Radio Would Be Acceptable to Listeners

By Ellyn G. Sheffield, Ph.D.
Salisbury University, Salisbury, Md.

I. INTRODUCTION

With the introduction of HD Radio, important questions have arisen concerning optimal allocation of the 96 kbps data stream. In order to recommend to National Public Radio member stations the best allocation schemes available for primary and secondary audio channels, NPR conducted a consumer test over the summer of 2004. This study was intended to explore consumer acceptance of Ibiquity's HD Radio coder (HDC) at bit rates between 16 and 96 kbps.

Due to the variety of programming in today's marketplace and the flexibility of Ibiquity's system, an exhaustive study of all bit-rate combinations was not possible. Therefore, in order to quantify consumer satisfaction and to establish patterns of potential consumer behavior, bit rates from 16 to 96 kpbs were incrementally tested over a range of musical and speech genres typical to broadcast radio.

Specifically, this study was designed to explore whether: (a) general public listeners could detect quality differences in the HD coder at particular bit rates; (b) listeners rated these differences as meaningful and significant; and (c) listeners would change their listening behavior based on the differences in quality.

The study was conducted in two phases during the months of July and August. The first phase narrowed the field of testable bit rates in order to limit the number of test conditions on which the general public would be tested. This phase was conducted with a small sample of NPR audio engineers and personnel. The second phase was designed to obtain absolute category rating mean opinion scores (ACR-MOS) for a wide range of HDC bit rates and to test specific bit-rate comparisons that were found to be of interest from Phase 1 testing. This phase was conducted with 40 listeners from the general public. The details of both test phases are described in the remaining sections of this report.

II. TEST METHODOLOGY

Test Environment

Testing was conducted in a 1,700 square-foot sound studio at NPR, in Washington, D.C. The studio is approximately 53 x 32 feet, with a ceiling height of 15 feet. The ceiling has a spring-isolated acoustic lid at 18 feet, and the walls are built of concrete block. They sit on a four-inch thick "poured-in-place" floating concrete floor slab. The observed Noise Criteria for the studio was measured at PNC-19.

The studio was divided into six listening stations. Audio samples were presented to listeners binaurally over Sennheiser HD-600 open-backed headphones. Because the audio samples were delivered over open-back headphones, there was concern that leakage

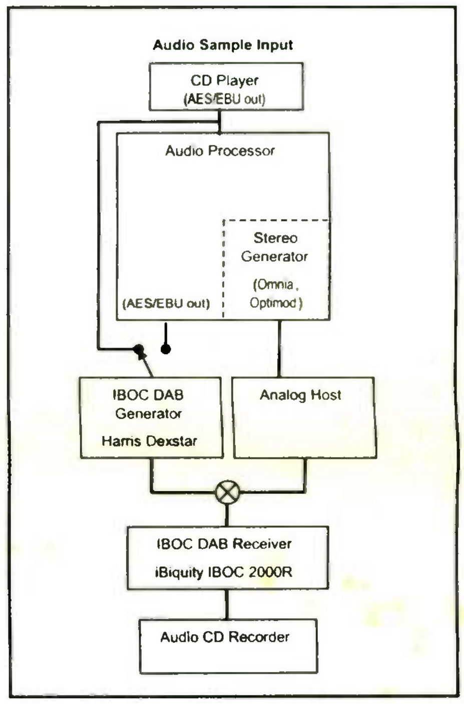

Fig. 1: Basic Equipment Configuration to Prepare HD Coder Audio Samples

would create audio interference between participants. Therefore, large foam blocks, measuring 4 feet square by 2 feet thick, separated listening stations from each other. The blocks, fabricated of 2 lb. per-cubic-foot open cell urethane foam, were stacked four feet high, providing acoustic and visual isolation between the listening stations.

Audio Samples

For both phases of this study, sound samples were taken from NRSC test material, NPR and Sun Sounds of Arizona program material, and music CDs. Speech,

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NPR Engineering Research Group Will Study Radio Technology

By Michael LeClair

Michael LeClair

In opening this column, I want to salute a positive development for broadcast engineering that was proposed at the Public Radio Engineering Conference in April. It has now become a reality.

Those of us who have been in radio engineering for a while are likely to remember the research done by CBS Labs, an arm of the broadcasting network that was dedicated to improving the technical quality of broadcasting.

Since the demise of CBS Labs in the mid-1980s there hasn't been a similar, full-time group with the resources to do empirical testing of broadcast transmission systems.

GETTING READY TO LAUNCH

Enter NPR Labs, a new arm of the engineering team at National Public Radio.

I asked Mike Starling, the vice president for engineering and operations at National Public Radio, to explain the impetus for such an initiative.

"With radio in the midst of a renaissance as it transitions to digital technology, we found this to be a good time for an independent group to help develop applications and test new technologies," Starling said. In the March 30 issue of *Radio World*, Starling wrote a full-page commentary about the need for such an entity.

Indeed, new possibilities for digital radio seem to multiply on a monthly basis. However, if these new capabilities are going to succeed it is essential for radio stations, broadcast manufacturers and receiver manufacturers to agree on which innovations to pursue and how. NPR Labs can help with this process.

NPR Labs will have an initial staff of four and will be located in Washington, D.C., at the headquarters of National Public Radio. It will be funded as part of the Engineering Division.

Although the official launch is yet to come, NPR Labs has already completed important research. A study was recently released on the use of booster transmitters at station KCSN and the technical implications of using HD Radio on a booster network.

Future projects will include a study of the impacts of HD Radio MP3 and MP4 extended hybrid-mode subcarriers on analog receivers.

Another possible project is the demonstration of emergency text services via digital radio for the hard of hearing.

SORTING OUT DETAILS

According to Starling, "The specific goal is to support the mission of National Public Radio and its member stations." That said, there is much interest in these areas of developing technology from commercial radio groups as well.

To support its research, NPR Labs has acquired a significant amount of test equipment. In addition to the typical spectrum analyzer and audio test sets, it has custom data-gathering systems, propagation analysis software, a multipath generator and multiple HD Radio exciters used to test analog reception bounded on either side by adjacent digital sidebands.

As an entity that is not in the business of selling products, NPR has the distance to make independent evaluations and engineering credibility developed from its long history of innovation. NPR Labs is going to help sort out the details of new radio technologies and with that comes the confidence that we can perhaps avoid some mistakes as we make our long, careful conversion to digital that's just in its beginning.

This will prove to be a valuable benefit to all radio broadcasters and the broadcast industry. Good job, NPR.

ON TO THIS ISSUE

As an example of the type of research to expect from NPR Labs, we have in this issue a report from Ellyn Sheffield, a former Lucent researcher during the development of digital radio. The paper considers the effects of low bit-rate audio coding, at rates that are being proposed for digital multicasting.

The question is whether these low bit-rate services will achieve listener acceptance. While it is hard to measure a subjective area such as audio quality, this report develops a methodology that is carefully designed to weed out subjective claims and determine whether listeners will be satisfied with multicast audio streams. The results are interesting and surprising.

We also have a paper from Tom Hartnett of Comrex discussing the company's new audio codec, designed to work with wireless Internet connections; an interview with Geoff Mendenhall of Harris Broadcast Corp.; and more from Guy Wire on the future of the AM band.

This issue of *Radio World Engineering Extra* marks our sixth and it completes our first full year in publication. The *Engineering Extra* has been a great success, and the response from the radio engineering community has been rewarding. Thank you for your support and please stay with us into the next year with more of our "deep tech" approach to radio engineering.

As always, write to me with your ideas and comments at *mlrwee@verizon.net*. ■

Correction

In the June 15 issue of *Radio World Engineering Extra* part of formula 3 was missing on page 18 in "How to Broadband AM Antenna Systems" by W.C. Alexander.

Here we reprint the formula in its entirety along with Fig. 3 with corrected component values for the Tee-Network.

$$B_T = \frac{R_{LOAD}}{R_{LOAD}^2 + X_A^2}$$

$$X_A = X_{LOAD} + X_1$$

$$Y_T = Y_A + Y_B$$

$$Y_A = \frac{X_A}{R_{LOAD}^2 + X_A^2}$$

$$Y_B = \frac{1}{X_2}$$

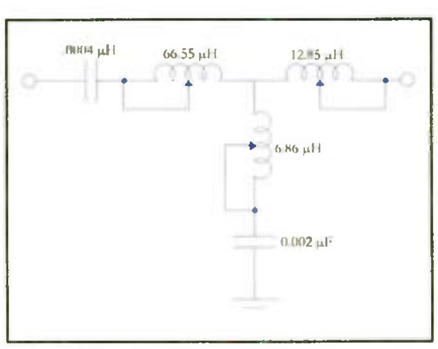

Fig. 3: Tee-Network With Input Leg Slope Correction

In This Issue

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voice-overs and music (rock, jazz and classical) were included.

Preparation of HD Coder Audio Samples

Audio samples were prepared on the FM test bed shown in Fig. 1. The system produced a hybrid digital and analog FM-band signal with stereo subchannels in compliance with the FCC Part 73 rules and applicable industry standards.

The test bed passed audio samples from an audio CD through a transmission/receiving chain. The resulting HD-encoded and decoded audio was recorded on audio CD, for later transfer to playback equipment used by the listeners. The stereo generator and FM Host analog exciter side-chain did not contribute to the audio sample transfer. It was included only to provide compliance with hybrid DAB transmission standards.

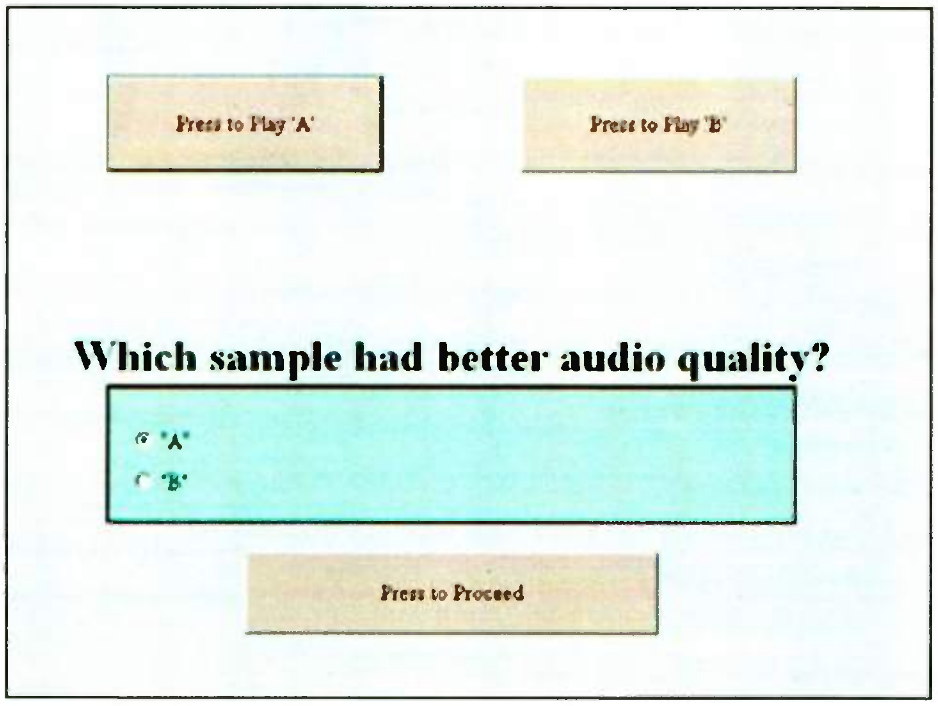

Fig. 2: Screen Capture of PC Response Display for A/B Discrimination Task

Audio transferred through the IBOC DAB side-chain remained digital at all times. For the main testing project, the digital audio from the CD player was fed directly to the IBOC DAB generator, completely bypassing the audio processors. All audio HD coder samples were unprocessed, thereby providing comparability to the CD source references. However, care was taken to match loudness levels between all the samples and ensure that peak levels did not reach 0 dBFS.

24 kbps	36 kbps	48 kbps	56 kbps	64 kbps	72 kbps
24 vs. 36	36 vs. 48	48 vs. 56	56 vs. 64	64 vs. 72	72 vs. 80
24 vs. 48	36 vs. 56	48 vs. 64	56 vs. 72	64 vs. 80	72 vs. 96
	36 vs. 64	48 vs. 72		64 vs. 96	

Table 1: Sample Pairs Used in Phase 1 Testing

Bit rates	Percentage of respondents claiming higher bit-rate sounded better	t-test, probability level
24 vs. 36	77%	t = 5.3693; p = .0001
24 vs. 48	76%	t = 4.9812; p = .0001
36 vs. 48	71%	t = 4.2696; p = .0001
36 vs. 56	67%	t = 3.3230; p = .0001
36 vs. 64	68%	t = 3.4320; p = .0001
48 vs. 56	50%	not significant
48 vs. 64	46%	not significant
48 vs. 72	54%	not significant
56 vs. 64	46%	not significant
56 vs. 72	54%	not significant
64 vs. 80	61%	t = 2.2103; p = .01
64 vs. 96	54%	not significant
72 vs. 80	59%	not significant
72 vs. 96	54%	not significant

Table 2: Results of A/B Discrimination Testing

Bit rate	Size of difference
24 vs. 36	4.17
24 vs. 48	5.06
36 vs. 48	3.11
36 vs. 56	3.48
36 vs. 64	3.53
48 vs. 56	3.09
48 vs. 64	2.51
48 vs. 72	3.64
56 vs. 64	2.70
56 vs. 72	3.02
64 vs. 80	2.61
64 vs. 96	2.49
72 vs. 80	2.20
72 vs. 96	2.82

Table 3: Size of Difference When Quality Was Identified Correctly

Bit-rate	Discontinue
24	43%
36	43%
48	15%
56	17%
64	17%
72	21%
80	11%
96	17%

Table 4: Percentage of Participants Who Would Discontinue Listening

Presentation Software

The playback of samples to listeners was controlled using a software package developed by Ibiquity Digital Corp., which has been utilized in prior testing submitted to the NRSC. Sound samples were stored on the hard-disk drives of PCs and presented to listeners individually at each station. The software collected and stored listener responses, requiring no experimenter control or interaction once the test session commenced.

Participants were free to take the test at their own pace, and were given instructions to play samples as many times as necessary to make good decisions.

III. NARROWING THE FIELD OF BIT-RATE COMPARISONS (PHASE 1)

Ten NPR employees participated in Phase 1. Listeners included four audio engineers and six additional staff members employed in various departments at NPR. By virtue of working at NPR, this listening population may be described as "well-educated" in terms of sound quality, but would

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not necessarily be characterized as "golden ears." However, all of the audio engineers who participated work extensively with sound, and thus, are likely to be more sensitive to very small changes in sound quality than the general public.

Design and Procedures

Listeners were presented with a total of 98 bit-rate pairs (each pair consisting of two samples, back-to-back), and were asked the following questions about each pair:

(a) Which sample had *better* audio quality, "A" or "B"?

(b) How big was the difference, on a scale of 1 to 10, with 10 being "extremely different," 5 being "different," and 1 being "I really couldn't tell a difference but you made me pick"?

(c) Would you turn either sample "A" or "B" off?

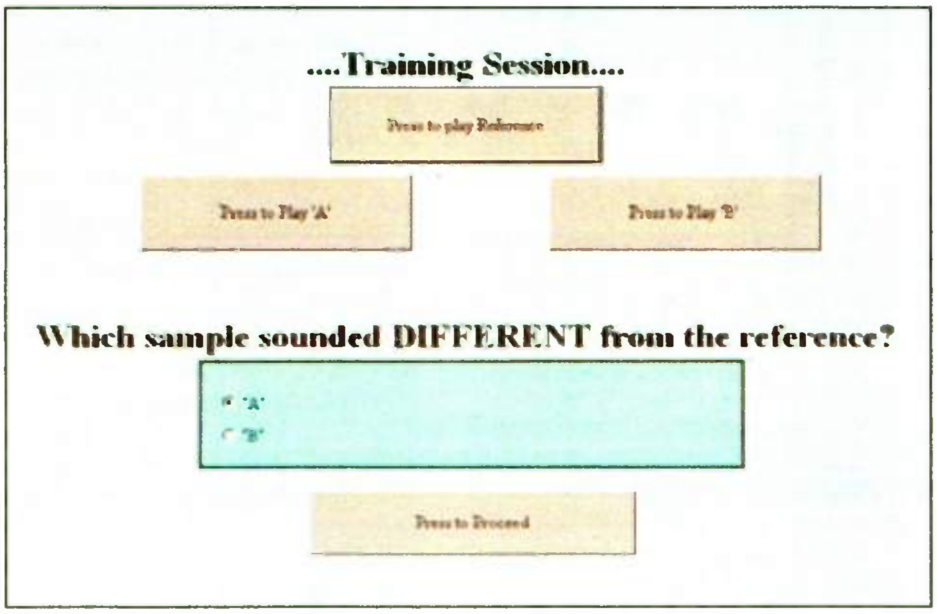

Fig. 3: PC Response Display for Screening Test

The test was divided into two sections, with listeners answering 49 trials before receiving a five-minute break. Listeners were encouraged to play the samples as many times as they needed to make these determinations. Allowing unlimited access to sample pairs afforded participants the greatest opportunity to discern small differences between the samples. Thus, we believe that their response data represents an extremely precise and stringent discrimination measure. Sample pairs were randomized, so that each participant heard the pairs in a different order. Pairs were counterbalanced. For half the pairs, the lower bit rate was sample "A," and for the other half, the lower bit rate was sample "B." Fig. 2 shows the PC response display used for the A/B discrimination task.

Table 1 shows the sample pairs used for this test. Notice that at each bit rate, sample pairs that were quite close (8 and 16 kbps difference) were included. At points of special interest, pairs that were further apart (36 vs. 64, 48 vs. 72 and 64 vs. 96) were included. At each bit rate, samples included: (a) male speech; (b) female speech; (c) classical music; (d) jazz; (e) rock; (f) male voice-over; (g) female voice-over.

	16	24	36	48	56	64	72	80	96	CD Source	Total
Speech (2 male; 2 female)	4	4	4	4	4	4	4	4	4	4	40
Classical	4	4	4	4	4	4	4	4	4	4	40
Jazz	4	4	4	4	4	4	4	4	4	4	40
Rock	4	4	4	4	4	4	4	4	4	4	40
VoiceOver	4	4	4	4	4	4	4	4	4	4	40
Total	**20**	**20**	**20**	**20**	**20**	**20**	**20**	**20**	**20**	**20**	**200**

Fig. 4: Samples Used in ACR Test

Age	Female	Male
18-29	6	6
30-39	5	4
40-49	5	4
50+	4	6

Table 5: Demographic Breakdown of Participants Included in Results

	24 vs. 36	24 vs. 48	36 vs. 64	48 vs. 64	48 vs. 96	64 vs. 96	Total
Speech (1 male; 1 female)	2	2	2	2	2	2	12
Classical	1	1	1	1	1	1	6
Rock	1	1	1	1	1	1	6
Jazz	1	1	1	1	1	1	6
Total	**5**	**5**	**5**	**5**	**5**	**5**	**30**

Table 6: Samples Used in A/B Test

Results for Phase 1 Testing

Table 2 shows total results for discrimination testing. Statistical tests were run to see if the percentage of respondents claiming that the higher bit rate sounded better than the lower bit rate was statistically different from chance, or 50 percent. The statistical test we chose was the "paired t-test." The t-test yields a value that, at higher levels, indicates greater accuracy. The "p" value associated with the t-test is a measure of how reliable the findings of the sample group are in relation to the population at large. A p-value of 0.05 means that 5 percent of the time the significant difference that appears may be an inaccurate reflection of what the real-world listening population would decide. A 0.05 p-value is used in research as the standard measure of reliability. A p-value of over 0.05 indicates that the finding is "not significant," or unreliable.

At lower bit rates, listeners were able to accurately report that the higher bit rate sounded better than the next adjacent bit rate (see 24 vs. 36; and 36 vs. 48). However, with one exception (64 to 80 kbps), at mid-range bit rates and above, listeners were unable to reliably tell the difference when the samples differed by 8 or 16 kbps.

Size of Difference

Participants were also asked how big the difference was between the audio samples in an audio pair on a 1-to-10 scale, 1 being no difference at all; 5 being a difference; 10 being extremely different and noticeable. Table 3 shows these results for participants who had correctly identified the higher bit-rate sample. Notice that participants claimed small differences at the higher bit rates, indicating that although they heard a difference between the two samples, the perceived quality difference was minimal.

Listener Behavior

Finally, listeners were asked whether they would continue to listen to sample "A," sample "B," "neither" or "both" at various bit rates. Table 4 shows the rate of discontinue listening for each bit rate. Notice that over 40 percent of participants reacted negatively to samples coded at 24 and 36 kbps, but this number dropped substantially to 15 percent at 48 kbps. This result indicated that somewhere around 48 kbps the great majority of listeners began to react favorably to HDC.

	16	24	36	48	56	64	72	80	96	CD Source Reference
Classical	2.8*	3.2*	4.0	4.0	4.0	4.1	4.1	4.0	4.1	4.1
Jazz	3.3*	3.7*	4.0	4.0	4.1	4.1	4.2	4.1	4.2	4.2
Rock	2.5*	3.1*	3.7*	3.9	3.9	4.0	4.0	4.1	4.1	4.2
Speech	2.0*	2.9*	3.4*	3.7*	3.8	3.7*	3.8	4.0	3.9	4.1
Voiceover	2.4*	3.0*	3.2*	3.3	3.5	3.5	3.5	3.4	3.5	3.4

Table 7: Mean Opinion Scores for Genres

Based on listening results of NPR personnel, we selected a subsample of bit-rate comparisons for inclusion in Phase 2 consumer testing. We included two low bit-rate comparisons that were reasonably large (i.e., 24 x 36; 24 x 48), to see if the general public corroborated NPR listener views. We also included two comparisons that were potentially important to the allocation of 96 kbps available in the Main Audio Program stream (MAP) (i.e., 48 vs. 64; 48 vs. 96). Finally, we included 64 vs. 96 to replicate test conditions in previous NRSC FM testing. Because Phase 1 indicated that NPR listeners could not reliably discern differences between 48 and 56 kbps, and 48 and 72 kbps, we did not include those comparisons in general consumer testing. However, we did include a 48 vs. 96 kpbs bit-rate pair to see whether consumers could hear a difference between the two more disparate bit rates.

IV. CONSUMER TESTING (PHASE 2)

Fifty-nine total listeners (29 males and 30 females) were initially screened, distributed between 18 and 65 years of age. Subjective data from 40 qualified listeners was collected, where qualification was based on performance on the initial screening test and a post-hoc screening test designed to eliminate outliers. Four males and five females were excluded from final results because they failed the screening test. Seven participants were excluded because they did not complete the test. Three more female participants were excluded in order to make even the number of responses from each gender. Table 5 shows the demographic breakdown of general public listeners. Listeners were recruited from several sources, including friends and family members of NPR staff, flyers posted in the downtown Washington area and outlying suburbs, and online postings.

Design and Procedures

General consumer testing was conducted between July 19 and Aug. 3 (of 2004). Participants were tested individually over Sennheiser HD-600 headphones for approximately 2-1/4 hours. The test session was divided as follows:

1. Experimenter welcomed participants and described the equipment and test procedures.
2. Participants were given a screening test, followed by a short break.
3. Participants were given an ACR-MOS test, followed by another short break.
4. Participants were given an A/B pairwise comparison test.
5. Participants were debriefed, paid and escorted out.

Notice that in this study, participants rated the same samples in two ways: (a) they completed an Absolute Category Rating-Mean Opinion Score test and (b) they completed an A/B comparison test on selected sample pairs.

Why use both methodologies? The ACR opinion scores derived from a single stimulus presentation test tend to be highly predictive of real-world consumer satisfaction. Listeners are rating samples one at a time, using their internal reference to guide their decisions. This is how most consumers judge audio on a daily basis. However, it has been argued that the ACR-MOS is not as sensitive to differences as other kinds of testing, such as directly comparing one audio sample to another in an A/B presentation. Therefore, in order to test stringently and thoroughly, both the ACR-MOS and A/B comparison test methodologies were included in this study.

	16	24	36	48	56	64	72	80	96	CD Source Reference
Female	1.8*	2.8*	3.3*	3.5*	3.6*	3.6*	3.7	3.9	3.8	4.1
Male	2.1*	3.0*	3.6*	4.0	4.0	3.9	4.0	4.0	4.1	4.1

Table 8: Mean Opinion Scores for Female and Male Speech

Screening Test

Screening was conducted to ensure that listeners were reliably able to distinguish between significantly different audio qualities. There were seven screening trials. For each trial, participants were asked to listen to three samples, two of which were the same and the third different (for example, two female speech source samples and the same female speech sample processed through an AM receiver; two rock source samples from a CD and the third sample coded at HDC 24 kbps).

The listener's task was to decide which of two "test" samples ("A" or "B") was different from the reference sample. In each trial, the first sample they heard was always the "reference" sample. They then listened to the "A" and "B" samples and judged which of the samples was different from the reference. Listeners were free to replay any or all of the three samples until they were ready to enter their response and proceed to the next trial.

In order to "pass" the screening test, participants had to answer *six* of *seven* screening triads correctly. Listeners were provided no feedback on the "correctness" of their responses during the screening test, nor were they informed of their specific performance after they were finished.

Playback of samples was under listeners' control, but the screening software required them to listen to all three samples, from beginning to end, before the response options became available. Fig. 3 shows the PC response display that was used for the screening task.

Single Stimulus ACR Test

In the ACR test, participants listened to 200 samples, one by one, and then rated

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CONTINUED FROM PAGE 7

each sample individually. The test was divided into several subsections, with participants answering 67 trials and receiving five-minute breaks, until all trials were finished. The ACR test yielded a Mean Opinion Score (MOS), a measure of overall audio quality. Listeners were required to judge the quality of an audio sample using a five-category rating scale (Excellent=5, Good=4, Fair=3, Poor=2 and Bad=1).

Listeners controlled playback of the audio samples, but were not allowed to register their answer until the entire sample was played. Listeners were given the opportunity to adjust the playback volume during one practice trial, and this level was maintained throughout the remainder of the experiment. Fig. 4 lists single stimulus samples used in the ACR test.

Bit rates	Percentage of respondents claiming higher bit-rate sounded better	t-test, probability level
24 vs. 36	77%	t = 9.2935; p = .0001
24 vs. 48	80%	t = 10.8652; p = .0001
36 vs. 64	64%	t = 4.1145; p = .0001
48 vs. 64	54%	not significant
48 vs. 96	56%	t = 1.8491; p = .03
64 vs. 96	57%	t = 1.9932; p = .02

Table 9: Results From A/B Discrimination Testing

Double Stimulus A/B Test

In the double stimulus test, participants were given 30 sample-pairs and asked the same three questions that Phase I listeners were asked:

(a) Which sample had *better* audio quality, "A" or "B"?

(b) How big was the difference, on a scale of 1 to 10, with 10 being "extremely different" and 1 being "I really couldn't tell a difference but you made me pick"?

(c) Would you discontinue listening to sample "A" or "B," neither or both?

Table 6 lists the sample pairs participants were asked to rate.

Bit rate 48 vs. 96	Percentage of respondents claiming higher bit-rate sounded better	t-test, probability level
Jazz (n = 40)	55%	not significant
Rock (n = 40)	60%	not significant
Speech (n = 80)	58%	not significant
Classical (n = 40)	53%	not significant
Bit rate 64 vs. 96		
Jazz (n = 40)	55%	not significant
Rock (n = 40)	55%	not significant
Speech (n = 80)	64%	t = 2.5423; p = .0001
Classical (n = 40)	48%	not significant

Table 10: Results From Discrimination Testing at 48 and 64 kbps by Genre

V. CONSUMER TEST RESULTS

A preliminary analysis of variance (ANOVA) was conducted to examine whether participants rated audio quality of samples differently because of their age or gender. This test was run to ensure that the data from all participants could be combined for further statistical analyses. If group differences were extremely large, then it would not be wise to treat all participants' data as homogeneous.

The ANOVA showed differences between age groups, indicating that older participants rated samples less critically than younger participants. This effect has been seen in past audio testing and is thought to be a function of slight hearing loss affecting older people, particularly at higher frequencies. The range of mean scores, however, was rather small between the youngest and oldest groups: 18-29-year-old participants' mean was 3.5; 50+-year-old participants' mean was 3.8.

Further, the ANOVA showed that females and males rated samples similarly. Thus, because differences were quite minimal between age groups, and nonexistent among females and males, participants' data was combined for all other analyses and total results are reported.

Absolute Quality Rating

Table 7 shows the ACR-MOS for bit rates from 16 kbps to 96 kbps, as well as the CD source reference samples. The results are listed by genres. A one-way analysis of variance was conducted for each genre to see if the scores at various bit rates were significantly different from each other. These analyses yielded significant differences, which are highlighted on the table by asterisks.

In classical and jazz, 16 and 24 kbps were rated significantly lower than all other bit rates and the reference. In Rock, 16, 24 and 36 kbps were rated significantly lower than all other bit rates. In Voice-over, 16, 24 and 36 kbps were rated significantly lower than all higher bit rates. In Speech, 16, 24, 36, 48 and 64 kbps were all rated statistically lower than the reference. However, while 16, 24 and 36 were rated significantly lower than 96 kpbs, 48 kpbs and 96 kpbs were rated equivalently.

In order to examine the speech genre more closely, it was divided into male and female speech. Table 8 shows slight differences between participants' scores for female and male speech. For female speech, 48, 56 and 64 kbps were rated significantly different from the reference (but not from 96, 80, 72), whereas with male speech 48 kbps was rated significantly the same as all of the higher bit rates.

HD CODER, PAGE 10

Case Study

Simple EAS Switching

Solution developed for numerous Logitek customers

THE ISSUES:

- Analog EAS system, digital transmission chain—how do you route the audio?
- Analog AUX audio source also needs to be switched
- Switching needs to happen automatically when EAS relay activates

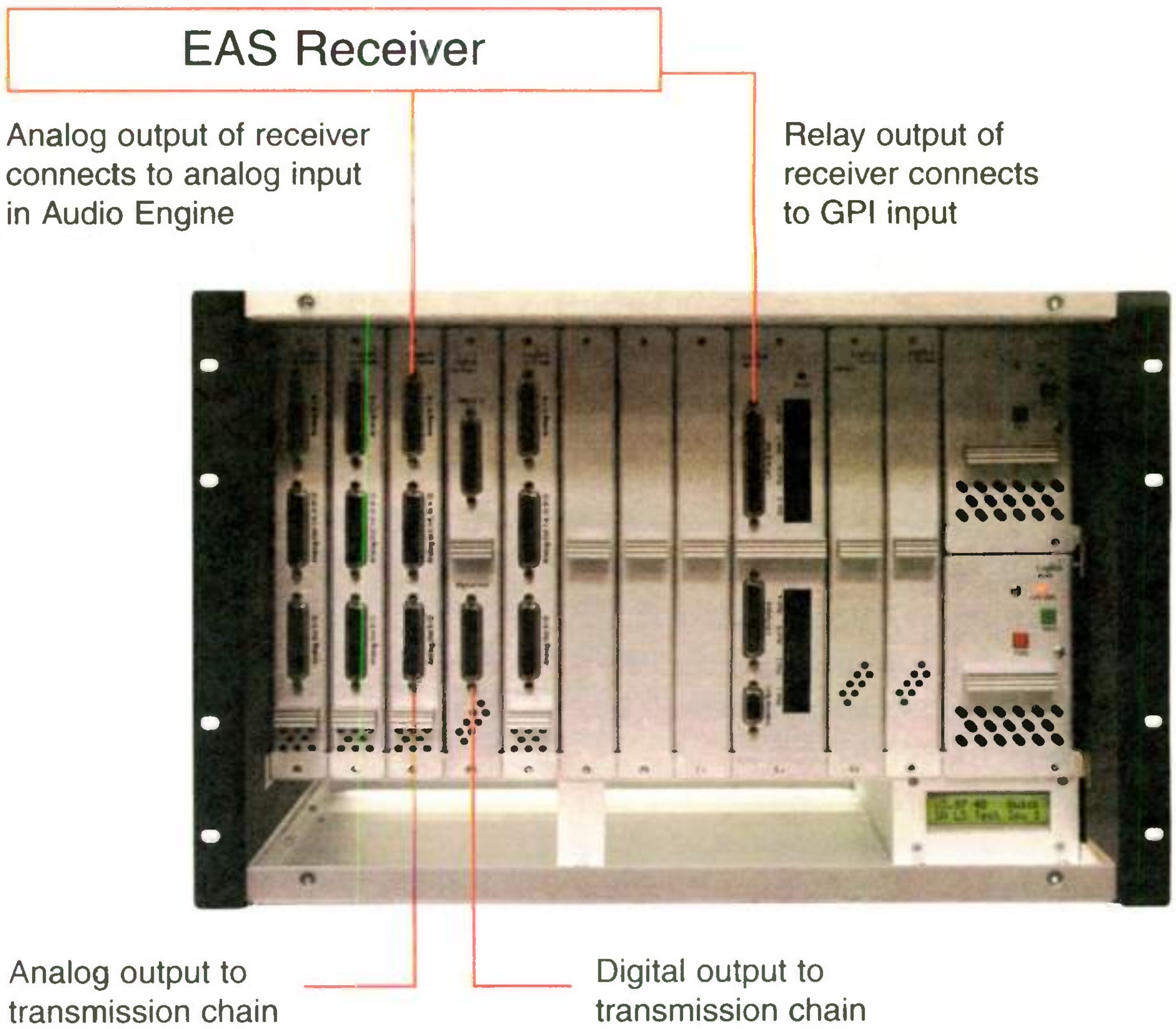
It's a common problem: *You have an analog EAS system located somewhere in your facility, and when it activates, you need to switch all of your program output channels (for all stations under your roof) to the EAS output.*

When you have a digital transmission chain, the problem is compounded: *How do you get that analog input to your digital output without a lot of trouble?*

THE LOGITEK SOLUTION:

Logitek's full featured digital audio router, the Audio Engine, is used along with a trigger set in the Logitek Supervisor software for the Audio Engine. The EAS receiver is connected to the Audio Engine as shown. Analog and digital outputs from the Audio Engine are set up for automatic switching to the EAS signal when the EAS relay activates. With multiple stations, a networked Audio Engine system will accommodate switching for everyone. When the relay releases, another trigger automatically reverts audio to the designated program and auxiliary sources.

With the Audio Engine as your audio routing source, you can provide a clean program feed to yet another auxiliary output so the EAS signal isn't included in program recording.



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HD CODER

CONTINUED FROM PAGE 8

Taken together, these results suggest that in general there is a difference in people's perception of quality at lower bit rates than at higher bit rates, and that this difference emerges between 36 and 48 kbps. With the exception of female speech, participants reported quality parity until 36 kbps. At 36 kbps, participants' scores ranged from "fair" (3.0 – 3.5) in voice-over and speech to "good" in classical and jazz (4.0). Notice that at the lowest bit rates the quality ratings dropped dramatically. At 24 kbps, participants rated most genres as "fair," and at 16 kbps participants rated samples between "poor" (2.0) and "fair" (3.3). With regard to female speech, participants were slightly more sensitive, noticing minor differences at bit rates as high as 64 kpbs.

A/B Test Comparisons

Table 9 shows results for which sample had better audio quality in A/B testing. As with Phase 1 participants, paired t-tests were conducted to see if the percentage of respondents claiming that the higher bit rate sounded better than the lower bit rate was statistically different from chance, or 50 percent.

Again, in keeping with Phase 1 participants, general public listeners were able to correctly identify the higher bit rate of the bit-rate pair at very low bit rates. The majority of participants heard differences between 24 and 36 kbps; 24 and 48 kbps; and 36 and 64 kpbs. The majority did not hear differences at 48 vs. 64 kbps, but a slight majority accurately reported hearing differences between 48 and 96 kbps and 64 and 96 kpbs. The t and p values indicate, however, that while significantly different from chance, the percentage of people accurately reporting differences was minimal.

In order to explore specifically where participants were hearing differences, t-tests were run for each genre at 48 vs. 96 and 64 vs. 96 kbps. Table 10 shows these results. T-tests showed significance in "speech" at 64 vs. 96 kpbs, but not at 48 vs. 96 kpbs. The inability to find significant differences by genre at 48 kbps is most likely an artifact of statistical testing: the smaller the number of responses, the larger the difference must be for statistical significance. Because the number of responses in the genre analyses was substantially smaller than the number included in analyses conducted for total responses, statistical differences did not show up. However, if results from 48 vs. 96 and 64 vs. 96 are taken together, there is a strong indication that more participants heard differences in "rock" and "speech" than they did in "jazz" and "classical."

Results from ACR-MOS and A/B testing support the notion that, for most music and speech, listeners either do not notice differences between HDC bit rates of 48 kbps or higher, or notice very small differences.

Bit-rate	Size of difference – Phase 2 listeners	Size of difference – Phase 1 listeners
24 vs. 36	5.23	4.17
24 vs. 48	5.27	5.06
36 vs. 64	2.75	3.53
48 vs. 64	2.13	2.51
48 vs. 96	2.55	Not given during Phase 1
64 vs. 96	2.04	2.49

Table 11: Size of Difference When Quality Was Identified Correctly

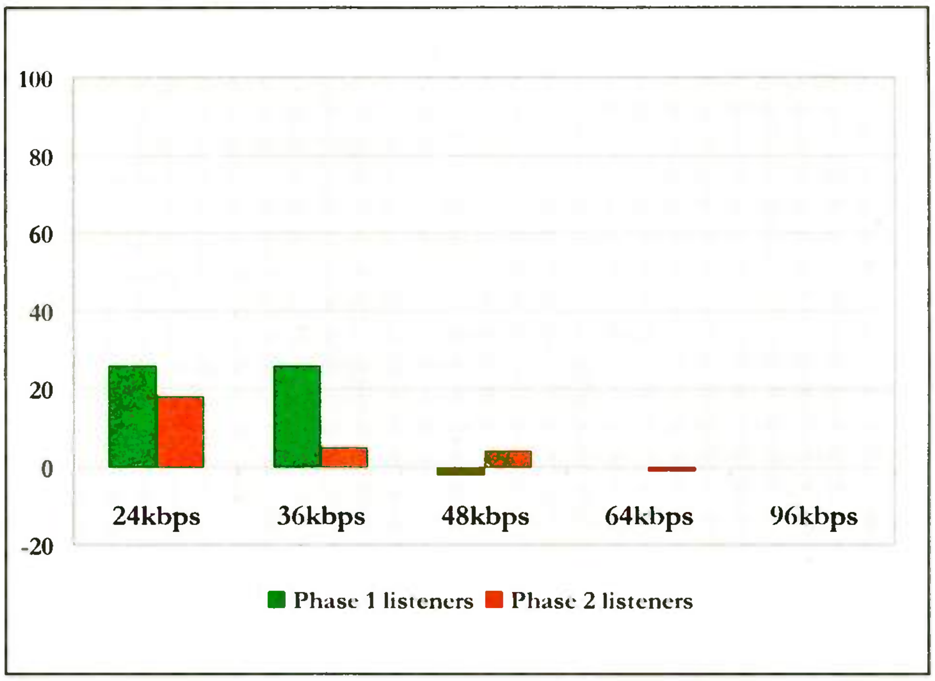

Fig. 5: Percentage Difference Between 96 kbps and Lower Bit Rates

Size of Difference

As in Phase 1, Phase 2 participants were also asked how big the difference was between the audio samples in an audio pair on a 1-to-10 scale, 1 being no difference at all; 10 being extremely different and noticeable. Table 11 shows these results for participants who correctly identified the higher bit-rate sample.

Note that participants claimed larger differences at lower bit-rates and smaller differences at higher bit rates. Further, a comparison of results from both phases indicates that NPR listeners and general public listeners rated the size of the difference similarly.

Listening Behavior

Finally, listeners were asked whether they would continue to listen to sample "A," sample "B," "neither" or "both" at various bit rates. Fig. 5 shows the *difference* between the turn off rate for 96 kbps and other bit rates. In this figure, 96 kbps was set to "0." The difference then is the additional rate of discontinuation participants claimed at various bit rates. Notice that fewer general public listeners reacted negatively to samples coded at 24 and 36 kbps than did NPR participants, but at 48, 64 and 96 kbps, the numbers are virtually the same. These results again indicate that between 36 and 48 kbps participants' behavior changes, with a large majority contending that they would maintain listening at 48 kbps.

CONCLUSIONS

Results from ACR-MOS and A/B testing support the notion that, for most music and speech, listeners either do not notice differences between HDC bit rates of 48 kbps or higher, or notice very small differences. As noted, these differences were heightened by allowing each participant to audition the choices as many times as needed to make a comparative decision, an opportunity obviously unavailable to radio consumers. However, participants do notice significant differences at lower bit rates of 16, 24 and 36 kbps.

As with previous testing, participants were more sensitive to differences when rating speech than when rating music and voice-overs. This is presumably due to reduced psychoacoustic masking opportunities (i.e., there is less masking of digital artifacts associated with speech's overall lower acoustic density and frequent wave front pauses) or because humans are particularly sensitive to voices and voice quality — or some combination of these factors.

Results from this study clearly indicate that for the HDC coder it is possible to separate 96 kbps into two 48 kbps streams with minimal, if any disturbance to listeners.

Interestingly, when making choices about bit allocation for HDC, it is apparent that music may require fewer bits than speech to maintain transparency. ■

AT LAST, A BOARD GEEKS AND JOCKS CAN BOTH DROOL OVER.

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MENDENHALL

CONTINUED FROM PAGE 1

In 1999 Mendenhall received the NAB Radio Engineering Achievement Award, recognizing his many innovations and contributions to the broadcast industry.

How did you get your start in broadcast engineering?

I started as a part-time broadcast technician in 1963 while still in high school, then worked my way through college doing engineering work at several Atlanta-area radio and television broadcast stations. While attending the Georgia Institute of Technology as an electrical engineering student, I designed, hand-built and FCC type-accepted an FM broadcast transmitter for WREK, the Georgia Tech student FM station.

After graduating with a degree in electrical engineering from the Georgia Tech in 1970, I designed radio communication products for several years before joining Gates Radio as a broadcast equipment design engineer. I was involved with many different design tasks at Gates/Harris including directional-antenna phasing equipment, TV demodulators and the MS-15 FM exciter.

Who else have you worked for in a design capacity?

From 1978 to 1993, I led the research and development activities at Broadcast Electronics, where my team developed a full line of AM/FM radio transmitters and exciters, including the FX-30 and FX-50 FM exciters.

In 1993 I rejoined Harris as vice president-radio product line manager, where my team successfully launched the Digit-CD, Z-FM, Super Power DX and CD Link products. In 1995, I assumed overall responsibility for all Harris Broadcast radio and television transmission product development with a focus on new digital radio and television products including the new DAX, 3DX and FlexStar HD Radio families of radio products. I am now leading the Harris engineering teams designing the next generation of digital-radio and high-definition digital-television transmitters.

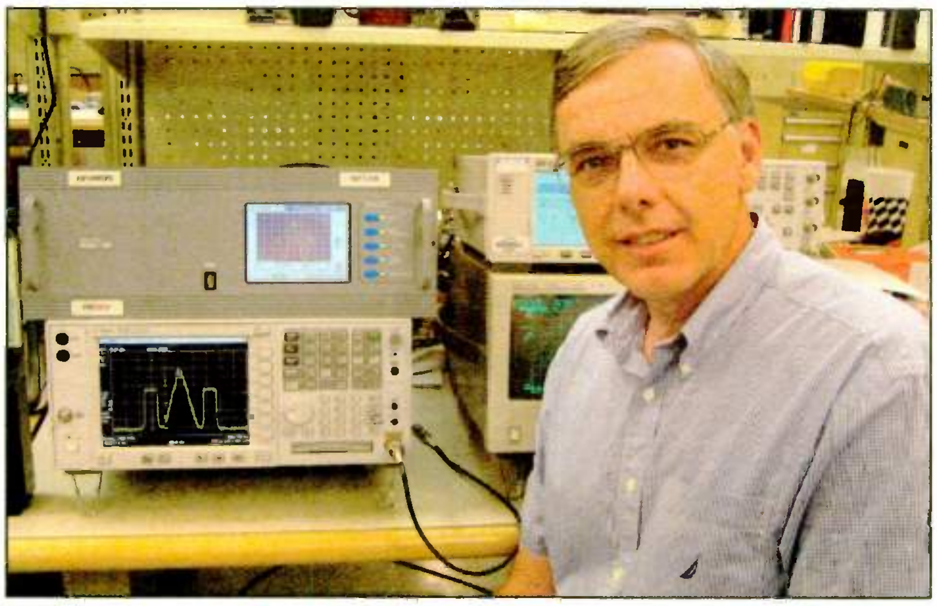

Fig. 1: Geoff Mendenhall in the Lab With Flexstar HD Radio Exciter

Can you talk about what it was like designing the MS-15 exciter?

The MS-15 was the Harris response to the Collins 310-Z2 FM exciter, which had begun to take market share away from the Gates TE-3. The MS-15 was also motivated by the introduction of quadraphonic FM broadcasting technology embodied in the RCA 4-3-4 system and the Lou Dorren 4-4-4 system. The MS-15 was designed to have optional quadraphonic generator modules plug into the backplane board.

Unfortunately, none of the discrete quadraphonic systems were ever adopted for broadcast use. Ironically, 5.1 surround sound is now one of the killer audio applications for HD Radio.

Designing the MS-15 was a lot of fun, because we had a team of smart, young engineers who were passionate about improving FM audio quality by an order of magnitude. We often listened to "rock and roll" music through the prototype circuits we were designing. Sometimes a new station would appear on the Quincy FM radio dial while we were working out the bugs in preparation for the first beta test. Today, another generation of equally passionate and highly talented engineers is listening to their favorite music through the new FlexStar HD Radio exciter.

The MS-15 was designed to be not only a significant improvement to the Harris line of FM transmitters, but also as an easy way to retrofit and upgrade existing FM transmitters of all brands. This strategy was repeated with later exciters including the Digit-CD, digital FM exciter and the new FlexStar FM/HD-Radio exciter.

How has the design process changed from the time you were first involved in manufacturing broadcast equipment?

The design process has evolved from a less structured, individual engineer-driven process to a customer input-driven, engineering team-oriented process.

Ten years ago, there were separate engineering departments for radio and TV products with little re-use of technology and no common look and feel between product lines. Even with an engineering organization as large as the one at Harris Broadcast Communications Division, it is difficult to have enough engineering resources to separately staff multiple engineering departments.

Technology has advanced at an accelerated pace that requires the span of design skills to be much broader if all of these new technologies are to be embodied in a new product design.

Several years ago Harris moved from separate radio and TV engineering departments to a centralized research-and-development function with teams of specialists focused on each part of the integrated system we call a transmitter. Some of the subspecialties we share across product lines are: embedded control systems, Graphical User Interface development, digital signal processing, digital hardware design, solid-state RF power amplifier design, power supply design and thermal/cooling system design.

The final integration of technology from the centralized research and development teams into a complete transmitter design is performed by the radio and TV product engineering team, which includes mechanical packaging, system integration and product performance verification.

Better and more accurate simulations of circuits are fully analyzed before these circuits are ever built. In the 1970s, computer modeling was done for the most part with in-house written, FORTRAN programs that ran on mini computers such as the DEC PDP-11. Early linear simulation tools such as SPICE and ECA were a big step forward with the availability of more powerful personal computers in the 1980s, but these tools couldn't model the nonlinear characteristics of RF power amplifiers. Today we have a complete suite of linear and nonlinear modeling tools that run on much faster computers.

One of our core competencies is the design of solid-state RF amplifiers and the associated microstrip or stripline combiner/splitter structures. Ten years ago, much of this work was "cut and try" based on approximate calculations. Today, we have electromagnetic modeling tools, RF device simulations and an automated load pull system to speed the optimization of RF power amplifier systems.

Early mechanical design tools such as AutoCAD were limited to 2-D drawings, but

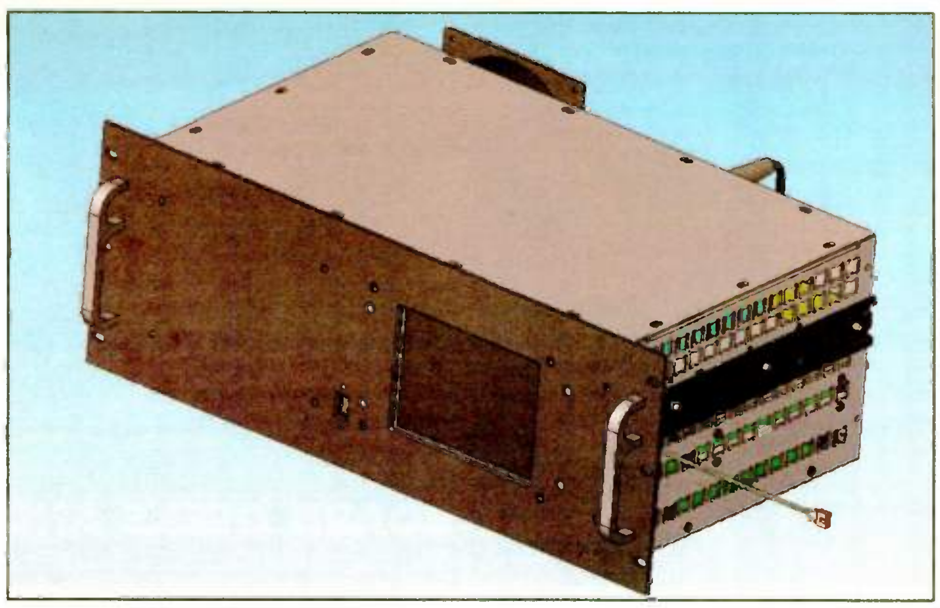

Fig. 2: 3D CAD Rendering of FlexStar HDx HD-Radio Exciter

this was still a big improvement over hand drawn parts. Mechanical packaging is now designed using 3-D, parametric design tools that allow all the parts to be verified for proper fit before they are actually fabricated. The parametric capability allows entire mechanical systems to be resized with all the associated parts changing together. This allows much more re-use of mechanical designs over a family of products.

Complex, air or liquid cooling systems are now designed much faster and better with the help of thermal simulations and IR imaging tools. *(See Figs. 2 and 3.)*

How does the design process work at Harris today?

Harris uses a multigate process to manage the new product development process from the concept stage through the release to production stage of new product development.

The beginning of any new product development begins with an idea that can come from anyone within the company or from a customer. This idea is then refined into the definition of the product features and performance specifications. The product-definition process involves the product management and engineering teams working together to get "voice-of-the-customer" inputs and integrate these requirements with new technology to create the definition of a new product.

A business case is then developed that fully defines the product features, specifications, market, cost and development schedule. After the business plan is approved the actual product development takes place; then a pilot run of pre-production units is built, which includes beta test units that are placed with key customers for evaluation and feedback. After a successful pilot run and beta test, the product is ready for full release to manufacturing.

Is there a set of principals underlying the design of new equipment?

Our current design philosophy is to re-use design elements that have been perfected in prior designs rather than "re-invent the wheel" every time we develop a new product. Re-using proven designs has many advantages including reduced design time, avoiding adding new parts to the inventory, and avoiding debugging from the ground up. The automotive industry has recently been very effective with design re-use.

There are limits to how much and how long designs can be re-used as technology changes. When technological advances offer enough advantages, then subsystems are redesigned to incorporate these benefits.

Harris has put a lot of effort into moving towards a more common look and feel in the user interfaces and control systems of our products. We have developed a common control architecture and communications bus that is now used across both radio and TV products. If a station engineer takes care of both radio and TV transmitters from Harris, he or she will find similarities between how the user interfaces of these new products operate, which reduces training time and makes troubleshooting easier.

The combination of a centralized R&D organization coupled with the best computer-aided design tools has allowed us to develop re-useable, scalable technologies and system architectures that are constantly refined in cost, reliability and functionality. These subsystems are re-used across product lines giving the customer a common look and feel. *(See Fig. 4)*

Let's talk about technology. What are some of the important developments that have affected transmitter design over the last 10 years or so?

Over the past decade, solid-state technology has steadily displaced vacuum tubes in broadcast equipment. The Harris broadcast transmitter product lines are completely solid-state except for high-power FM transmitters and high-power UHF TV transmitters. Solid-state devices first displaced tubes in the long/medium-wave AM transmitter product lines all the way up to 2 million watts because MOS-FET devices work with very high efficiency at this low-frequency range. There is no longer a price premium for solid-state technology at any power level in the medium-wave frequency range.

As the power handling capability of VHF field effect transistors has improved, solid-state FM transmitters are now becoming cost-competitive with vacuum technology up to 20 kW power levels. It will take a further 2:1 improvement in solid-state device power density to make solid-state cost competitive at the 35 kW power level. Many customers are willing to pay a premium for solid-state transmitters, because the redundancy of solid-state architectures offer much higher reliability, better technical performance and lower maintenance than tube technology. The total cost of ownership over the long haul is lower for solid-state transmitters.

What is the latest design challenge for FM service now that HD Radio is becoming a reality?

Class "C" or "D" amplifiers operate in a high-efficiency, saturated mode that is ideal for the amplification of a constant amplitude waveform such as FM. The transmission of digital-signal waveforms, including HD Radio, requires simultaneous amplitude and phase modulation, which means that the RF amplifier cannot operate in a fully saturated mode. A completely linear amplifier would need to operate in class "A" mode at very low efficiency like an audio amplifier. With advanced precorrection techniques, it is possible to use a quasi-linear amplifier operating in class "AB" that provides a moderate improvement in efficiency. Generally, class "C" FM amplifiers can be re-biased as class "AB" amplifiers with a back-off in power, but the efficiency drops and the cooling requirements increase.

This makes high-power FM transmitters, above 20 kW, expensive to purchase and operate in "hybrid mode," which is the

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common amplification of both the FM and HD Radio signals through a single power amplifier. Separate amplification with two

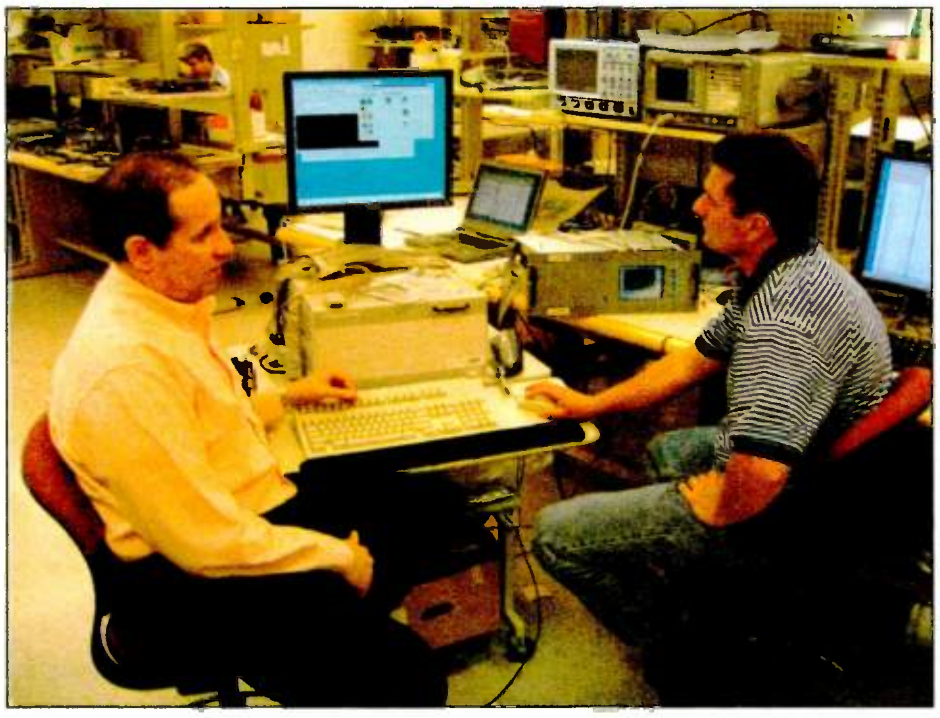

Fig. 3: FlexStar Design Team Members Performing Tests

transmitters that are high-level combined, split-level combined or space combined in separate antennas are alternative ways to solve this problem, but there are trade-offs.

Space combining has the advantage of eliminating combining losses, which reduces the power and cost of the HD Radio transmitter, but it has the disadvantage of not providing identical signal ratio tracking between the FM and HD signals at all receiving locations. Space combining in dual-feed antennas or closely spaced antennas has led to problems with maintaining enough isolation between the FM transmitter and the HD Radio transmitter to prevent RF intermodulation between the two transmitters.

The ideal solution would be a lossless, high-level combining system, but the extremely close frequency spacing between the host analog FM signal and the HD Radio sidebands makes this difficult without doing damage to the FM sidebands.

High-level combining involves some significant loss of the FM signal and many FM transmitters in service do not have enough reserve power to make up for this loss. Split-level combining is one answer to this problem, because it allows the existing FM transmitter to operate at its current power or even less power depending on the combining ratio with the HD Radio transmitter operating in common amplification mode. Split-level combining also provides a full-time, partial, analog FM backup by having a portion of the FM signal flow through the HD Radio transmitter in parallel with the main FM transmitter.

Research into better methods of combining FM and HD Radio signals at output of the two transmitters continues.

How about for AM or medium-wave service?

Pulse Duration Modulation has been the mainstay of high-efficiency AM modulation over the past three decades. It has many advantages for simple, analog AM modulation, but has limitations for digital-modulation waveforms such as HD Radio or Digital Radio Mondiale.

Digital-modulation waveforms require simultaneous amplitude and phase modulation with great precision to accurately reconstruct the RF spectrum. The digital signal is decomposed into separate amplitude and phase components that can be amplified in a high-efficiency AM transmitter and then recombined in the output stage of the transmitter. This technique is known as Envelope Elimination and Restoration or EER.

The audio filter required to remove the PDM switching frequency causes a nonconstant time delay to the AM envelope signal that must be perfectly time-aligned with the phase modulated RF signal in the output stage of the transmitter to accurately produce the digital spectrum with high efficiency. Large amounts of precorrection (predistortion) are required to accurately convey a digital waveform through a typical PDM transmitter.

Digital modulation, as implemented in the Harris DX and 3DX AM transmitter lines, completely eliminates the modulator and modulator filter by directly synthesizing the RF envelope at the output of a high-efficiency, RF, digital-to-analog converter.

The main advantage for high-accuracy EER reconstruction of digital waveforms, such as HD Radio, is that the time delay and amplitude matching between the phase and envelope signals is constant.

Many new products are now offered with switching power supplies but it seems they are not always as reliable as the older linear designs. What are the trade-offs between linear and switching power supplies?

A reliable power supply is fundamental to any product. This basic system component is often not given enough attention in the overall system architecture. It is often taken for granted and outsourced as just another off-the-shelf component, but it is so fundamental to staying on the air that it deserves more attention in the design process.

Switching power supplies can offer better voltage regulation, unity power factor, less weight, smaller size and lower cost. Although switch-mode power supplies offer advantages over linear power supplies, their complexity and parts count make them less reliable than a simple, linear supply, particularly at the higher power levels.

Switching supplies are usually used in a distributed, redundant configuration with one or more power supplies supporting each RF power amplifier. If one supply does fail, the system stays on the air at reduced power.

If the system architecture is based on only one or two power supplies, such as in a high-power transmitter, linear power supplies are usually chosen because of their higher reliability.

What are the trends in control systems for transmitters?

Twenty-five years ago, basic relay logic was replaced first by hard-wired, digital logic, and later by microprocessor/software-based control systems. There was always anxiety about the reliability of solid-state logic and microprocessors in transmitter site environments where a lightning strike could wipe out the control system.

Fig. 4: Distributed Transmitter Control Architecture Breadboard That Is Re-used Across Product Lines

Over the years, techniques for protecting solid-state control systems have advanced to the point where damage from lightning and

MENDENHALL, PAGE 20

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Digital Audio Analyzer

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- Bit statistics
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- Enables data storage in analyzer for later upload to PC
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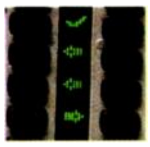

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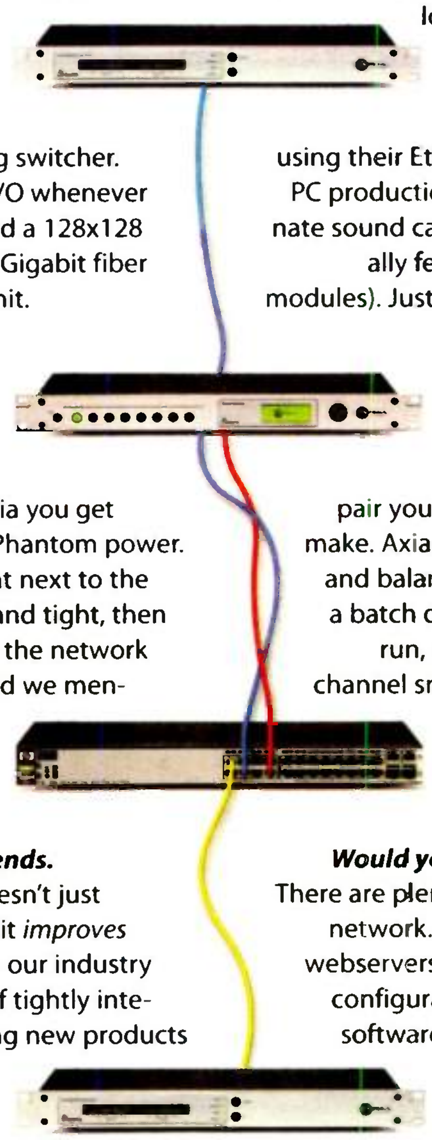

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The Broadcast Reliable Internet Codec

Telephone Company Change Will Force Broadcasters to Deploy New Codec Technology

By Tom Hartnett

The author is director of engineering for Comrex Corp.

I can imagine the conversation: "Back in my day," says the old broadcaster engineer to the new kid, "we didn't have these newfangled ISDN and POTS codecs."

It seems like a lifetime ago that every telephone company had its "broadcast division," and equalized loops were easy and cheap to get. When those started going away, analog frequency extenders filled the gap until technology progressed to provide the codecs we take for granted today.

But technology marches on, and not always to the beat of the broadcast industry. Sweeping changes are happening in the telephone network, and they threaten to once again change the way we do radio remote broadcasts.

Anyone who keeps track of technology is familiar these days with Voice-over-IP. Companies like Vonage and Skype are increasingly successful in bringing this to consumers, and enterprise level migration to VoIP is gaining momentum.

MIGRATION TO VoIP

It's popular with users because it can be delivered at low cost and (reasonably) high quality. The efficiencies it provides are not lost on the "old" telephone companies, either. Most major local and long-distance telephone providers have announced their intentions to migrate their existing Circuit Switched Data networks to IP. It is becoming increasingly rare for a phone call to travel entirely over CSD networks.

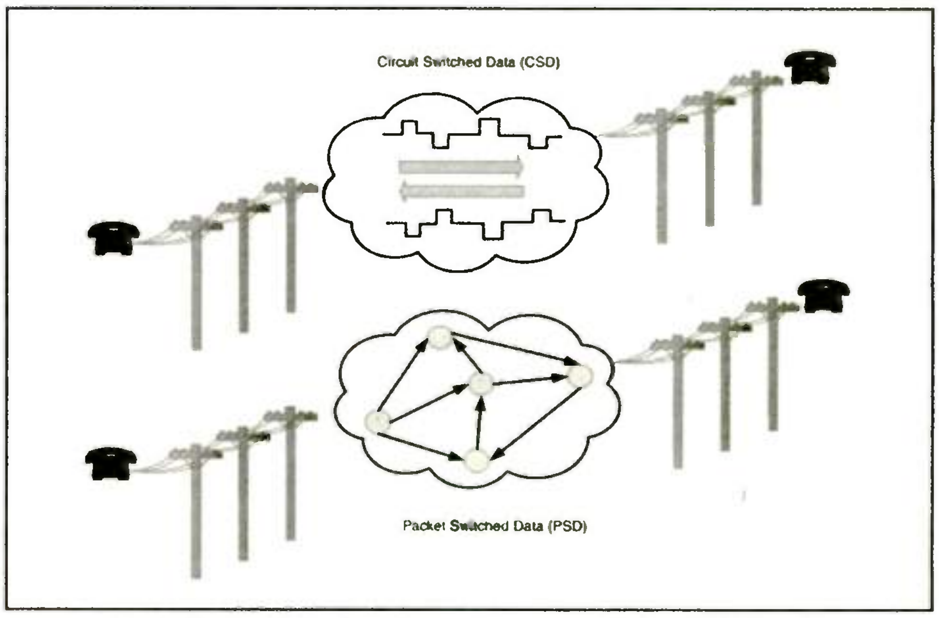

Fig. 1: Circuit Switched Data Network vs. Packet Switched VoIP Network

The difference between a VoIP and a CSD-based telephone network is shown in Fig. 1. Note that the change is often transparent to voice users, since the analog local phone line remains intact. This can pose a real problem for legacy equipment like modems (such as those in POTS codecs) and ISDN codecs that rely on CSD, because this gear often performs poorly on networks that utilize VoIP.

There have been standards developed to ease this migration by emulating the required protocols over VoIP. But the ISDN network never found its "killer app," and didn't really take off in North America except amongst broadcasters. It's already being phased out in select areas. Likewise, easy access to broadband Internet is making modems more rare. The fax is almost the only legacy CSD technology that still sees heavy, widespread use, especially over long-distance networks. Supporting fax emulation over VoIP links is much simpler and more common than supporting high-speed modems and ISDN. It remains to be seen how much effort is actually going to be put into supporting ISDN and modem connections over IP networks, considering

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MENDENHALL

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AC line transients has been largely eliminated. Most modern control architectures separate the basic "life-support" functions of the transmitter from the more sophisticated, remote monitoring functions. Microprocessor-based control systems allow the transmitter to optimize operating efficiency and to do self-diagnosis of most problems. With IP connectivity, it is now possible to replicate the GUI on the front panel of the transmitter from any PC on the planet with standard Web browser software.

The most reliable microprocessor control systems utilize embedded, real-time, operating systems that boot in milliseconds from EPROM or Flash memory rather than using PC operating systems like Windows or Linux that are less reliable and take longer to reboot.

It is interesting to note the change in broadcast engineering attitudes as the older generation of chief engineers has retired and the new generation of engineers, who grew up in the personal computer age, takes their place. The older generation was focused on the design and reliability of the RF amplifier and the control system simplicity. The new generation of station engineers is focused on the GUI, IP connectivity and control sophistication, but take for granted that the RF amplifier is reliable like the engine under the hood of a new car.

What new technologies that are being developed now interest or excite you? Do you see anything on the horizon that might be of interest to the radio engineering community?

The continuing evolution and rollout of HD Radio and Digital Radio Mondiale offer the radio broadcasters exciting new possibilities and new design challenges to the Harris design team.

In the HD Radio arena, we are now implementing multiple audio channels and new ways to consolidate the transport of multiple audio and data channels from the studio to transmitter via terrestrial and satellite venues. Harris will be continuing to push solid-state RF amplification technology to higher power levels with greater operating efficiency.

Audio coding technology will continue to improve with 5.1 surround sound becoming the norm in both home and mobile listening environments.

We will be continuing to evolve our control and monitoring capabilities to make equipment more automatic and to reduce the workload of the station engineer.

Looking further out, there will be many new digital content delivery and audio services for radio including interactivity.

Another exciting new broadcast technology is wireless, wide bandwidth, video and audio on demand to handheld devices similar to, but different from cellular telephones. These new services will be interactive and different from television and radio as we know it today. Harris is already involved in developing prototype equipment for these new services.

I think the future is very bright for wireless broadcast and narrowcast services because only wireless can deliver the content the consumer wants, when and anywhere the consumer wants it. ■

BRIC

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the diminishing return-on-investment from the Telco's point of view.

Even networks that support Modem-over-IP (MoIP) may not support it in a spread deployment of broadband-wired Internet (like cable and DSL), 802.11x (WiFi), and high-speed cellular data networks. The last year has seen deployment of two new, high-speed wireless technologies, Verizon Wireless' 1X-EV-DO network, and Cingular's EDGE network. In some environments like retail stores, it's becom-

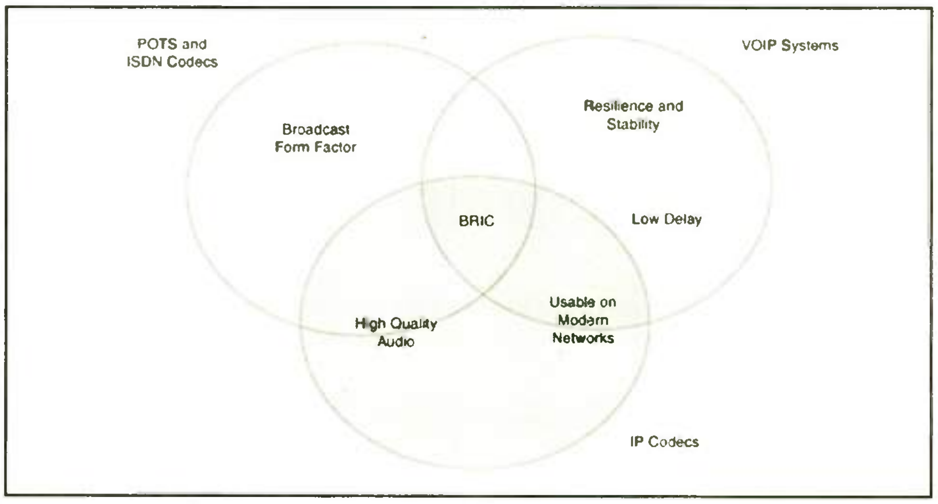

Fig. 2: Characteristics of Existing Technologies Combined in BRIC

POTS codec-friendly fashion. POTS codecs typically use a specific modem coding technique, which disables error correction and lowers channel overhead. This mode is so application-specific that it's unlikely to be emulated on MoIP channels. Finally, if emulation does work, it's not likely to rival true CSD, since adding a new, complex layer to an already somewhat fragile protocol like modems is not likely to enhance reliability.

Simultaneously, access to "IP dial tone" has increased dramatically through wide-

ing easier to access a WiFi hotspot than a dial-up phone line.

IP CODECS LACKING

We've seen in recent years the introduction of products coined "IP codecs." These are typically add-on modes to existing ISDN codecs, utilizing ISDN-type algorithms and wrapping the data into packets for transmission over IP networks. These can be useful in environments where the network traffic is managed to provide

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Control Freaks

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VAD-2

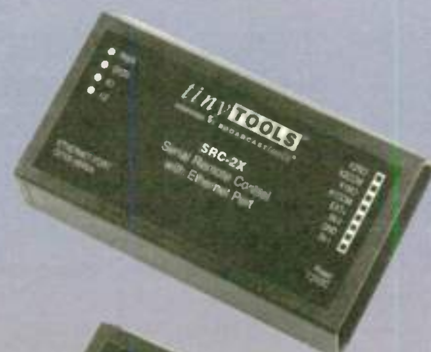
The tiny TOOLS VAD-2 is a user programmable two-input multi-number voice/pager auto dialer with integrated stereo silence sensor, designed for dial out paging and/or voice message notification. The VAD-2 is equipped with two dry contact inputs and stereo silence sensor, which when tripped, will sequentially dial a pager and/or up to four different phone numbers and play back a user recorded message corresponding to the tripped input. The VAD-2 also provides two SPST one amp relays for the control of external equipment.

WRC-4

The tiny TOOLS WRC-4 is a fresh approach to remote site monitoring and control, or providing an inexpensive solution to Internet enabling your present remote control system. The WRC-4 combined with web access and your favorite web browser brings you the following features; A powerful built-in web-server with non-volatile memory; 10/100base-T Ethernet port; four each channels of 10-bit analog inputs with a large monitoring range; optically-isolated status (contact closures or external voltages) inputs; normally open dry contact relays; open collector outputs; front panel status indicators, a single front panel temperature sensor and 4-email alarm notification addresses. The WRC-4 is also SNMP enabled. The WRC-4 has carefully been RFI proofed, while including the accessories other manufacturers consider optional. The WRC-4 is supplied with removable screw terminals and loaded with a generic web page that may be easily edited by the end user.

Time Sync Plus

The tiny TOOLS Time Sync Plus provides four separate GPS time referenced outputs. The first is a SPST relay, which pulses at 12:00, 22:00, 42:00, 54:30 each hour and is user programmable in each of four locations for any minute and second each hour. The second output is an active high driver with a 100 ms pulse each second, while the third output is a 4800-baud, RS-232 serial port providing a time zone adjustable hours, minute and seconds time code. The forth output provides an active high driver in the ESE TC-90 serial time code format. Indicator LED's are provided to display power/valid GPS data, programming mode and time sync relay operation. A Garmin 12 Channel GPS receiver with embedded antenna is suppled.



SRC-2/SRC-2x

The tiny TOOLS SRC-2 interfaces two optically isolated inputs and two SPST relays to a RS-232 or USB port, while the SRC-2x does this via a 10/100baseT Ethernet port. Both the SRC-2 and SRC-2x can notify a user's PC software program that any of two optically isolated inputs have been opened or closed and allows your software to control two SPST, 1-amp relays. The SRC-2x is also able to send an email when either of the two inputs change state. The user may also add up to 48 ASCII strings per input and 16 user defined strings per relay. Communication with the SRC-2(x) is accomplished via short "burst" type ASCII commands from the user's PC. Also, two units may be operated in a standalone mode (master/slave mode) to form a "Relay extension cord," with two channels of control in each direction. The SRC-2 communicates using RS-232 at baud rates up to 9600 and the SRC-2x via 10/100baseT Ethernet. The SRC-2(x) is powered by a surge protected internal power supply. Either unit may be rack mounted on the optional RA-1 mounting shelf.

ESS-1

The ESS-1 provides a cost-effective, small profile solution for standard serial-to-Ethernet connectivity. Designed with the broadcaster in mind, the ESS-1 is equipped with extensive RFI protection. It is ideal for applications requiring data support for both RS-232 and RS-422 communications. The ESS-1 allows any device with a serial port, Ethernet connectivity and is ideal as a serial bridge/tunneling or applications where a COM port, TCP Socket, UDP Socket, or UDP Multicast functionality is needed. The small profile of the ESS-1 makes installation hassle-free.

AVR-8

The AVR-8 is a voice remote control system that automatically reports changes detected on any of its eight status inputs to a remote telephone and/or pager. After speaking a greeting message that may identify the source of the call, the AVR-8 then speaks a unique message for each status input. The user may customize each factory-recorded message. After reporting, the AVR-8 is ready to receive commands through your telephone keypad. Functions include telling the AVR-8 to report on the input state of any of the eight status inputs, commanding the AVR-8 to pulse any one of its four SPDT relays for 750 ms and/or turning any one of the relays on or off. When a relay command is given, the AVR-8 speaks the relay 'name' followed by the 'on' or 'off' message.

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BRIC

CONTINUED FROM PAGE 20

priority to real-time audio, but their use on the public Internet has been disappointing. This is primarily due to the choice of audio coding algorithms.

The nature of algorithms like MPEG Layer III and AAC is that the codecs attempt near-transparent audio reproduction, using a fixed compression ratio. This means that their useful data rate is 64 kbps or above for most modes. It also limits the use of data-reduction techniques like voice-activity detection and error-hiding techniques like packet-loss concealment.

On the contrary, much work has gone on in the VoIP field to make real-time audio relatively immune to the congestive nature of the Internet. Although the voice coding algorithms usually only supply "telephone-grade" audio quality, they tend to degrade much more gracefully under high network jitter and packet-loss environments. The hardware used to access these services is not terribly broadcast-friendly; they usually emulate a standard telephone or are accessed from a computer desktop.

What is needed is a new way of providing real-time, duplex, high-quality, high-fidelity audio over existing IP networks. As shown in Fig. 2, it should combine the superior quality of IP codecs with the stability of VoIP systems. Lastly, it should exist in robust, road-ready hardware that can be easily set up and used, even by non-technical people.

SAY HELLO TO BRIC

As a successor to ISDN and POTS, BRIC, for Broadcast Reliable Internet Codec, is the answer to these needs.

By borrowing on the strengths of each existing approach, it delivers a way for radio broadcasters to utilize easily the available IP networks for real-time delivery of program audio. The first products using BRIC technology are the Comrex Access series of codecs, which we'll describe shortly. But first, let's introduce the concepts that make BRIC work.

Algorithms

The best way to assure reliable transfer of data over an unreliable network (like the Internet) is to reduce the amount of data sent. Even if your Internet access point is capable of many Mbps, its path beyond the "last mile" is completely unknown.

So BRIC utilizes an audio-coding algorithm capable of sending very high-quality speech audio (7 kHz mono) in a stream under 10 kbps. That's 1/6 to 1/12 the data rate of an ISDN codec. We refer to this mode as BRIC-UR (for ultra-reliable) and it's the default mode for the Access.

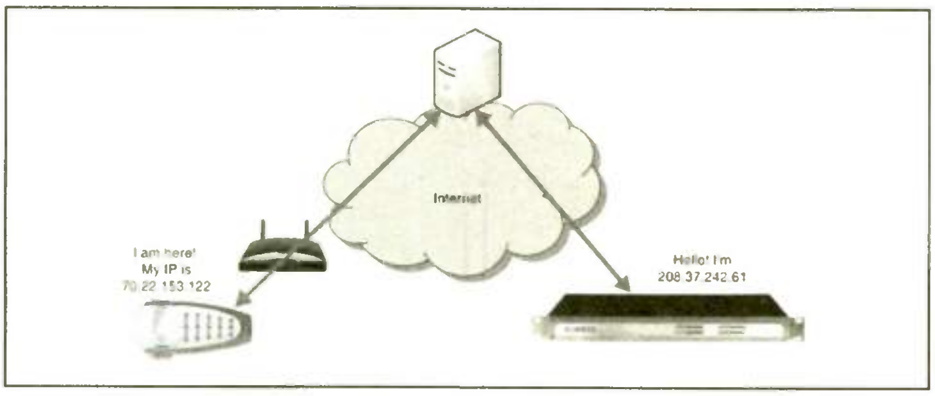

Fig. 3: BRIC Transversal Server

Another unique thing about the audio codec is that the data rate is variable. Since IP connections don't have a fixed data rate, it doesn't make sense that your codec should. BRIC-UR dynamically changes its packet sizes based on the complexity of the encoded audio and measured network congestion.

A second choice of audio coding is available in these codecs, called BRIC-HQ, for high quality. This is an algorithm capable of FM quality stereo at data rates around 24 kbps. BRIC-HQ can be enabled when the networks used are known to have reasonable data throughput. It should also be noted here that BRIC has a mode that can use point-to-point modem connections (like existing POTS codecs). In this mode, BRIC achieves 15 kHz stereo over a single dial-up phone call.

One of the biggest strengths of both BRIC algorithms is their low delay. BRIC-UR has an audio coding delay below 100 ms, with BRIC-HQ below 200ms. Although an IP network can sometimes add significant additional latency, which is out of BRIC control, the low delay number on a BRIC codec means even long network delays are often workable. Because audio delay and stability on IP networks are related, simple user controls are available to balance these two parameters.

NOT ALL NETWORKS ARE EQUAL

With their resilient data transfer modes, BRICs are designed to work reliably in very challenging network environments. The codec itself is quite reasonable to listen to on networks where packet loss approaches 30 percent. It's also important to realize that not all Internet access points are designed to support real-time audio. Many publicly available networks use Network Address Translation (NAT) and firewalls that limit the ability to connect VoIP technology to them. There's a third piece in the BRIC family to help with these issues: The BRIC Transversal Server.

Fig. 3 shows BRIC TS. It exists on the public Internet and is supported by Comrex. Use of an external server simplifies the process of getting around NAT and firewalls, since the codecs can maintain a "tunnel" through them to the server, and the server can deliver the current location of any other BRICs that are accessible through the Internet. The BRIC TS can provide information to build a "buddy list" of other BRICs, allowing easy connection, regardless of what type of Internet access they have. BRIC TS can also be useful in special applications, like when BRIC technology needs to be "bridged" over to legacy POTS and ISDN codecs, or when BRIC audio needs to be archived or distributed to multiple points.

Most major local and long-distance telephone providers have announced their intentions to migrate their existing Circuit Switched Data networks to IP.

ACCESS CODECS

The first embodiment of BRIC technology is the new Access codec, introduced at NAB2005. Like most broadcast codecs it comes in two flavors, one designed for fixed studio installation, the other for portable use.

Access Rack is designed to be a simple-to-use, "always-on" device. Network connections are made via either an Ethernet jack or a modem telephone port (or both). Because it's designed to always be connected to a LAN, user interface is done entirely via pointing a browser to the IP address of Access. An optional module is available for backwards compatibility with all earlier Comrex POTS codecs.

For field use, the Access portable is the really interesting part. This is about the size of a small camcorder and can run for extended periods on battery power. It includes Ethernet as a default but also includes a Cardbus (also known as PCMCIA or PC Card) slot to allow connection to other networks. These include 802.11x WiFi hotspots, new high-speed cellular nets like UMTS, 1X-EVDO and EDGE and dial-up modem connections.

Access portable is designed to be small and hand-held, but can be docked into a multichannel mixer accessory to provide better stereo mixing and headphone management.

SUMMARY

Migration away from legacy telephone systems is inevitable in the field of live broadcast audio. The BRIC system has been designed to be powerful enough to fill the need, yet simple enough to ease the transition. It provides a reliable and rugged way to utilize the public Internet to deliver high-quality audio between sites. Comrex Access codecs, the first BRIC-based products, provide a flexible platform for use of BRIC technology on a range of networks available today. ■

The Great AM Debate

Guy Defends His Proposals to Thin the Population of a 'Herd Full of Cripples'

Guy Wire, Radio World's masked engineer, is the pseudonym of a well-known radio veteran. Opinions are his own.

I've received numerous comments from readers on my column about saving the AM band, some of which appear in this issue of *Radio World Engineering Extra*.

It's always good to read and consider the many cogent and insightful observations of fellow engineers who care deeply about radio's future and well-being.

We hear a lot of criticism that profit-minded broadcasters have brought the current AM conundrum on themselves by insisting that more and more stations be crammed into an already crowded bandscape. The FCC is just as culpable by enabling the abuse, and now with HD Radio, the situation will only become much worse.

It's easy to cite the problems and the reasons solutions seem difficult and elusive. Yet I see few others stepping forward, as I have, to offer constructive ideas that just might help extricate us from the mess.

My proposal that the band needs to be thinned out and cleaned up is hardly a back-room conspiracy inspired by megagroups to drive small-market operators out of business, as some respondents suggest. It's rather about brainstorming an equitable plan to help the industry as a whole as well as those AM operators who are struggling to stay in business by offering them reasonable options.

These would either allow them to continue with their broadcasting enterprises using an alternative LPFM service, or at least provide them something of value in exchange for retiring their licenses. The existing stations that are economically viable and continue to provide local community service will most certainly have the opportunity to remain as they are. But most would agree such stations are in a dwindling minority.

STATION NUMBERS ARE UNCHANGED

Granted, AM stations are allocated on a demand basis and can be bought out of existence now, as some have. But there is no evidence of any ongoing natural attrition process that will thin the AM band on its own.

For every station that goes dark, a daytimer adds nighttime or increases power, or a new rimshot allocation is shoehorned in someplace. The number of AM stations in the United States actually has remained almost constant over 15 years, fluctuating between 5,000 and 4,700.

Even in the face of high conversion costs and the inability of many smaller AMs to afford adding HD Radio, the number of stations will not likely drop rapidly enough to allow for significant interference mitigation and a rebirth for AM with full-time HD.

It will no doubt be many years before such stations would be forced into deciding between going digital or going dark, unless mandated by the commission. If anything, the advent of multichannel HD-FM will drive the do-or-die economic issue more directly as formats carried on narrow-casted AMs migrate to multichannel HD-FMs.

The AM band clearly is a herd full of cripples and has been for a very long time. In order to create a cleaner band so that surviving stations can fully leverage the benefits of HD to attract significant audiences with competitive service both day and night, the marginal members need to be thinned out more quickly.

As long as there are buyers — mostly religious and ethnic, who want to serve small audiences — and as long as there remains the tenuous hope that HD Radio will somehow transform all AMs into high-fidelity head-to-head music competition with FMs, there is no incentive for owners to give up or turn in their licenses. Most will just hang onto them for tax-loss benefits, or sell to other narrow-casters to recoup as much of their investments as they can. And from there, the cycle just keeps repeating.

HD CONVERSION IS NOT THE TOTAL ANSWER

What many owners of marginal AM facilities might be overlooking is the plain and simple fact that adding the benefit of HD Radio will not fix deficient coverage problems. It could shrink coverage in many cases.

While most AM stations are not bothered by interference from other stations during the day, maintaining their coverage at night is heavily undermined by the onslaught of skywave. Even if an AM station has decent daytime market coverage, but its nighttime coverage shrinks dramatically, as many do, its ability to compete with FM stations and other AM stations that cover their markets well at all hours is severely compromised. Unless an AM station has solid full-time coverage of its target area or market, it's not going to be a player in the new era.

HD deployment has accelerated nicely for FM but has lagged badly for AM. There's a good reason most AM owners have not stepped up and made commitments to add HD Radio. Even without authority to use HD at night just yet, AM operators understandably are apprehensive about what's going to happen with increased interference and the probability other stations will file complaints forcing them to reduce digital power or turn off HD Radio at night. Unless you can run it full-time, the risks of committing to HD now are just too high for too many. What is needed is a cleaner environment in which concern about interference issues is not so overriding.

The remedies I propose would promote more attractive incentives for marginal station owners to sell and/or go dark; such remedies simply are not available now.

The first would be to establish new rules to allow trading in an AM for a commercially enabled LPFM of 100 to 1,000 watts where allocations rules permit. Understandably, there are few opportunities for such new channels in congested areas under the present FM spacing rules.

There has been considerable lobbying effort to relax third-adjacent spacing limits to permit more LPFM station allocations. While I was indeed concerned when the

GUY WIRE, PAGE 29

Products & Services SHOWCASE

Guy Wire Stirs Debate

I don't disagree with Guy's assessment (RW Engineering Extra, June 15) that running a small-market AM as a going concern is, at best, a challenge. And he's right that many operators have permitted directional antennas to deteriorate or have even taken to operating with day facilities at night. This isn't limited to just the small markets, either.

For that matter, the FCC has willingly facilitated interference levels that make night service on the former Class IV frequencies (1230, 1240, 1340, 1400, 1450, 1490 — now Class C) virtually useless.

Tune your FM to one of these frequencies that isn't assigned in your town. Typically there is a millivolt of skywave interference. This means nighttime "interference-free" service exists only inside the 20 millivolt contour. For a Class IV, this is about as far as you can see the top of the tower. So I agree there is a significant problem. Enforcement might get some of it but most is systemic.

The good news is that almost all of the solution ideas Guy suggests already exist. AM is and has always been a "demand-allocation" service. If desired, an AM licensee can turn in his license and it no longer needs to be protected from interference. Absent "demand," the FCC presumes no service is needed, so even if it is the last station allotted to a particular community, it can be turned off and disappear.

The rules also give wide latitude to so-called "interference-reduction agreements" between stations. These usually mean station A pulls in so station B can expand. The sale or modification of unproductive, interfering AMs is already possible. Generally, it's only a question of money.

And there's the rub. Guy says that licensees seeking to improve "shouldn't be held up for inflated prices by owners who sense a captive opportunity." While I appreciate the sentiment, Guy earlier says that these marginal AMs "get full-time life support from sister FM stations." So one man's "inflated price" is another man's payback for all those years of keeping an AM's heat and lights on.

And the idea that the Federal Treasury might be brought in will simply raise the price for cooperation and have all taxpayers footing the bill, whether it is cash payments, tax forgiveness or other free stuff for AM licensees.

I realize that since AM stereo we've all been a bit gun-shy about a "market-based" decisional process (though I would characterize the Motorola-Kahn fistfight as more "lawyer-based" than "market-based").

But for AM, since the trend lines are headed the right way anyway, I say let's adopt a hands-off approach and let technical and market nature take its course.

What should happen is what has been happening for years. AM stations disappear. They lose their licenses because they are dark beyond the one-year statutory limit of time. They lose their sites because development has made the tower land worth more than the license. The advent of HD will accelerate the process. A combination of the interference received and the investment required will spell the end for some more AMs.

Those AMs that survive at the margins will discover new things to do with a 50 kilobit forward delivery stream, even if it only works in the daytime. Don't you suppose a car parked for hours while the owner is at his desk represents an exciting delivery target for a podcasting-type technology? Our drive-time commuter might have his playback device loaded overnight in the garage from the wireless LAN for the morning commute, then again for the PM commute while it's parked in the employee lot. Hey, who knows? It could happen. If it gains traction, there'll be money to buy off some more marginal AMs. But I believe the likelihood of such innovation is reduced, not increased, by an intervention of the kind proposed by Guy.

> **For AM, since the trend lines are headed the right way anyway, I say let's adopt a hands-off approach and let technical and market nature take its course.**
>
> **— Frank McCoy**

Frank McCoy
American Media Services
Chicago, Ill.

So I guess we have come full circle. It turns out that there really isn't room for all those AMs that were shoe-horned in over the previous decades after all ... particularly with the out-of-band products that will be generated by so-called in-band, on-channel transmissions.

And what is Mr. Wire's solution? Simple: sweep away all those "useless" chattering voices of small broadcasters so that the corporate owners of the "big time" stations can maximize their profits without interference.

After all, the public has become used to corporate pabulum. Local service is no longer required. None of those diverse voices will be missed.

Mr. Wire, I got the message: Money talks, diversity walks.

John Higdon
Chief Engineer
Coast Radio Co.
KKIQ(FM)/KUIC(FM)/KKDV(FM)
Vacaville, Calif.

I agree that AM radio is worth saving, and that IBOC means nothing but trouble for small AMs. I'd go so far as to say that IBOC is bad news for almost all small broadcasters.

It's not that I'm opposed to digital radio. But a digital system that has a side effect of obliterating your neighbors hardly seems like a good idea. On the other hand, it is a great way for large, well-financed stations to eliminate smaller competition. A broadcaster's viewpoint about IBOC may depend on which side of the fence he is standing.

In no case will improved technology cure what ails radio. The problem is content, not technology. As a result of programming that ignores the taste and needs of their communities, lots of people don't

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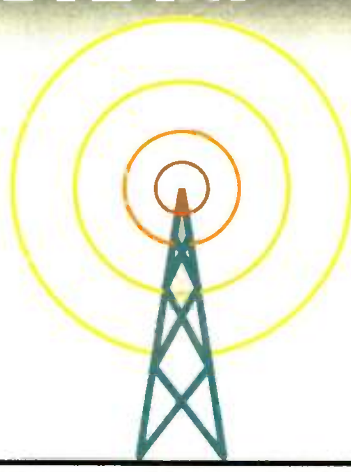

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LETTERS

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listen to the radio at all.

I almost fell out of my chair when I read Guy's comment that the solution to a cluttered band is to give small AM community broadcasters a nice shiny LPFM license, in exchange for surrendering their AM ticket. Where does he propose to put them? This is the writer who denounced the original idea of LPFM stations, claiming that the FM band was too crowded. That was five years ago.

In case he hasn't noticed, it's even more crowded today. And now there are IBOC sidebands to contend with.

When LPFM was proposed, there was adequate space to support more small stations, especially if the FCC's concept of second-adjacent-channel protection was to be made the law of the land. But the Radio Preservation Act of 2001 changed all that. Third-adjacent channel protection was made the rule and thousands of possible channels vanished. Even so, nearly 1,000 LPFM applicants have made it on the air or have CPs.

Although interference complaints have been practically non-existent, available space is getting smaller. Let's not forget about all those translator applications from the "Great Translator Invasion."

So is Guy suggesting that maybe we should change spacing requirements to accommodate displaced AMs? I'd think you'd have to; otherwise, "there is no room at the inn," unless you are willing to swap your AM license for a low-power FM in rural Wyoming.

Most AMs rely on commercial advertising, while LPFMs are prohibited from running commercials, though "Enhanced Underwriting" is OK. That's going to be a tough sell for many AM owners. They are used to selling commercials, complete with comparative statements, a call to action and pricing information. That's prohibited on LPFM. Maybe we could change that too. I'd be all for it, as long as it applied to all LPFM stations.

LPFMs and small AMs have a lot in common. They are not enemies, despite what they've had drilled into them by various trade associations. With the exception of the 50 kW clear channel stations, most AM broadcasters are decidedly local operations. Or they were; these days they are more likely to be relaying a signal from a bartered satellite network, running brokered foreign-language programming or playing Bible-thumping religious programming piped in from some distant city.

That's too bad, because with a little effort they could once again become valuable assets. The concept of "community" broadcasting was one of the ideas behind LPFM. In fact, the FCC has used LPFM as a way to restore a little localism to the dial. It seems that this new generation of low-power stations is taking up where traditional small-town AM broadcasters left off.

Maybe it's time for some kind of national alliance of small broadcasters. Many of these people may have a lot more in common than they suspect. Now if we could only figure out how to make those IBOC sidebands go away, broadcasting might be fun again.

Chuck Conrad
General Manager
KZQX(LP)
Kilgore, Texas

Ed. Note: Additional readers' letters on this topic can be found in the Sept. 9 issue of Radio World.

GUY WIRE

CONTINUED FROM PAGE 24

service was proposed and launched, the record will show that I later supported LPFM and fully acknowledged that the FM service already successfully supports third-adjacent and many second-adjacent same-market facilities. I chided the NAB that it could not "have it both ways," opposing new LPFM drop-ins on technical grounds while supporting the reality of hundreds of existing, fully licensed, grandfathered, short-spaced stations and many translators that comfortably coexist.

The FM service can tolerate higher-density allocation of adjacent-channel operations because each channel has built-in guard-band protection from its first-adjacent neighbor. Plus, VHF propagation stays mostly constant from day to night. The AM band was never so blessed or so lucky. And based on the ongoing rollout of HD-FM stations, adjacent-channel interference has remained almost a non-issue. Changing the FM spacing rules to accommodate more noncommercial community-service LPFM stations as well as replacement commercial-channel LPFM stations for retired AM licenses could well be the key to making this concept work.

FINANCIAL INCENTIVES ARE NEEDED

My second proposal to jump-start AM cleanup is to empower the government to award tax certificates and allow accelerated depreciation write-offs so that more stations will go dark sooner than later.

This would make it easier and more efficient for remaining stations with more potential to buy out lesser facilities to allow increased power and expanded coverage, ensuring HD can be successful for them.

Getting the government to offer these options should not inflate station prices but instead have offsetting effects on market values for stations that go dark since the exchange gives the seller something of a more tangible and immediate value rather than gambling on the future.

If time limits were imposed on deals that allowed weaker stations to go dark and sell to stronger operators, sellers would have no incentive to hold out for more money. They could take the one-time opportunity without letting natural market forces continue to erode their investment downward.

Taxpayers would be indirectly subsidizing the tax certificates and write-offs; but the total amount of money — involving perhaps a thousand stations that may be worth on average less than $1 million each — is a meaninglessly tiny number on the national balance sheet. The value of such tax certificates would only be a percentage of their total market value. Plus, the sellers of many such stations would realize real-estate sales profits and pay capital gains taxes on those that provide further offset.

If 1,000 or more stations can be permanently retired via tax certificates or subsidized sales to stronger owners, and another thousand traded in for high-power LPFM stations, the majority of the remaining 2,700 will be able to breathe a lot easier. They certainly would have a better shot at obtaining improved full-time coverage for their markets.

We must remember that if the FCC authorizes full-time HD AM and the interference fallout becomes as serious as many are predicting, the government will have willfully contributed to the probable demise of many radio stations, invested in by owners as stewards of a public trust. Those owners fully expected those investments to be protected by technical standards and not diminished by the FCC allowing a new technology to significantly obviate the intent of those standards.

LITTLE HELP

The AM band, once proud and powerful, has become an overpopulated and polluted resource, not unlike a popular resort area that allowed too much development and abuse. After a while, the lake got fished out, the wild game disappeared, too many noisy neighbors spoiled the experience and property values plummeted. Folks stopped coming and went elsewhere.

Except for imposing NRSC bandwidth limits, the FCC has done little to help clean up our decaying AM resort.

Imagine the possibilities: An AM band that could once again restore wide-area groundwave and skywave coverage with limited interference for many more of its stations. A band that could truly showcase the enhancements of HD Radio on remaining stations, whose owners are willing to keep and improve them and in turn benefit from such a renewed and protected resource for all to enjoy in the future.

The AM band enjoys attractive propagation advantages over VHF, including superior coverage in areas blessed with good ground conductivity, and dramatically expanded coverage at night via skywave. These advantages should hardly be wasted on data-only services or turning the band over to taxi and police radios or utility companies, as some AM naysayers have suggested. Our government should be prepared to assume some responsibility for the AM stations' dilemma and offer meaningful amelioration to promote this general concept. Clean it up and then improve the capabilities of the service to allow its more committed and able custodians to make it flourish in both traditional, and yet undiscovered, ways.

Sadly, I have few illusions the NAB or FCC will take my suggestions seriously. The NAB's interests lie in maximizing the number of stations and dues-paying members, so any plan to reduce the station count does not serve that end. The commission typically avoids hard decisions that carry risk, even those that could significantly benefit the industry, as its long, disappointing track record demonstrates.

Nonetheless, the ideas proposed are worth defending and exploring.

I'm hoping some of the more enlightened folks at both institutions will prove me wrong and dare to look outside their bureaucratic boxes and special-interest obligations. It shouldn't be that hard for a few key individuals to think about what might be achievable and hopefully initiate a Notice of Inquiry proceeding on this topic.

I look forward to the ongoing debate and invite more input from readers on this modest proposal. E-mail to *gwire@imaspub.com.* ■

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The Broadcaster's Choice: Your Space or My Space

By Barry Blesser

The obvious approach to broadcasting an announcer's voice becomes only one of many choices if we examine the hidden assumptions buried in our modern traditions.

Consider those assumptions. A broadcast studio should be acoustically isolated from all external sounds. It should have sound absorbing surfaces to suppress reverberation, resonances and reflections. It should have microphones placed a few inches from the announcer's mouth. By scrubbing the studio of spatial information, the space becomes the aural analog to a sanitized hospital operating room — the announcer's voice is "pure."

The tradition of spaceless sound arose from the early days of Edison recording and primitive broadcasting, all of which suffered from weak signals and high noise. Close microphones originated almost a century ago as a solution to an otherwise insolvable technical problem. The solution survives the problem. Close microphones remove acoustics.

There are, however, other choices.

Dr. Barry Blesser is director of engineering for 25-Seven Systems, a former associate professor at MIT and past president of the AES. This introductory discussion on hearing space is an extract from the author's new book provisionally titled "Aural Architecture," to be published by MIT Press in 2006.

YOU ARE THERE

In the 1930s, NBC negotiated the rights to transmit live performances of the New York Metropolitan Opera, made famous by its host of 40 years, Milton Cross. For many reasons, microphones were not close to the singers and musicians, and audience sounds and spatial reverberation were part of the broadcast. Rural farmers sitting in their kitchens had the feeling of being at the opera, sitting with the audience in the opera hall.

Similarly, in the days of radio theater, special effects created the experience of a haunted house by adding the sounds of creaking floors, spatial resonances and reverberation. The resulting illusions of a specific space, even with primitive techniques of the 1940s, were compelling. Listeners were in the house.

Using Marshall McLuhan's innovative idea that the "medium was the message," radio is a hot medium, while television is a cool medium. Radio requires imagination; the active use of imagination actively engages the listener. He is absorbed into the experience that he creates in his head.

This raises the issue of how we experience a space. Real spaces have an aural personality that originates from two sources: its unique sounds and its local acoustics.

A forest has the sounds of birds and rustling leaves combined with the acoustics of dense foliage. An old-fashioned railroad station has the sounds of train wheels combined with the cavernous echoes of a grand space. The combination of signature sounds and acoustic personality creates the *soundscape*, the aural analog to a visual landscape.

In the case of an announcer in a sanitized studio, there is no soundscape. Listeners experience his voice only in *their* soundscape, be it in their automobile or local athletic gymnasium. And with headphone reproduction, there is neither an originating nor a listening soundscape. The aural experience of the voice is *spaceless*. Yet both announcers and listeners have to be someplace.

Human beings evolved an auditory system that can experience spatial attributes even though most of us remain oblivious to that ability. Try a simple experiment. Walk toward a wall with your eyes closed, stopping when your nose is just a few inches from the surface. We hear the wall. We hear how the wall changes the spectral balance of background sound, a kind of bass boost. Similarly, we can hear an open door and the depth of a cave.

Some blind individuals, as illustrated by Ray Charles and others, can ride bicycles in mountains and city streets without crashing into obstacles. If we can hear space, why should broadcasters remove the spatial personality of the studio? Like the mixing engineer producing recorded music with spatial synthesizers, broadcast engineers can also provide a virtual space for an announcer. Creating a soundscape and spatial texture for the announcer's voice is another choice at the opposite extreme from our current tradition. The technology is available to support such an artistic approach to space. While experiential illusions are part of 21st century media, radio remains anchored in the archaic past.

The topic of soundscapes is broader than radio. For an introduction to the topic of hearing space, listen to the BBC program "Acoustic Shadows" by visiting the link *www.bbc.co.uk/radio4* and scroll to Science.

SUBTLE WARMTH

While our culture thinks of experiencing space entirely as visual attributes, there is a larger tradition of experiencing space by its sensory architecture. Eyes are only one means of sensing an environment. Clearly, broadcasting technology cannot transmit olfactory or tactile experiences, but to an extent radio can broadcast the aural experience of space.

Part of the explanation for our culture's lack of interest in aural architecture arises from our preoccupation with vision, as exemplified by the dominance of television over radio. Nevertheless, there are situations where the eyes are otherwise unavailable, and where the world is entirely aural. In the rehabilitation profession, it is known that those with an aural deficit have a more difficult burden adjusting than those with a visual deficit.

For those who are not enchanted by arguments for including a soundscape, consider that the addition of synthetic space modestly boosts the perceived loudness in a way that cannot be duplicated by a compression processor.

This is especially true for the speech of a male announcer. Reverberation reduces the peak-to-average ratio by smearing energy over a wider time span, but without creating an unnatural sound. Reverberation is natural. A few broadcast engineers already include modest amounts of reverberation as part of their dynamics processing chain.

While broadcasters look for new ways to capture listener head space, subtle forms of experience create attractive warmth. Listeners need not be presented only with high-impact, in-your-face audio experiences. As I said in the opening sentence, there are choices. ■

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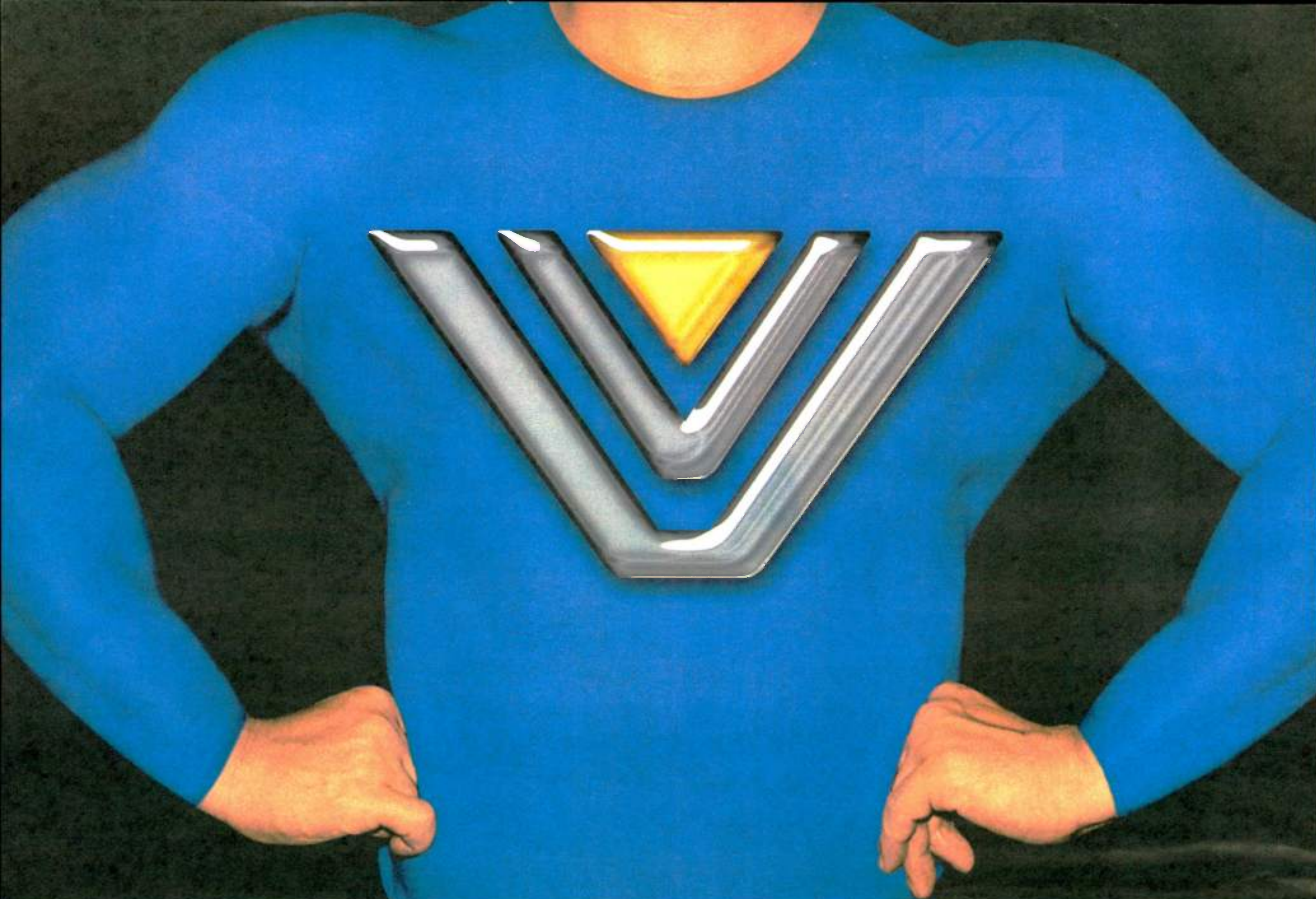

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