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Digital Phasor Jumps AM Array Hurdles

AM Directional Antenna Control System Offers Economical Solution, Eliminates Unwanted Networks

by Mario Hieb, P.E.

The author is president of Phasor Physics Inc. in Salt Lake City and a frequent contributor to Radio World.

As the IBOC transition to digital transmission moves forward, one of the biggest challenges will be the implementation of the AM-IBOC system with AM arrays. There are many technical issues involving directional arrays; solutions to these are in the theoretical stage and are moving toward realization. Manufacturers and consultants are currently conducting tests.

One such solution is the Digital Phasor, made by Phasor Physics and for which I have a patent with other patents pending. It is a computer-based control system that provides the functionality of a conventional phasor but with fewer drawbacks.

The system is still in the R&D phase and not yet for sale. While some testing has been performed and theoretical properties confirmed, more testing is planned before a system will go on sale.

As an introduction to the Digital Phasor and its relationship with IBOC and AM directional transmission systems, it is appropriate to review AM directional and non-directional systems.

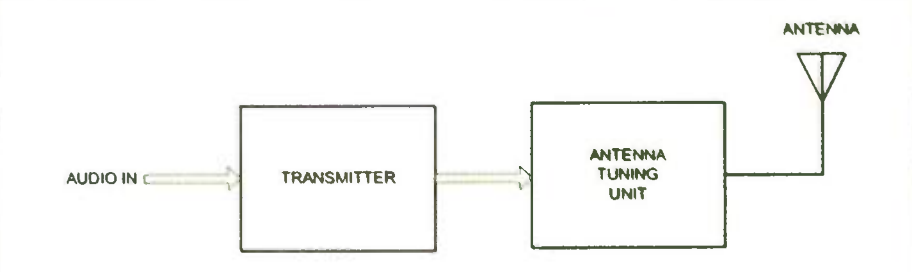

Figure 1. Basic Non-Directional AM Transmission System

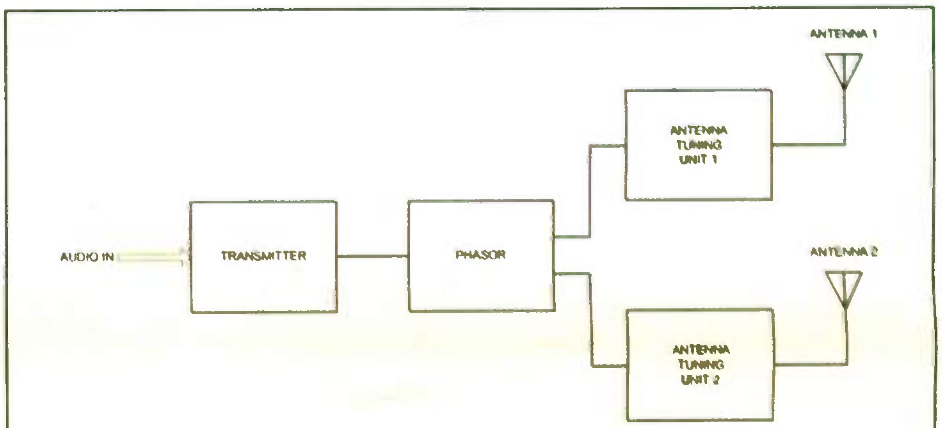

Figure 2. Basic Directional AM Transmission System

I. AM SYSTEMS REVIEW

Fig. 1 illustrates a basic AM transmission system. A transmitter, modulated by a program audio signal, drives a transmission line that feeds an antenna-tuning unit, or ATU. The ATU matches the transmission line to the antenna.

In a traditional directional AM transmission system, as shown in Fig. 2, the transmitter drives a phasor, a network of capacitors and inductors. Each output of the phasor feeds a transmission line of predetermined length.

An antenna monitor, driven with current samples from each antenna, provides measurements of the phase and current magnitude relative to one of the antennas chosen as a reference tower. The phasor is adjusted to a pre-determined parameter via visual feedback from the antenna monitor.

The purpose of the phasor is to provide a load to the transmitter; divide the power from the transmitter in a pre-determined ratio to a reference antenna; and delay the signal to each antenna by an amount relative to a reference antenna.

Fig. 3 shows a block diagram of the traditional phasor. The size, complexity and cost of the traditional phasor vary with the following

SEE PHASOR, PAGE 4

DESIGNER INTERVIEW


Dr. Ellyn Sheffield

Listener Research Charts IBOC Future

Dr. Ellyn Sheffield Explains Her Work in Digital Radio Studies and the Importance Of Unbiased Tests

by Steve Callahan

It's a brave new digital world out there. While radio engineers once expressed a healthy skepticism about the use of lossy digital compression, the industry is now busily deploying HD Radio systems using the latest digital coders. Many stations are even

SEE SHEFFIELD, PAGE 22

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FROM THE TECH EDITOR

Michael LeClair

The AM IBOC Digital Conundrum

Where Do Factors Such as Coverage, Interference and Audio Quality Stand in the Deployment of HD AM?

I have been waiting for some time to have the chance to make first-hand tests of the HD Radio system on an AM station. Last month I finally had that chance. Armed with a field strength meter, spectrum analyzer and Kenwood HD-capable radio, I set out for a day of measurements, both scientific and subjective.

There has been a swirling controversy around the design of the AM version of IBOC that has refused to go away. Proponents point to the potential for greatly improved audio quality through the use of digital transmission techniques, bolstered by the powerful use of psychoacoustic audio coders as reasons why AM should convert.

Indeed, this argument is powerful for AM HD; in this era of inexpensive and sophisticated electronics, most consumers can easily distinguish the lower quality of AM from just about any other audio delivery system, including the ubiquitous Internet.

The susceptibility of AM to man-made noise has resulted in an erosion of noise-free coverage area as everything from power lines to home security systems have proliferated in recent years, causing interference that renders AM service unlistenable in locations where it worked previously.

On the other hand, opponents to HD have noted that the system relies on generating significant energy on adjacent radio channels, well in excess of the technical standards that have been in place on the AM band for decades. The more frantic of these opponents have suggested that widespread deployment of HD will render all AM listening impossible. More reasoned analysis shows that HD will quite likely cause some interference, particularly where station allocation has been tightly squeezed.

LISTENER'S VIEW OF INTERFERENCE

Before a large number of knowledgeable people bolt up and run to their computers to take me to task for using the term "interference" incorrectly, I would like to emphasize that in the above sub-head I am using the term in an unguarded way. Not as an engineer, but rather in the way a listener might use it when describing the effects of a local AM station on their telephone service.

Sure, we know the law does not strictly protect telephones from AM signals and therefore we in the radio industry don't call it interference. But to the home user it sure sounds like "interference" when you hear other voices and music jingles on your phone. Likewise, a steady hiss of white noise caused by another station lighting up their digital carriers is going to sound to a listener like some kind of interference.

I would argue that when we talk about the potential for HD Radio to generate "interference" it would be a more honest approach to consider what the end listener is hearing.

But if we are going to try to be honest about interference, we should also be honest about the word "coverage" as well.

At the heart of this argument is the wide variation in how people measure the coverage of an AM radio station. When I look at coverage maps generated by engineering firms in the past it is typical to include contours out to the 0.5 mV or even the 0.25 mV.

Such contours may have been appropriate a long time ago when few alternatives existed for radio listening, man-made noise was less prevalent and the AM band allocation scheme was highly protective. In the case of clear-channel AMs, these contours may still work fine.

But for the large majority of AM stations, having significant listenership out to these distances is nothing but a fantasy. These days in AM it is more realistic to talk about the 2 mV contour as the coverage area within which most people could pick up a signal that is listenable for more than a few minutes at a time when driving. Coverage into buildings and houses with typical portable radios requires even more signal strength.

Given the differences of opinion on the meaning of words such as "coverage" and "interference," it's not surprising that we have experienced quite an argument about the relative benefits or negatives of AM HD. To summarize my point of view — others are going to differ — the AM stations I work with are lucky to get what I would call coverage out to the 1 mV contour. Outside of this area, interference is so prevalent that effectively it is the limit of reception and most listeners would acknowledge this.

Given the differences of opinion on the meaning of words such as 'coverage' and 'interference,' it's not surprising we have experienced an argument about the relative benefits or negatives of AM HD.

A BUDGET INSTALLATION

Getting back to my listening and coverage tests, let me give some background on this particular installation.

The AM station is a local channel with 1 kW unlimited operation. Probably a thousand similar stations exist across the United States.

But the large number of these stations means that coverage is limited by interference to generally no more than about 10 or 15 miles.

The transmission system is a simple skirt-fed radiator with a relatively tall tower, 135 degrees, that supports an FM station and other users. The tall tower affords fairly good analog coverage for this AM station. There is a co-channel station located about 70 miles away with similar facilities that limits the coverage in that direction.

This installation was what I would call a budget approach to HD. We used the existing PDM-style transmitter, about seven years old, and no effort was made to optimize the antenna system impedance with phase rotation. We simply purchased the necessary AM exciter (a Dexstar), modified the transmitter slightly to allow remote switching from internal to external oscillator and plugged it in.

With help from Harris Field Service, we adjusted the Dexstar for best third-order intermodulation performance to meet the Ibiquity recommended mask. A reduction in the digital sideband power below the allowable maximum was needed in order to keep

SEE LECLAIR, PAGE 4

IN THIS ISSUE

Phasor

CONTINUED FROM PAGE 1

parameters: number of towers; number of patterns; and transmitter power output.

The pattern of a directional system is determined by tower spacing and geometry; ratio of current delivered to each tower, relative to a reference tower; delay of the RF signal, relative to the reference tower; and mutual tower coupling.

Mutual coupling

Mutual tower coupling is a phenomenon whereby current from an adjacent tower is induced and then absorbed or re-radiated. In some array designs, a tower may absorb more power than it radiates; such a tower is described as having negative impedance.

The bottom line is that we do not know exactly what the driving point impedance of a tower will be until everything is turned on and tuned up.

Phasor issues

The traditional phasor has drawbacks such as using large, expensive reactive and mechanical components. Components are usually de-rated up to 50 percent of current and voltage ratings. Expensive vacuum capacitors often are employed.

Higher power requires larger and more expensive components. Additional patterns require more components. Each pattern requires its own phasor and, often, ATU network.

Tuning is difficult due to interacting components, and there is a limited range of adjustment.

Phasors are susceptible to failure due to high-power levels and lightning, and are prone to drift due to environmental factors, such as wet ground system or dry ground system.

Distortions may be caused by non-constant bandwidth impedance. The common point impedance varies from carrier frequency to sideband frequencies due to reactive components, causing poor audio performance in analog AM and bit errors in AM-IBOC.

There are problems with pattern bandwidth. The directional pattern varies from carrier frequency to sideband frequencies. The directional array may not adequately protect adjacent-channel stations and audio performance is affected.

It is difficult to match phasor and ATU to transmission line impedance.

Additionally, phasors must be custom-designed and custom-built.

II. AM-IBOC IMPLEMENTATION

AM-IBOC overview

AM-IBOC is different than analog AM in that encoded digital information, which a properly equipped receiver can use to re-create the audio signal, is sent along with the conventional AM signal in the hybrid mode or in place of the conventional AM signal in the all-digital mode. The hybrid mode will most likely be implemented first, with stations migrating to the full-digital mode, as IBOC receivers become commonplace.

The digital information is carried on additional digital sidebands that extend out

SEE PHASOR, PAGE 6

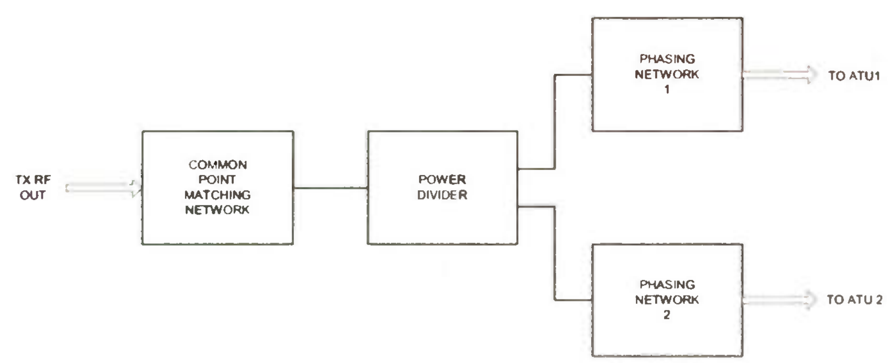

Figure 3. Block Diagram of a Directional Antenna System Phasor

LeClair

CONTINUED FROM PAGE 3

the intermodulation to acceptable levels.

Intermodulation is a particularly important factor to consider for reasons of interference generation — not to a neighboring station but to the HD radio itself. When the intermodulation products get too large the digital radio has trouble decoding the desired information and digital coverage suffers. Thus it will provide better digital coverage if you keep the power down and meet the mask, rather than trying to operate with a distorted maximum power.

COVERAGE TESTS

We tested the coverage of the digital signal using a Kenwood model HD receiver mounted in one of our station vehicles. When the signal began to switch out of digital on a regular basis (defined as multiple dropouts over the space of about 20 seconds) we would pull the vehicle over and measure the field strength.

As expected, a reliable digital signal extended out to about the 0.8 mV contour. We drove in multiple directions and this value seems to be pretty consistent.

Once the edge of digital coverage had been reached the analog performance had begun to degrade, with a noticeable increase in background noise. More importantly, driving along secondary roads with their omnipresent power lines caused the analog signal to fade to noise at regular intervals.

While it was possible to stop the vehicle in locations that provided good signal quality it was not generally possible to drive further away and get consistent reception on the analog signal only.

If we accept my rule of thumb above that 2 mV is the limit of consistent analog reception, it would appear digital has extended the basic coverage of an AM station by a large margin. Even if we accept a more generous value of 1 mV for analog, the digital is still delivering more coverage to mobile listeners in our situation.

It also should be noted that with a better transmitter and optimized antenna system, reliable digital service contour could probably be extended a bit further. I'm guessing it might be possible to get digital out to about the 0.5 mV contour when everything is done exactly right and the intermodulation products are brought down by another 5–10 dB.

AUDIO QUALITY IMPROVEMENTS

As far as audio quality goes, the digital system is very close to the performance of FM analog radio. There is a dramatic change in the sound when it switches to digital. This signal quality is held as you drive around until you leave the region of reliable digital service, exhibiting none of the periodic dropouts and fades that accompany analog AM.

The system delivers as promised as far as audio quality goes. Much credit for this goes to the efficient and high-quality audio codec that is part of the HD system.

More coverage and dramatically improved audio — those are pretty powerful test results.

WHAT'S INSIDE?

But what about the other side of the equation: Does the system generate interference? I'll get to that in our next issue.

Until then, take a look at Mario Hieb's article on the design of a digital phasing system for AM directional stations. This is an interesting proposal and we hope to hear more about the results once a few of these are built.

We also feature an interview with Ellyn Sheffield in this issue. Last summer we published the results of one of her audio testing studies, which validated the use of low-bit-rate audio channels with HD multicasting. This concept has taken off rapidly and now many stations are already offering multicast channels. Her audio testing and radio insights as an outsider to our field — she comes from the academic world of cognitive psychology — make for interesting reading.

We also have computer security from Stephen Poole, observations on the new Boston Acoustics Recepter radio from our own Guy Wire and, of course, Barry Blesser continues to provide his thoughts on broadcasting in the Last Word, this time bringing chaos to radio.

I hope and expect our readers to continue to weigh in with their comments to us via letters or e-mail. The new, simpler address is *rwee@imaspub.com*. Let us know what you are thinking. ■

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Phasor

CONTINUED FROM PAGE 4

to approximately +/- 15 kHz in the case of the hybrid mode and approximately +/- 10 kHz in the all-digital mode.

AM-IBOC directional array system issues

In conventional non-IBOC directional arrays, many of the aforementioned transmission system performance issues manifest themselves in audio degradation. The proposed AM-IBOC system makes more efficient use of the available spectrum. Unfortunately, directional array performance issues may cause degradation of the digitally modulated AM-IBOC signal.

Digital modulation, successfully employed in higher frequency bands, finds a challenge in the complex reactive networks of phasors and ATUs, and less-than-ideal characteristics of mutually coupled antennas.

According to Ben Dawson, P.E. of Hatfield and Dawson, Consulting Engineers, a recent ascertainment study of 53 CPB AM stations shows that 45 percent of the transmitter loads do not meet AM-IBOC VSWR standards. Problems include phasor and antenna system distortions, and lobe and null bandpass issues.

Many of the problems with AM-IBOC have to do with maintenance, but some AM systems have inherent characteristics that make it difficult for them to pass IBOC digital modulation. These characteristics include phasors, ATUs, antennas and combining networks.

Reactive components

Phasors are made mainly of reactive networks, namely inductors and capacitors. Although they are linear devices, these components store electrical energy.

In an inductor, energy is stored in the magnetic field; the time-domain voltage and current are expressed by:

$$V = L\,di/dt$$

In a capacitor, energy is stored in an electric field; the time-domain voltage and current are expressed by:

$$I = C\,dv/dt$$

Illustrated by the di/dt and dv/dt terms of these equations, the energy in these components vary with time and hence, frequency.

In the frequency domain, the impedance of a capacitor is:

$$X_C = 1/2\pi fC$$

And the impedance of an inductor is:

$$X_L = 2\pi fL$$

Reactive components can be used to create filters with attenuation vs. frequency (bandwidth) characteristics.

Impedance bandwidth

AM antenna system performance is important in that the input impedance must be sufficiently flat and symmetrical above and below the carrier frequency. This allows the final amplifier of the transmitter to "see" a load to which it can deliver the digital sidebands in their proper relationships without interference from the analog signal.

With good input impedance bandwidth, a high-quality hybrid IBOC signal can be put into an antenna system. The impedance bandwidth issue will be a major factor for a larger number of AM directional antenna systems than will be the case for non-directional antennas.

Pattern bandwidth

The next important performance criterion is pattern bandwidth. Because the IBOC system utilizes amplitude and phase modulation, the magnitude response and the delay characteristics of the transmitter-output to far-field process are important.

Regarding pattern bandwidth, directional antennas present a whole new level of complexity. The DA (directional antenna) pattern shape depends on characteristics such as transmission line length, tower spacing and tower height, which change in terms of the wavelength with frequency, as well as the reactance of the inductors and capacitors in the phasing and coupling system, which also change with frequency.

DAs that provide protective nulls at the carrier frequency may allow substantial energy to be transmitted at the sideband frequencies in an AM-IBOC scenario.

Solving pattern bandwidth problems involves redesigning the complete phasing and coupling system, meaning the networks within a phasor cabinet and at the tower bases will have to be extensively modified to an engineering-intensive custom design.

Although the prospects for providing acceptable hybrid IBOC service through directional antenna systems appear to be good, the corrective actions for those having problems that cannot be solved with simple RF networks at their inputs might be considerably more costly than will be the case for non-directional antennas.

There will be some cases where it will be necessary to construct replacement directional antenna systems using a combination of more land, more towers and taller towers before acceptable performance can be achieved.

Transmitter linearity

A low transmitter noise floor is important to digital transmission. The main factors affecting this noise floor are incidental phase modulation (IPM) and intermodulation distortion; both are a function of transmitter linearity.

System VSWR

Why is system VSWR more critical for AM-IBOC, and less so for traditional AM?

In traditional AM, signal information is contained in the amplitude of the sidebands. The information-bandwidth efficiency is somewhat low, and has nearly complete information redundancy from sideband to sideband.

In frequency and phase modulation systems, poor matching between transmitter stages and poor power amplifier to antenna matching (high VSWR) can cause non-symmetrical group delay, which can cause signal distortions and degraded reception.

In AM-IBOC digital modulation, with both amplitude and phase modulation going on at the same time, there is relatively high information-bandwidth with digital modulation; hence its higher spectral efficiency. Any transmission errors caused by linearity, loads, reflections, etc., will occur over a relatively narrow information bandwidth and the bit-error rates will be more severe.

Nulls and lobes

Many directional patterns have very deep nulls, and narrow lobes that challenge the performance of the directional system. Generally, nulls exhibit notch filtering and lobes exhibit bandpass filtering at the carrier frequency. Null areas have always suffered from an attenuation of sideband energy known as "null-talk."

Unfortunately, AM-IBOC systems could suffer narrow bandpass filtering, which would filter the digital sidebands of the hybrid IBOC system; these would occur on certain pattern lobes, the lobes usually being directed towards areas of significant population.

The system reduces the number of reactive components in the phasing and matching networks, thus simplifying, and in some cases, eliminating the networks that affect bandwidth, VSWR, IPM and group delay characteristics.

Antenna bandwidth

Many, if not most, of the non-directional antennas in use today will be able to perform well with hybrid-mode AM-IBOC. Most of those that will not will require only the installation of a new phase-rotation network between the transmitter and the antenna tuning unit to improve the symmetry of the sideboard load impedances at the transmitter's final amplifier.

The exceptions would mainly be shunt-fed towers that happen to have been built with high-quality wire skirts, and towers that are fed through filters that have a non-symmetrical effect on the input impedance, such as are found at some diplexed sites. Modifications beyond the installation of a simple RF network, such as changing the skirt feed arrangement or

SEE PHASOR, PAGE 8

Phasor

CONTINUED FROM PAGE 6

installing new filters, may be required in such cases.

Mutual coupling

The impedance of each antenna in a directional array changes constantly as the phasor is tuned. Because of this, the source and load impedance that the transmission line sees also change. Once a phasor is adjusted for proper phase and magnitude, it should then be re-adjusted for proper line matching; but this is difficult and is rarely done. For these reasons, it is often difficult to achieve good transmission line matching and VSWR performance.

Precorrection

Several transmitter manufacturers have developed AM transmitters that employ a pre-distortion or pre-correction of the transmitter output signal that is a converse of the load response. Although these systems should work well with non-directional systems, it seems that in directional arrays the pre-distortion systems will equalize mainly the phasor common point, and not the individual loads driven by the phasor output.

Additionally, the point of pre-distortion reference, whether it's along the transmission chain, the near-field or the far-field, will affect performance.

Negative towers

Antennas that have negative impedance actually absorb energy through mutual coupling. This energy then returns to the phasor through the respective output branch, where it combines with currents from the common point.

If the tower has positive impedance, net power is radiated and all is well. In a negative tower, more power is absorbed and then either feeds a dummy load or is fed back into the phasor. Neither of these scenarios is ideal. Burning up absorbed power in a dummy load lowers both array and system efficiency. Feeding absorbed power back to the phasor isn't that great of an idea either.

The returning power is delayed by an amount determined by the distance from the phasor to the array and the characteristics of the transmission line. This "dirty power" is nearly certain to be out-of-phase in reference to the phasor current. When added back at the phasor-summing node, it will reduce phasor power by an amount determined by the phase and amplitude of the signal.

Ideally, this power combines in phase creating a net contribution of power. For this to happen, the transmission line must be of a particular, precise length. Unfortunately, in this scenario if the carrier is in phase, the sidebands will not be completely in phase.

Summary

AM-IBOC requires high performance, not just from the transmitter, but also from the load and any intermediary networks. AM directional transmission systems that were adequate for analog AM may be inadequate for AM-IBOC.

It should be noted that work involving AM-IBOC and directional antenna arrays is ongoing and the theoretical and practical aspects of these systems will continue to evolve as this work continues.

III. NEW TECHNIQUES IN DIRECTIONAL SYSTEMS AND AM-IBOC

One way of developing a modern directional antenna system for AM-IBOC is to determine what the ideal system would look like. Because AM-IBOC employs digital modulation, a hybrid of amplitude and phase modulation, the ideal system would have the performance characteristics of the best AM and PM systems. It should have the following RF characteristics throughout the transmission chain:

- Low Incidental Phase Modulation (IPM)
- Symmetrical Group Delay
- Good bandwidth and low VSWR
- Good impedance matching of each transmission line
- Load Hermitian symmetry to +/- 15 kHz

The effects of not meeting these criteria include: less than optimum signal coverage due to impedance bandwidth issues; interference caused by pattern bandwidth issues; crosstalk from the digital sidebands into the analog band (hybrid mode) due to IPM and group delay issues; crosstalk from the analog signal into the digital sidebands due to IPM and group delay issues; and damage to transmission lines from voltage reflections.

I believe it is difficult to design, construct and maintain a conventional, passive phasor with ideal characteristics. The conventional phasor is a complex, multi-nodal, reactive network with component interaction.

Computer-modeled solutions require the equivalent circuit of the transmitter power amplifier to be known. Adjustment of any component in the network will affect the phase and magnitude of every branch current in and out of the network.

With the aforementioned drawbacks of passive phasors, and the performance requirements of the IBOC system, new techniques for creating directional patterns are being considered.

An ideal AM-IBOC directional system would have the following characteristics:

- Branch currents are completely isolated from one another
- ATUs provide 50 + j0 ohm loads at +/- 15 kHz; or if not, isolated dynamic pre-correction on each branch
- Stable directional array parameters over time
- No negative impedance towers
- Wide range of phase and ratio adjustment
- Excellent impedance and pattern bandwidth
- Ability to adjust for seasonal and environmental antenna system variations
- Economical
- Low-maintenance

Today an ideal AM-IBOC directional system may not be completely possible due to factors such as mutual coupling and antenna geometry. But there may be some ways to improve AM-IBOC DA performance over conventional directional systems.

The Digital Phasor

As mentioned, a different approach to feeding power to directional arrays may be required to exploit the full potential of AM-IBOC. Enter the Digital Phasor, an AM directional antenna transmission and control system.

Power division and phase shift of the RF signal are the primary functions of the phasor. The Digital Phasor utilizes different methods and systems to create RF phase shift and power division.

The Digital Phasor is really a control system, and it comprises a Virtual Power Divider; Digital RF Signal Delay; and Phasor Control Unit.

The primary advantage of this system is that it reduces the number of reactive components in the phasing and matching networks, thus simplifying, and in some cases, eliminating the networks that affect bandwidth, VSWR, IPM and group delay characteristics. A secondary advantage is that, in some configurations, the Digital Phasor is more economical.

A second advantage is that, in some configurations, the Digital Phasor is more economical.

This system could be part of a "black box" that modifies an existing solid-state AM transmitter, or part of a new transmitter topology. In this discussion, the term "RF signal" is defined as the modulated carrier.

Virtual power divider

A traditional phasor uses a network of passive reactive components to divide common point current. Variable inductors, through a system of mechanical linkages, allow for manual adjustment of the divider.

The Digital Phasor creates power division virtually, meaning no common point exists and no circuitry divides the common point current. Instead, banks of solid-state power amplifier modules generate tower current. The number of power amplifier banks equals the number of towers in the array.

Fig. 4 shows the topology of one possible embodiment. This example utilizes "n" transmitters, shown as the shaded blocks, where n is the number of towers in the array. A modulator drives each PA bank with the output power controlled by a Phasor Control Unit (PCU). Measurement of absolute output power of each PA bank is

SEE PHASOR, PAGE 10

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Phasor

CONTINUED FROM PAGE 8

necessary; current ratios are calculated and controlled by the PCU. The tower sees the same currents it would with a traditional phasor.

Digital RF signal delay

A traditional phasor uses a network of reactive components — usually inductors — to create phase shift or signal delay. Variable inductors, through a system of mechanical linkages, allow for manual adjustment of the delay. The phase shift is created at a high power level.

The Digital Phasor creates RF signal delay by digitally delaying the carrier signal and program audio at a low power level prior to modulation. A DSP or digital delay unit is utilized to create the delays in the digital domain; the number of delay stages is equal to the number of towers. The PCU controls the amount of delay of each stage. Fig. 5 shows the topology of such a system.

RF signal delay is calculated using the following parameters and terms:

$T = 1/f$

Where:
f = oscillator frequency in cycles/second
T = period of 1 cycle in seconds

The general equation for calculating time-delay per degree of phase is:

$T_{\phi 1} = 1/360f$

Where:
$T_{\phi 1}$ = one degree of phase in seconds
f = carrier frequency in Hertz

EXAMPLE: Assume a carrier frequency of 540 kHz. Find the period of one degree of phase.

$$T = 1/f = \frac{1}{540 \times 10^3 \text{ c/s}} = 1.8519 \times 10^{-6} \text{ s/c}$$

$$T_{\phi 1} = \text{one degree of phase in seconds} = T/360 = \frac{1.8519 \times 10^{-6}}{360}$$

$$= 5.1440 \times 10^{-9} \text{ sec/deg at 540 kHz}$$

Phasor Control Unit (PCU)

Both the virtual power divider and the RF signal delay stages are controlled by a PCU. The antenna monitor is linked to the PCU, providing feedback of antenna phase and current. Absolute power samples are taken from each PA bank and fed to the PCU where virtual common point current is calculated.

The PCU adjusts the amount of RF signal delay and PA power against an internal table of system parameters. This gives the Digital Phasor the ability to keep DA power ratios and phase parameters within tight specifications. When the array drifts out of tolerance, the PCU adjusts signal delay and power output to bring it back in.

Each PA bank will have a maximum power output equal to the highest power level required by the related tower. PA banks could be located in a cabinet or building at the base of each tower, eliminating the high-power transmission line. Individual PA banks, signal delays and the PCU could communicate with each other via wired or wireless data networks.

Standard user interfaces such as monitor, keyboard and modem or network would be considered part of the PCU. The Digital Phasor could be a "black box" that modifies an existing AM transmitter, or it could be part of a new transmitter topology.

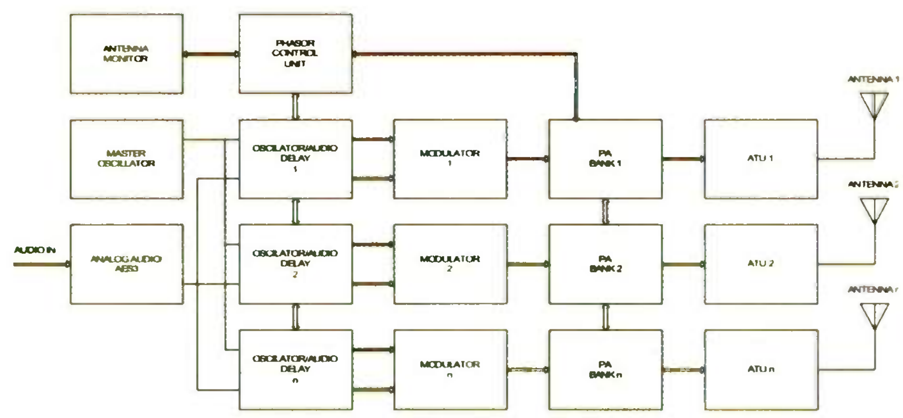

Figure 4. Virtual Power Divider Section of Digital Phasor (shaded)

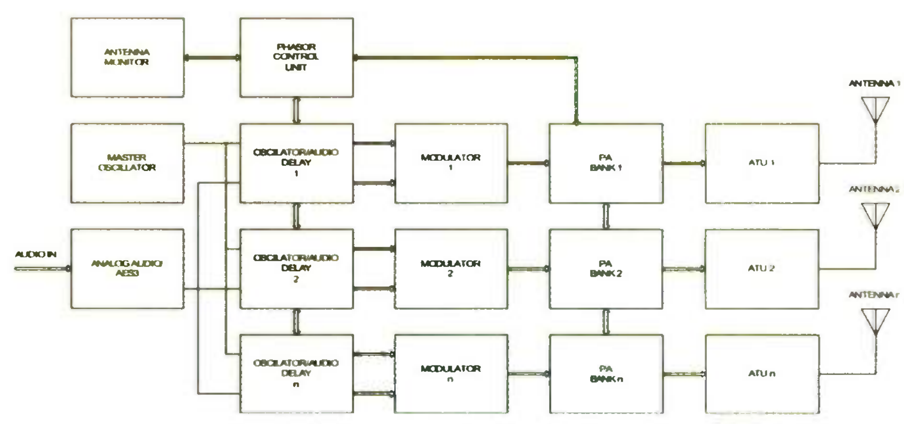

Figure 5. RF Envelope Delay Section of Digital Phasor (shaded)

Power at the tower

Another advantage of the Digital Phasor is the option of putting an amplifier bank at each tower. When this is done, the transmission line and ATU go away, and the transmitter drives the antenna directly. This greatly simplifies the transmission system in that impedance matching of the transmission line is no longer necessary, thus improving the bandwidth, VSWR and performance of the transmission system.

AM-IBOC and the Digital Phasor

Simply put, the Digital Phasor controls but does not store energy, a significant drawback of the traditional phasor. The theoretical advantages of the Digital Phasor over traditional phasors include better impedance bandwidth, better pattern bandwidth, better transmission line matching and lower branch VSWR due to no reactive phasor components.

Also predicted are better signal coverage in the AM-IBOC mode and better audio quality in the analog mode, due to better impedance and pattern bandwidth, lower IPM and group delay. True load pre-distortion per branch is possible and the system is self-adjusting with a feedback link to the antenna monitor. Multiple patterns are possible with marginal increase in cost and, in many cases, the Digital Phasor should be more economical.

At the present time, a Digital Phasor prototype has been tested on a two-tower, analog AM station at low power levels. Several of the aforementioned theoretical benefits have already been confirmed. The next-generation prototype is under development and will soon be ready for testing. More tests are planned at full power levels using off-the-shelf transmitters and an AM-IBOC exciter.

Comment on this or any story in RW Engineering Extra by writing us at rwee@imaspub.com.

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THE
NEWS

by James G. Withers

When Emergency Strikes, File for an STA

Jim Withers is owner of KSIX(AM) in Corpus Christi, Texas.

I doubt any broadcast station owner would suggest the FCC Rules and Regulations are there to make their life easier. But Section 73.1635, which deals with the Special Temporary Authorization, is one exception.

The purpose of an STA is to allow a station to operate under conditions that deviate in some significant way from its license. As the term suggests, these authorizations are temporary and exist to deal with "special" circumstances that make it impossible for the licensee to operate in accordance with its normal authorization.

Broadcasters all along the Gulf Coast became intimately familiar with STA operation during and after the hurricanes that last year, but emergencies can crop up at any time, and unlike hurricanes, seldom give advance warning. So it makes sense to know what the rule says and what it is for, in advance of actually needing to file for an STA.

The underlying reason for the STA section of the rules is simple: the commission wants your station on the air, even if circumstances dictate a deviation from the license. That does not mean, however, that the FCC just rubber-stamps STA requests. In fact, there are some specific requirements in place to prevent unwarranted STA operation.

If your request meets those requirements and your justification has merit, the commission, in my experience, is rather liberal in granting an STA. This even applies to occasions when a licensee must, with a reasonable explanation, take the station "dark." In such a case, the commission may grant Silence Authority, which it does under the aegis of the STA Rule.

The most obvious reason for filing an STA request is because of a problem at the transmitter site; for some reason, a station simply cannot operate from its authorized site, or with its authorized power/antenna. This was true for many stations affected by the hurricanes. Sites were flooded, transmitters damaged,

power was out and antennas were damaged by lightning.

The STA Rule is clear in these circumstances. The problem was clearly beyond the control of the licensee. Likewise, the problem is temporary — so far, so good. However, an STA request also must specify operation that would continue to serve the population of the station's community of license, and conversely would not extend the station's coverage beyond its normal range.

So, if your request for STA proposes operation to cover your nearby major metropolitan area while leaving the bedroom community (which just happens to be your city of license) totally bereft of signal, save your postage; a grant is unlikely.

Once you have determined the need, the FCC makes it exceedingly easy to file for STA, and it is now possible to file electronically, although a paper letter also is fine. I file the old-fashioned way, as that allows you to add pictures, if needed, and to expand on your justification as much as you want.

There is no absolute timetable for filing an STA, but sooner is better than later. Mainly, this is the case because as a licensee, you cannot assume a grant of STA will be forthcoming, and so, may not operate outside the parameters of your normal license until you have specific authorization to do so.

In the case of a general emergency, such as the hurricanes, I doubt the FCC would drop the hammer on a station for coming back on at low power so as to keep as many people as possible informed of conditions. But the rules are clear: You are not allowed to deviate from your station license unless and until the commission has granted the STA.

FCC WANTS TO HELP

The good news is, though, when warranted, the feds will jump through hoops to keep your station up.

A station I owned had its antenna on a leased tower that blew over in a particularly vicious storm. We were in the process of moving to a new site when it happened, so we rigged a quick emergency antenna on the tower at the new site.

I ran the numbers and faxed a two-page STA application to the Audio Division: (a) Here's the problem; (b) Here's our proposed solution; (c) Here are the temporary predicted contours. It was approved and faxed back the same day. Total time off the air: less than 24 hours. The statement "Hi, I'm from the FCC and I'm here to help you," is not always the punchline of a joke.

With regard to justifying your request: Do it well, but don't overdo it. That is to say, don't make things up. If your 40-year-old transmitter finally gave up, but you don't have the money to replace it until next month, do not tell the FCC that you have ordered one, but it cannot be shipped. Rather, tell them the truth: the transmitter failed and you need silence authority until you can replace it. It's much simpler and avoids that "Lack of Candor" thing they get so upset about.

One note of extreme caution, though, with regard to being silent. A few years back, an audit of silent stations showed an alarming number of stations that had been dark for years. It kind of seemed to some folks like certain licensees were simply "place-holding" the frequency until they could sell (shocking, I know, but there you are). Not much "public interest, convenience and necessity" performed by those stations, so Congress mandated an absolute one-year silent rule.

If your station is dark for one year and a day, I hope you have retirement plans, because you will no longer be licensee of that particular station. There might be a tiny bit of wiggle room, as in a case of "circumstances clearly beyond the control of the licensee," but I personally would not risk it. If I have to hand-crank a generator and rig up a coat hanger antenna, I'm on the air no later than 364 days after I went dark.

Provide as much documentation as you think the FCC will need to judge your request fairly, and do not be shy about calling and asking for the status of your request. I have done this and can report that they do not bite.

So when the stuff hits the fan, just remember, the FCC truly wants your station to be on the air and will respond favorably to almost any reasonable request. Provide as much documentation as you think they will need in order to judge your request fairly, and do not be shy about calling them and asking for the status of your request. I have done this and can report that they do not bite.

In the end, the litmus test is how your request for STA operation will benefit the public. If you are doing your best to remain on the air (or get back on, if you are requesting Silence Authority), and serve your community of license, and can demonstrate that the STA will help you accomplish that, chances are the FCC will grant your application.

Comment on this or any story by emailing to rwee@imaspub.com. ■

Application Note

Getting the most out of your digital console router system

Behind all the sizzle and flash of your new audio console lies the true power of your digital console router system…the router! Understanding some basic router concepts will help you find new ways to use your router system that you may never have realized were possible.

At its heart, a router system can be thought of as a virtual patch-bay, but with some additional capabilities. Let's take a quick look at a traditional, non-router based installation.

We're all used to having racks and racks of distribution amplifiers to get signals around the facility. A digital router system eliminates the need for D.A.'s—completely—by providing the ability to route ("patch") any source to one or many outputs internally.

Fig. 1:

Traditional use of D.A.'s to distribute mulitple sources to multiple destinations.

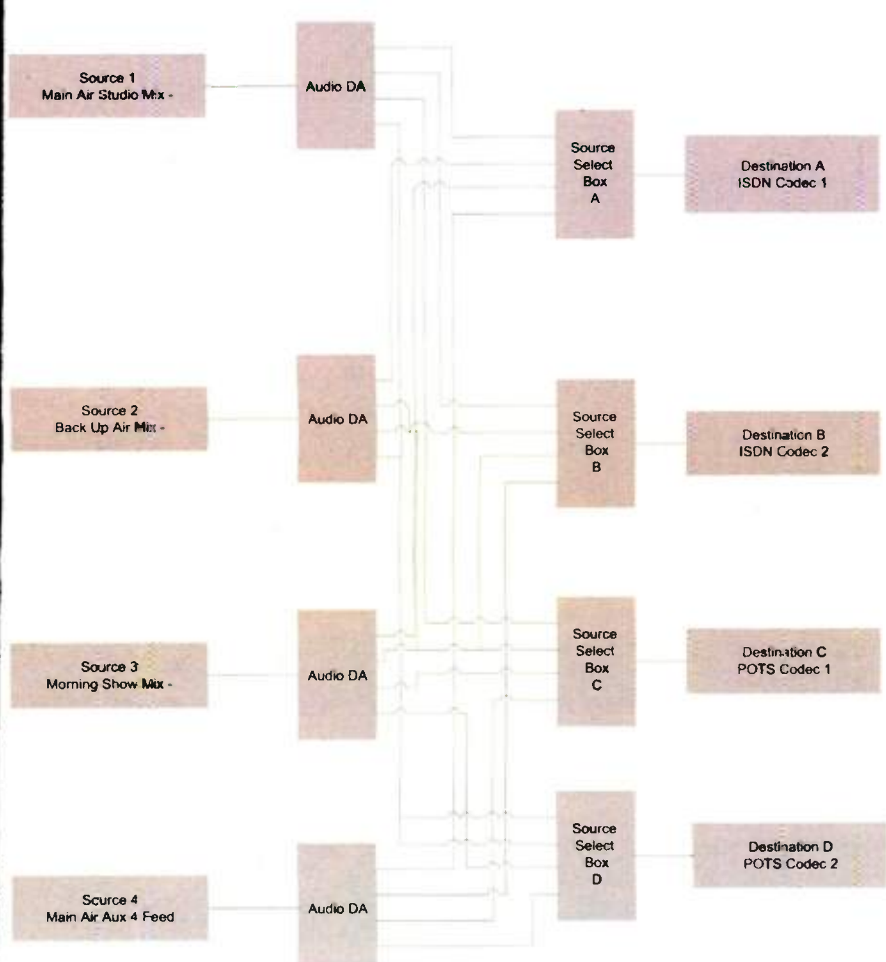
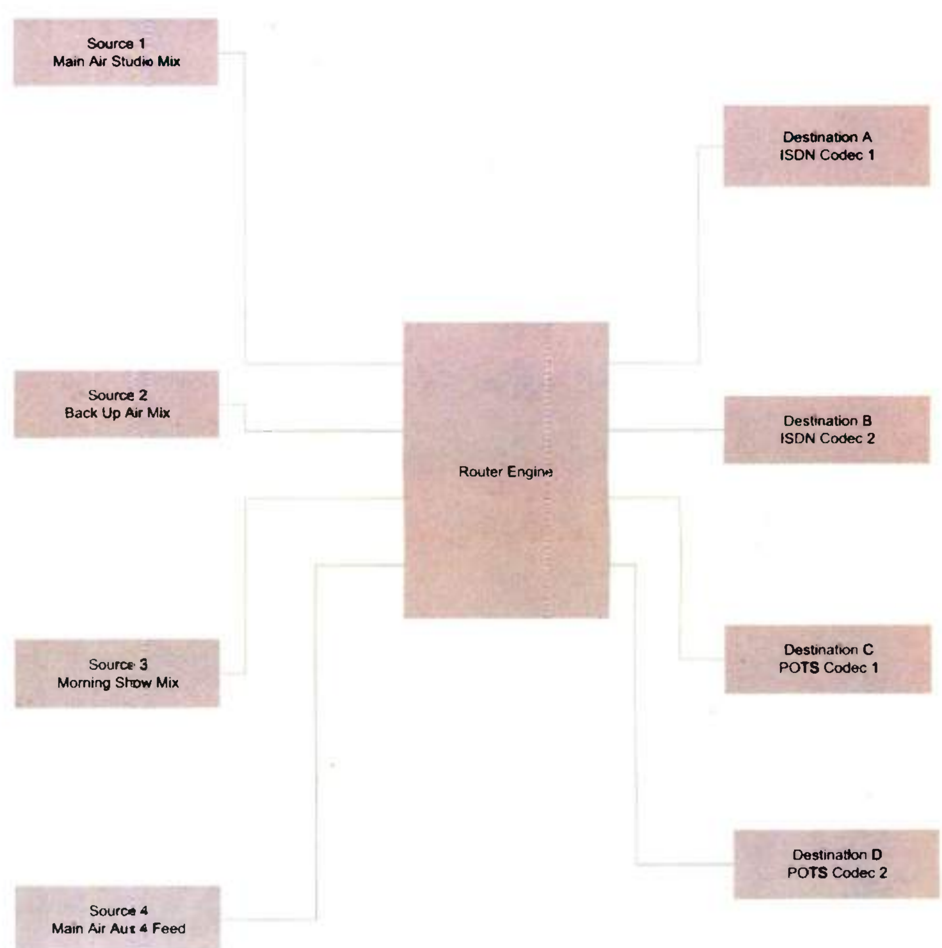


Fig. 2:

Router based distribution of multiple sources to multiple destinations.



While this capability may seem trivial at first glance, it belies the true power of your digital router system. By exploiting this capability you can, for example, build complex intercoms between studios or route your newsroom output on to all your stations with a single button press.

For a more detailed explanation of router concepts please visit www.logitekaudio.com/support.html and download our "Router 101" paper.

Do you have an interesting application or challenge that was resolved using Logitek equipment? We'd love to hear from you.

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by Bob Band

A Packetized Look at Audio Over IP

Exploring Two Approaches of IP Audio Transport and The Challenges of Packet Audio Distribution

Bob Band is the channel manager for Harris Networking Solutions.

Today's radio broadcasters are dependent upon their contribution and distribution networks to deliver broadcasts to listeners.

Contribution networks bring audio sources from multiple locations into one or more studios. Distribution networks are typically point-to-multipoint, and carry program audio to remote transmitters and translators. These networks handle the signals that ultimately feed the transmitters; as such, they must reliably and seamlessly pass high-fidelity audio for transmission. They also allow for remote control, signal monitoring and data transmission.

To date, fixed-bandwidth digital services such as T1, E1 and X.21 have been the systems of choice, as they provide reliable communications over dedicated circuits. With the advent of packet networking, and in particular high-speed Internet Protocol, the need to use expensive fixed-bandwidth services is being challenged.

IP services offer scalable bandwidth, are widely accepted and available and are considerably less expensive than dedicated, fixed-bandwidth lines. However, meeting the mission-critical needs of broadcasters can be a challenge. The reasons have to do with the way packet networks operate.

FIXED-BANDWIDTH VS. PACKET NETWORKS

In a fixed-bandwidth network such as T1 running at 1.5 Mbps, you get a "nailed-up" circuit dedicated to your use alone, carrying a constant stream of information. Picture a conveyor belt moving non-stop, onto which you can place your user data (audio programs or other); the belt keeps moving at a steady rate, whether you've put anything onto it or not.

The reliability on such a circuit is extremely high, but it comes at a price. You're paying for the full 1.5 Mbps of transport, 24 hours a day, 365 days a year, regardless of how much or how little you actually use. Also, such circuits are essentially point-to-point, which means you may need multiple T1s if, for example, you are distributing a program from one studio to multiple transmitters, or connecting several regional studios to a main production facility.

Packet networks such as IP, by contrast, are designed to be shared resources. Anyone with access to the network can put data onto it.

> **Packet loss can occur in several ways... if the jitter is so high that a packet arrives too late to be inserted back into its proper place in the jitter buffer before playout, then it is essentially lost.**

IP networks typically operate in one of two modes: Transmission Control Protocol or User Datagram Protocol. Essentially, TCP is designed for point-to-point bi-directional communication, while UDP is more suitable for unidirectional links, either point-to-point or point-to-multipoint broadcast.

Regardless of which mode, or protocol, is used, the information is broken up into small segments and placed into packets, a process knows as packetization. Along with user data, each packet contains several pieces of housekeeping data such as the destination address, a sequential packet ID and, when implemented, a mechanism for priority tagging.

For example, a packet containing bits of real-time audio could be tagged as urgent, while a packet containing bits from a data file transfer would be tagged as lower priority. Once the data is packetized, the packets are given to the network for delivery.

In the network, the packets are sent through to their destination via one or more switches and routers. Each device looks at each packet, makes a decision on the best route available at that moment in time, and sends the packet on its way. Gateways, which allow packets to move between networks using different protocols, can easily become sources of short-term congestion, as they must handle all packets moving between the networks involved.

For the audio to be successfully decoded at the far end, it is essential that all the packets reach their destination and they are played out in the same sequence as they originated. For this to happen, we have to address two inherent challenges presented by IP networks: jitter and packet loss.

Jitter is a condition where packets arrive at irregular intervals, or even out of sequence altogether. It occurs because there is no way to pre-determine the exact route any given packet will take through the network. As a packet reaches each switch, that device sends it on to the next one based on constantly-updated information about the congestion conditions on the various possible paths. Thus, sequential packets may be sent along different routes.

As the routes vary, so too will arrival times. Differences in arrival time translate into packet jitter, and as packet jitter increases, packets may begin to arrive out of order. To cope with this, a well-designed IP audio transport system should have a receive jitter buffer deep enough to hold the incoming audio packets until any delayed ones have arrived, so they can be played out in proper sequence.

Packet loss can occur in several ways. One relates to the foregoing discussion of jitter. If the jitter is so high that a packet arrives too late to be inserted back in its proper place in the jitter buffer before playout, then it is essentially lost.

Another is that IP networks are designed such that if a switch along the way becomes congested, packets can be dropped as needed to reduce the congestion.

There are several methods available to deal with packet loss. One is priority tagging, which allows the switch to differentiate between high- and low-priority packets. As congestion increases, low-priority packets are dropped first.

Second, it is important to have an adequate Service Level Agreement (SLA) with one's network provider. An SLA is a service contract in which the provider agrees to provide a certain level of guaranteed throughput across their network for your traffic.

A low-level SLA is fine for TCP traffic such as file transfers, where if a packet is missing, the receiver simply asks for it to be sent again. However, in real-time media transport, there is no tolerance for the system to wait while requesting a packet to be re-sent, so a higher-level SLA is required.

Despite everyone's best efforts, there may still be times when for one reason or another some packets do not arrive at their destination. One way to deal with this is through the use of Forward Error Correction.

FEC rearranges the transmitted bits and adds redundant data so that a missing packet can be reconstructed based on nearby packets. The amount of FEC needed depends on the conditions in the network — especially the last mile — and an analysis of traffic over time.

The right amount of FEC can take a reasonable network and make it quite robust. On the other hand, too much FEC is a waste of bandwidth, as increased FEC levels mean increased overhead. Typically, more FEC also means larger buffers, and larger buffers mean larger delays. An ideal system allows the user to adjust the amount of FEC on each stream to best suit the actual conditions.

Finally, there is the issue of unicast vs. multicast. Basically, a unicast IP stream is equivalent to a T1 point-to-point service; packets are sent from one address to another, intended for receipt by that location only.

A multicast stream, on the other hand, is made available to the network on a special IP address that allows an unlimited number of users to gain access to it. This is ideal, for example, for distribution of a program from a studio to multiple transmitter sites. It also is highly bandwidth-efficient; the multicast protocol allows the same program to be made available in multiple locations without the need to reduplicate the band-

SEE IP AUDIO, PAGE 16

JUST ENOUGH TEST

Is your bulky bench analyzer more test than you use and more weight than you want?

Sophisticated Minstruments from NTI give you just enough test capability, plus functions not even available on their larger siblings... and these flexible instruments fit in the palm of your hand

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The ML1 is a full function high performance audio analyzer and signal monitor that fits in the palm of your hand. The comprehensive feature set includes standard measurements of level, frequency and THD+N, but also VU+PPM meter mode, scope mode, a 1/3 octave analyzer and the ability to acquire, measure and display external sweeps of frequency response generated by the MR1 or other external generator.

With the addition of the optional MiniSPL measurement microphone, the ML1 also functions as a Sound Pressure Level Meter and 1/3 octave room and system analyzer. Add the optional MiniLINK USB computer interface and Windows-based software and you may store measurements, including sweeps, on the instrument for download to your PC, as well as send commands and display real time results to and from the analyzer.

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- THD+N and individual harmonic measurements k2→k5
- VU + PPM meter/monitor
- 1/3 octave spectrum analyzer
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With all the power and digital audio measurement functions of more expensive instruments, the DL1 analyzes and measures both the digital carrier signal (AES/EBU, SPDIF or ADAT) as well as the embedded audio. In addition, the DL1 functions as a smart monitor and meter for tracking down signals around the studio. Plugged into either an analog or digital signal line, it automatically detects and measures digital signals or informs if you are on an analog line. In addition to customary audio, carrier and status bit measurements, the DL1 also includes a sophisticated event logging capability.

- AES/EBU, SPDIF, ADAT signals
- 32k to 96k digital sample rates
- Measure digital carrier level, frequency
- Status/User bits
- Event logging
- Bit statistics
- VU + PPM level meter for the embedded audio
- Monitor DA converter and headphone/speaker amp

NEW! AL1 Acoustilyzer
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The AL1 Acoustilyzer is the newest member of the Minstruments family, featuring extensive acoustical measurement capabilities as well as core analog audio electrical measurements such as level, frequency and THD+N. With both true RTA and high resolution FFT capability, the AL1 also measures delay and reverberation times. With the optional STI-PA Speech Intelligibility function, rapid and convenient standardized "one-number" intelligibility measurements may be made on all types of sound systems, from venue sound reinforcement to regulated "life and safety" audio systems.

- Real Time Analyzer
- Reverb Time (RT60)
- High resolution FFT with zoom
- Optional STI-PA Speech Intelligibility function
- THD+N, RMS Level, Polarity

MR1 Minirator
Analog Audio Generator

The MR1 Minirator is the popular behind-the-scenes star of hundreds of live performances, remotes and broadcast feeds. The pocket-sized analog generator includes a comprehensive set of audio test signals, including sweep and polarity signals which work in conjunction with the ML1 Minilyzer.

- Sine and square waves
- Pink & white noise
- Polarity test signal
- Stepped sweep for response plots
- Balanced and unbalanced outputs

MiniSPL
Measurement Microphone

The precision MiniSPL measurement microphone (required for the AL1 Acoustilyzer and optional for the ML1 Minilyzer) is a precision reference mic for acoustics measurements, allowing dBSPL, spectrum and other acoustical measurements to be made directly.

- 1/2" precision measurement microphone
- Self powered with automatic on/off
- Omni-directional reference microphone for acoustical measurements
- Required for the Acoustilyzer; optional for the Minilyzer

MiniLink USB interface
and PC software

Add the MiniLINK USB interface and Windows software to any ML1 or DL1 analyzer to add both display and storage of measurement results to the PC and control from the PC. Individual measurements and sweeps are captured and stored on the instrument and may be uploaded to the PC. When connected to the PC the analyzer is powered via the USB interface to conserve battery power. Another feature of MiniLINK is instant online firmware updates and feature additions from the NTI web site via the USB interface and your internet-connected PC.

- USB interface fits any ML1 or DL1
- Powers analyzer via USB when connected
- Enables data storage in analyzer for later upload to PC
- Display real time measurements and plots on the PC
- Control the analyzer from the PC
- Firmware updates via PC
- MiniLINK USB interface is standard

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IP Audio

CONTINUED FROM PAGE 14

width more than once on any given link.

APPROACHES TO IP AUDIO TRANSPORT

Thus far there have been two basic approaches to IP audio transport: simple standalone audio codecs, and "homegrown" systems solutions, often involving a combination of software and hardware from multiple vendors. Stand-alone codecs do not scale easily as needs and technologies change, while homegrown solutions grow rapidly in complexity and present significant problems in the area of management and control.

Harris has taken a new approach to IP audio transport: the Intraplex NetXpress managed platform.

What, in this context, is a managed platform? It is an integrated combination of hardware and software that allows broadcasters to handle multiple audio programs, each with the appropriate type of audio compression (or none at all); integrate with existing T1/E1 services; carry legacy data circuits; and connect to PABX telephone systems.

It provides T1/E1 circuit emulation, and the ability to carry T1/E1 circuits intact over IP networks, thus allowing users to move in a gradual, controlled manner from legacy fixed-bandwidth services to the new world of IP — on the same managed platform.

GETTING IP RIGHT

As we discussed earlier, IP circuits present special challenges in transporting real-time program audio reliably. The right tools can help overcome these challenges:

Redundancy. Hot-standby power supplies and support for redundant networks, automatically switching traffic to a backup link (where provided) in case of primary network failure.

Packet size. Control of packet size is a key tool in network management. The optimum packet size for any given stream is a tradeoff between latency (delay) and overhead. Where low delay is critical, set the packet size to a minimum, and the packetization delay can be kept below 1 ms — but this will create more overhead. Conversely, where bandwidth is tight, use larger packets. This will incur more delay, but requires less overhead.

Jitter buffering and resequencing. As stated above, packets may not arrive at the receive end in the sequence they were sent. A deep, user-adjustable jitter buffer accommodates the real-world conditions encountered in IP networks, with the ability to retain and restore out-of-sequence packets.

FEC. Forward Error Correction allows the system to rebuild lost or dropped packets, restoring the original audio quality before playout.

Network statistics. Provide cumulative and current statistics on packets sent, received, lost and delayed on every stream, allowing you to see what's going on in your IP network and adapt to changing conditions.

Event logging. Alarms and failures, on both the network and hardware side, are recorded for maintenance and review.

Stream type. Every stream can be set up for one-way or duplex unicast, or for multicast, for maximum network efficiency.

Priority tagging. Support for multiple levels of prioritization, ensuring that real-time audio traffic moves through the network ahead of less-critical data.

Input policing. Provide transport of LAN traffic along with the audio channels, and police the input to limit the amount of bandwidth available to the LAN, ensuring it can never create undue congestion levels that might interfere with the audio packets.

Legacy telephone support. The ability to carry traffic from traditional telephone systems, linking PABXs and establishing Off-Premise Extensions without moving to VoIP telephony. To this end, it incorporates built-in echo cancellation and supports traditional telephone signaling.

MANAGEMENT AND CONTROL

A managed platform cannot be truly managed without an effective means of monitoring and control. Accordingly, the management interface on NetXpress is an element manager based on Simple Network Management Protocol, with access to all aspects of the hardware and network environment.

SNMP is the most widely accepted protocol in IP network management. It provides a means to monitor and control many different types of network devices, allowing the user to change configurations and collect performance statistics.

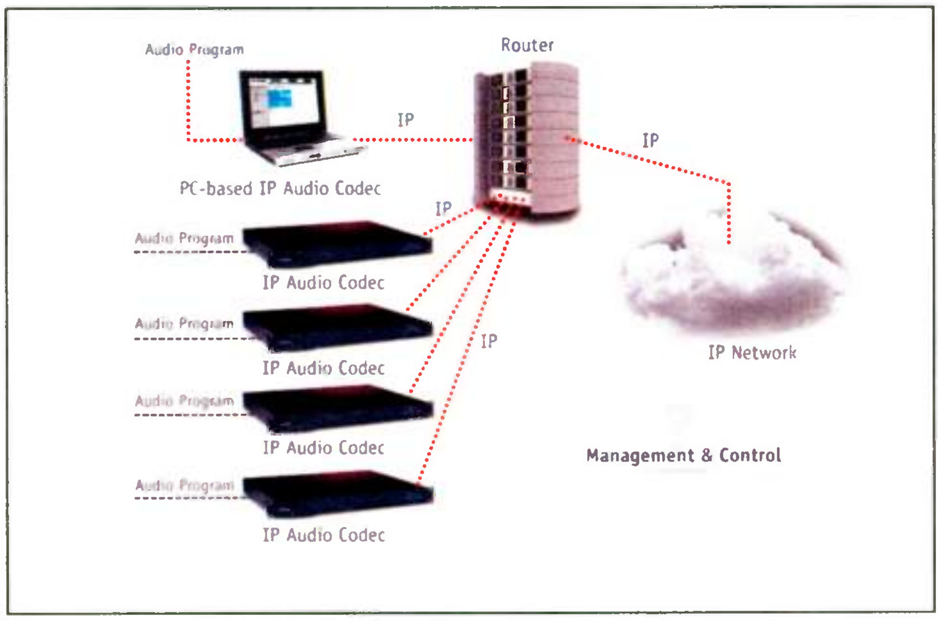
A big advantage of using SNMP is that it allows the system to communicate with higher-level network management software, such as Harris NetBoss or HP Openview, making it part of a complete operations management system. The basic architecture is that of client/server; the NetXpress acts as a client and maintains a Management Information Base, a database of information about itself and its internal processes, which it shares with the network manager.

NetXpress also contains a built-in Web screen generator, which means it can be viewed and operated via any Web browser, with appropriate security and password control.

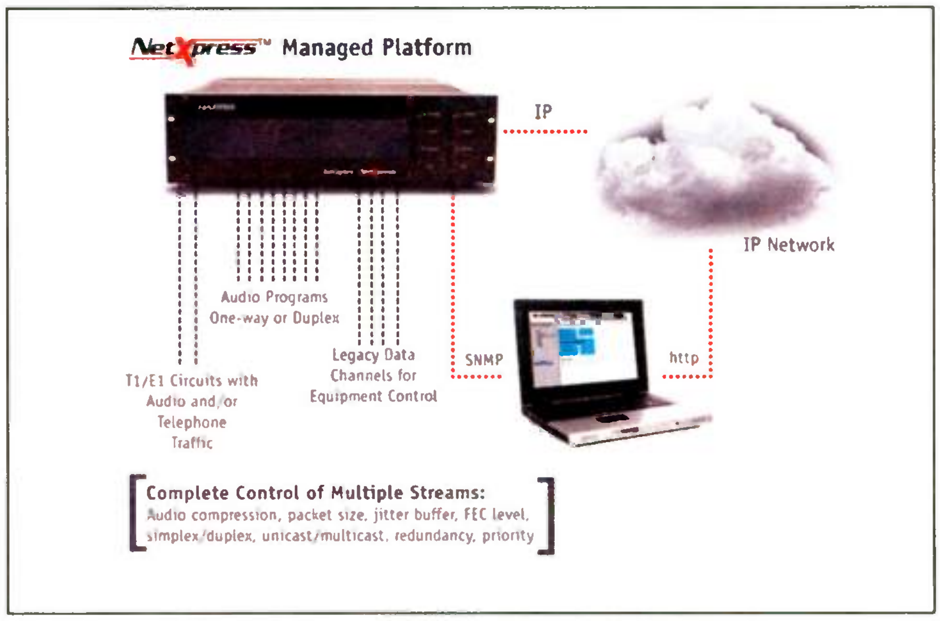
SUMMARY

IP networks offer the possibility of highly flexible, low-cost audio transport, and when properly managed can offer the degree of reliability required for use in contribution and distribution networks.

But there are many challenges to be overcome in ensuring this level of reliability. Radio broadcasters armed with a suite of software and hardware tools for redundancy, error mitigation, quality of service and network performance monitoring can more effectively tackle these challenges. ■



Traditional approach to transporting audio over IP networks



Managed platform approach to transporting audio over IP

The world's best-sounding POTS codec.

At Telos, we're obsessed with quality audio. We were the first to marry DSP with broadcast phone hybrids to achieve clean, clear caller audio. We invented Zephyr, Earth's most popular way to send CD-quality audio over ISDN. And now our DSP experts have built the **best-sounding POTS codec ever — Zephyr Xport.**

Instead of proprietary algorithms, we chose **MPEG-standard *aacPlus*®**, the same coding used by XM Satellite Radio, Digital Radio Mondiale, Minnesota Public Radio, Apple Computer and many others to deliver **superior audio at low bit rates.** (An optional ISDN interface lets Xport connect to Zephyr Xstream with Low-Delay MPEG AAC, or with nearly all third-party ISDN codecs using G.722.)

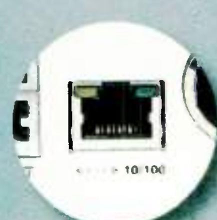
There's no need for a studio-side POTS line. Your studio's Zephyr Xstream receives Xport's POTS calls via its existing ISDN line, **eliminating the cost of a second POTS codec** and delivering smooth, clear digital audio to your listeners.

And Xport **makes unexpected modem re-training extinct** thanks to custom DSP algorithms that extract stable performance from even marginal phone lines. Xport gives you **surprisingly clean 15 kHz remote audio at bit rates as low as 19 kbps.**

No wonder clients tell us Zephyr Xport is the world's best-sounding POTS codec. But don't take their word for it — hear it for yourself.

Zephyr Xport: **It's all about the audio.**

Two-input mixer with sweetening by Omnia, switchable Phantom power, and send / receive headphone mix make life on the road easy.



Ethernet port isn't just for remote control: feed PCM audio right into the codec from any Windows™ laptop. Great for newsies on the go.

Xport's *aacPlus* and Low-Delay MPEG AAC deliver superb fidelity. G.722 coding enables connections with 3rd-party codecs, too.

Xport lets you easily send and receive audio using a cell phone headset jack. Gives a whole new meaning to the phrase "phoning it in."

Telos

AUDIO | NETWORKS

telos-systems.com

by Stephen M. Poole

How to Protect Broadcast Computers

We Worry About Viruses and Hackers Getting Into Our DAWs. Now We Have to Protect Our Transmitters Too

Stephen M. Poole is chief engineer for Crawford Broadcasting Corp. in Birmingham, Ala.

I had what the glitterati call an epiphany several weeks ago, when we were installing our first HD transmitter here in Birmingham.

"This thing," I said to no one in particular as I cast a gimlet eye on our BE FSi10 Digital Encoder, "ain't nothing but a glorified computer!" — an honest-to-goodness PC running GNU/Linux, complete with hard disc, USB ports, mouse and keyboard. It even had a CD-ROM drive.

People, I've never even contemplated having a need for a CD-ROM drive in a transmitter. When I was a young buck engineer first learning how to break Ohm's Law, if we ever even encountered a computer in a radio station, it would be a giant automation system with blinky lights and carousels and big reel-to-reel tape machines. Now transmitters come with CD-ROM drives, flat screen monitors and … network connections.

Don't miss that: a network connection. An Ethernet port. A way to communicate with the rest of the world via TCP/IP. Worst of all, a way for the rest of the world to communicate with that transmitter, if you're not careful!

This raises a whole new raft of unpleasant possibilities. We've already had to worry about worms, viruses and hackers getting into our audio workstations. Now we have to worry about that for transmitters, too.

Think about it: If someone attacks your Prophet system, that's bad enough. But if they find a way to hack into your HD equipment (and they will, just give them time), they could do anything from knocking you off the air to inserting profanity into your PAD data.

The inspiration for this screed, which in one of my usual bursts of incredible originality I shall call "Security 101," hit me when I was installing the importers for our HD systems. Ibiquity says in its Quick Setup guide, and I quote: you must be running Windows XP (whimper) with Microsoft's Internet Information Server (IIS — another whimper!) and — drum roll, please — "if you have installed Service Pack 2," (and if you have a functioning brain, I hope you have), "you must disable the Windows XP firewall."

Have we not learned anything over the past few years, what with one worm after another virus hammering Windows 2000 and XP machines right and left? This is worse than inexplicable, it's inexcusable. Now, to be fair, the units we received did have the firewall set up correctly, but that's what the documentation said. At best, it would mislead the user into thinking that the firewall is unnecessary; at worst, it might cause someone who installs (or reinstalls later on) the software themselves to expose a completely unprotected Windows XP machine to the world at large.

Therefore, I, your humble writer, am going to address this. Most discussions of this type jump straight into a discussion of IP Addresses, Ports, NAT and firewalls, but actually, for us, the very first thing to grasp is, what's a server?

WHAT'S A SERVER?

When we hear the word "server," we naturally think of a file server — a big, honkin' machine running Novell, Linux or Windows 2003 with a bunch of hard drives in a RAID array. But in fact, the term "server" in computer usage simply means, "a software program that provides a service that responds to client requests."

While the clients and servers that concern us here are those that work over a network connection, that's not even required. It's possible for a single PC to be running clients and servers that talk to each other on that same machine.

To repeat: A "server" is something that responds to a request and a "client" is the one who makes the request. The classic example is someone using a Web browser to fetch pages from a Web server. The browser is the client; he asks for a page, and the Web server responds by returning it (or a "404 — Not Found" page if the client asks for something that isn't on that server).

Most people don't realize that every single Windows machine that shares files on a peer-to-peer network is running a little mini-server. If Tom wants to copy a file from your computer, a client in his machine will contact the server in yours, make the request and then wait for a response. The little server in your PC will decide if Tom has permission; if so, it will stream that file's contents over the network back to Tom's client, which will copy it to Tom's disk drive.

That's a security risk to start with; you should never enable file and printer sharing on a Windows machine unless you really need to. But it gets worse. Over the years, a number of services/servers have been developed that will allow someone to run code remotely over a network. There are ways to do it safely (for example Java and JavaScript), but the truth is that the way that many services under Windows have done it in the past has been anything but safe.

In fact, if you want a quick and dirty, less-than-technical explanation of why Windows XP got so badly hammered by worms and viruses a while back, it's simple; it was running a bunch of services that did a poor job of vetting and validating client requests. Microsoft is finally cleaning up its act, but you had still better have a good firewall in place if your critical network runs 2000 or XP machines.

I'd like to recommend that each Windows machine have a virus/worm scanner in place, too. The problem is, so many of them are worthless at best and horrible at worst; they're buggy resource hogs that can cause more problems than the malware from which they're supposedly protecting you. Frankly, the top sellers, in my opinion, are both useless and expensive; there are scanners available on the Web that are better. But that's your choice.

That's protection at the individual machine level. Now for the network level, starting with a discussion of IP addresses and ports.

> **Any software on your computer that provides a service that can be used by a remote client is a 'server,' and it can become a security risk.**

IP ADDRESSES AND PORTS

You already know what IP addresses are: they're the four-byte (soon to be much larger) numbers that identify the machines on your network. Your computer might be at 192.168.1.10; Tom's laptop might be at 192.168.1.11 and so on. Every single machine that communicates on a TCP/IP network must have an IP address.

IP addresses can be compared to the street, city and Zip code on an envelope: they tell the network where your computer "is." Given that, ports are like the addressee's name. The street, city and Zip get the envelope to your house, but the name tells everyone who lives there just who should get the message.

Likewise, when you run a network server/service on your PC, it will listen to one or more ports at your IP address. When requests come in from the network, they will be addressed to your IP (the "address") and a port number (the "addressee"). The IP address gets the request into your computer, but the port number tells the network layer inside your PC precisely where that request should go next.

A single PC can have 65,000 (give or

SEE SECURITY 101, PAGE 20

Security 101

CONTINUED FROM PAGE 18

take a few hundred) different active ports open. Many port numbers are standardized: for example, the standard "http" Web server port is 80, the standard port numbers for e-mail send and receive are 25 and 110 and so on. You can find lists of these standardized port numbers all over the Web.

Other port numbers are specified by the vendor, and should tell you in the user's manual what port they'll be using or they'll let you choose a number. For example, if you want to communicate with a Telos 2x12 over your network, the little server inside that Telos uses port number 5001 by default.

OK? The critical point is this: when you run a network service/server, it will listen to certain ports. That's why the first step in securing a network is to simply kill any services that you really don't need — and the easiest way to do that is to simply uninstall any software that you don't absolutely have to have.

In some cases, though, you have no choice; you must enable some services that are inherently insecure, such as COM, the Component Object Model, under Windows. The only way to do this safely is to isolate the network. Simply put, all those machines that require these insecure services should be on their own separate network, with their own separate router/switch and separate cabling, and even (if possible) be in separate rooms with limited access. Ideally, these machines would never, ever even "see" the other machines in the building, and certainly wouldn't be able to access the Internet.

The problem is that this isn't always possible. For example, we want our Prophet NexGen audio system to provide song title and artist information for PAD display on HD receivers, so we must connect our HD system to the audio network. Our streaming computers, which are directly connected to the Internet, also need info from the Prophet network. Finally, we want to remotely control Prophet from the outside. We've now got multiple exposures, with more than one way for a hacker to get into our critical Prophet and HD networks from the world at large.

To address this, you need a firewall.

THE FIREWALL

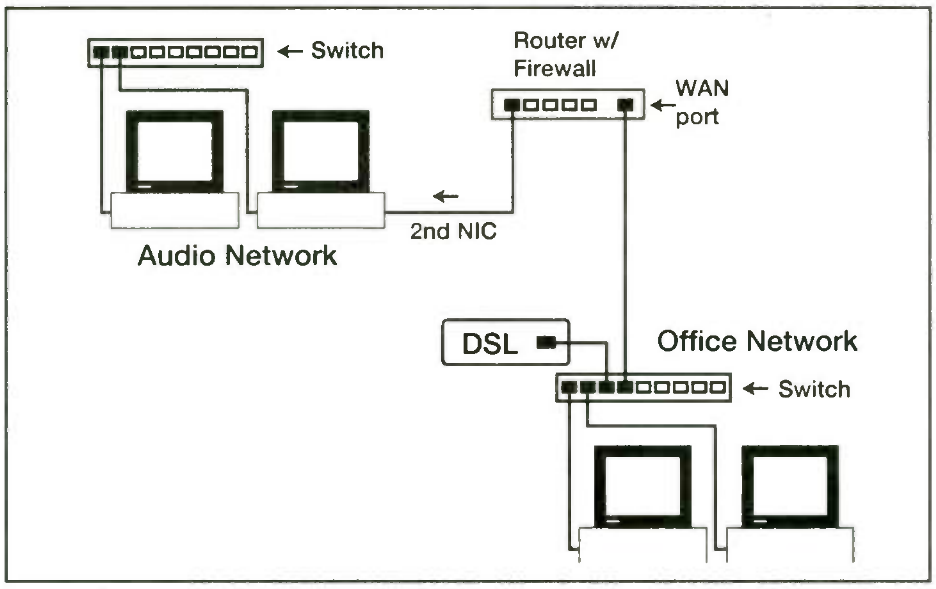
Entire books have been written about this, so I can only scratch a small rusty spot on the surface here. To keep this simple, a "firewall" is a device that will control access to certain ports and IP addresses. You typically put a firewall between any two sub-networks that need to be isolated.

I'm not going to show all of the possible interconnections. Let's just say we have an audio network and an office network with DSL access to the Internet. I want to remotely control the audio network over the Internet. We used PC Anywhere on a dialup for a while, but that was too slow. What to do?

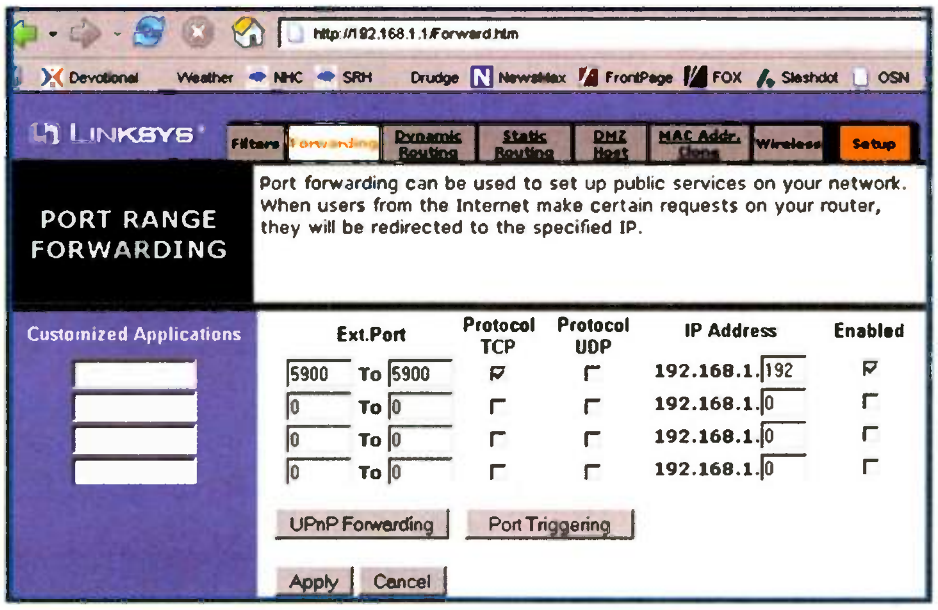
The program we chose, Ultr@VNC (*www.ultravnc.org*), only needs access to port 5900, so that part was easy; port 5900 would be the only port that we'd "open" to the Internet. We only needed to run the VNC server on one audio workstation, because Prophet NexGen will allow us to administer all five stations from that one PC.

There are many ways to do this, but we opted for the quick and inexpensive route. We inserted a LinkSys router/firewall between the two sub-networks and put an extra network interface card into one of the Prophet machines. The second NIC was assigned an IP address in the "office" range (192.168.1.xxx). Network cards are dirt-cheap nowadays, so why not? Quick and easy, and you get an extra layer of isolation to boot.

Routers like the LinkSys are available at any computer store such as CompUSA, and are normally used in homes and small offices to share an Internet connection. Typical of the breed, this one has a decent firewall and network address translation built in, along with a little built-in Web server that allows us to configure it with a browser.



Using a router to isolate a sub-network



An example using a Linksys router/firewall

Assuming you have everything connected as shown in the illustration, you might set it up this way:

- Connect the WAN jack on the router to your office network. Configure the router to ask for a dynamically assigned IP address from your office network's DHCP server. In our case, DHCP is built into the DSL router/modem.
- Now configure the router to block all incoming access by default. Refer to the documentation for the router.
- Finally, we use Network Address Port Translation to send anything coming in on port 5900 to the IP address of the card we added to the audio workstation.

That last one may be labeled "port forwarding" or "virtual server" on your router. Here's the actual screen in my Web browser for the LinkSys. I'm setting it up to forward anything from port 5900 to IP address 192.168.1.192, the address of the second network card I installed in the Prophet workstation.

That's the basic principle; you can easily take this and expand it as needed. Remember the two steps: isolate each sub-network, then interconnect with a firewall at one point only.

Basically, you learn to think in "clumps" or "clusters." The audio network is one cluster, the office network is another and the HD interconnections might be a third. Step one is to isolate each of them totally from the others. Once you have each isolated sub-network running perfectly, you then carefully "tap into it" at one, and only one, carefully firewalled point.

Using this basic approach, we have these little router/firewalls all over the place now. At any point that we want to interconnect with limited access, you'll find one of these boxes.

In a future article we'll take a detailed look at some of the services that might be running on your networks, with an eye to killing those you don't really need. We'll also play with some free software, available for download from the Internet, to further secure everything.

This article by Stephen M. Poole, CBRE, CBNT, is adapted from the Crawford Broadcasting Corp. engineering newsletter The Local Oscillator. ■

Solutions[6]

for your digital studio and program chain

INNOVATIVE PROBLEM SOLVING TOOLS FOR BROADCAST

ADCS III

The ADCS III provides professional quality 24-bit A/D conversion, along with AES path insertion and interruption functions. The ADCS III may be configured as a standalone A/D converter or to switch AES or stereo analog inputs into another AES stream. The ADCS III is a perfect companion for analog EAS encoders/decoders to be inserted into an AES program stream. The ADCS III is equipped with relay bypass in case power is removed from the unit. The A/D converter may be configured for sample rates of 44.1 or 48kHz (32kHz may be special ordered) or an external word clock from 32 to 96kHz. The ADCS III may be set on a desktop, mounted on a wall or up to three units may be mounted on the optional RA-1, Rack-Able mounting shelf.

CSD-1

The CSD-1 converts a composite stereo signal into discrete left and right balanced outputs. Features include; twin BNC input connectors; multi-turn input level control; twin power connectors allowing up to four units to be driven off of one power transformer; front panel output trimmers; front panel stereo and power LEDs and plug-in Euroblock output connectors. The CSD-1 is powered by a surge protected internal bi-polar 12vdc power supply affording superior headroom and high definition audio. The CSD-1 may be set on a desktop, mounted on a wall or up to four units may be mounted on the optional RA-1, Rack-Able mounting shelf.

DMS-III

The Broadcast Tools AES/EBU Digital Monitor & Switcher III is designed to accept and automatically or manually switch two AES/EBU signal sources when an AES digital error and/or analog silence are detected. Features include: Automatic control function that switches to a back up source upon failure of the main source; Switch functions can be triggered by loss of clock, AES digital error flags, front panel transfer switch, external switch contact and/or the internal analog stereo silence sensor. Additional features: Front panel error status and sample rate LED indicators; front panel headphone jack and level control; balanced stereo monitor output; remote control; removable screw terminals; Plug & Play installation; dipswitch selection of precise time delay from 2 seconds to 85 minutes and restore timing delay from off to 10.2 minutes; defeatable sonalert aural alarm; SPDT status relays; SPDT one-second pulse relay. The DMS III may be set on a desktop, mounted on a wall or as part of the new RA-1, Rack-Able mounting shelf.

RDDA 4x4

The RDDA 4x4 quad-mode AES/EBU digital distribution system provides up to sixteen outputs from a single digital source. The sixteen outputs may be split into four groups, providing four transformer-balanced outputs each from a single source.

The RDDA 4x4 is ideal for distributing AES/EBU signals of sample rates up to 96kHz and/or word clock around your facility without any signal alteration.

The RDDA 4x4 is equipped with four transformer balanced loop-thru inputs with selectable termination and sixteen transformer-balanced outputs with selectable termination. The half rack profile, allows the unit to be set on a desktop, mounted on a wall or as part of the new RA-1, Rack-Able mounting shelf.

ADCS III AES/EBU Analog to Digital Converter & Switcher

CSD-1 Composite Stereo Decoder

DMS III AES/EBU Digital Monitor Switcher III

RDDA 4x4 AES/EBU Digital Distribution System

DAS 8.4 AES/EBU Digital Audio Switcher

8x1 DAS Digital Audio Switcher

DAS 8.4

The Broadcast Tools DAS 8.4 provides AES/EBU digital audio routing/switching of any one of eight AES/EBU inputs to four AES/EBU outputs. The DAS 8.4 may be used in Multicast applications, where an AES enabled EAS device assigned to input eight is routed to all four outputs via a contact closure or serial command. When the alert/test is completed, the DAS 8.4 will return to its previous I/O configuration. The DAS 8.4 may be controlled via front panel switches, contact closures, 5-volt TTL/CMOS logic and/or the multi-drop RS-232 serial port. The front panel is equipped with input and output selection push buttons, output assignment LED's, sample rate LED indicators and a headphone jack with level control. Additional features: 96 KHz AES receiver, a 24-bit D/A converter with analog balanced stereo output, headphone amplifier and 16 x 16 GPIO port. Installation is simplified with plug-in euroblock screw terminals. 1-RU chassis.

8x1 DAS

The 8x1 AES/EBU Digital Audio Switcher routes any one of eight AES/EBU digital inputs to three AES/EBU outputs in a 1-RU space. Programmable or last selected source memory, safety lock out, output muting, remote control/status, multi-drop RS-232 and RS-485 serial ports. Two versions: balanced 110 ohm with XLR connectors, or 75 ohm unbalanced with BNC connectors.

*Be sure to visit our web site at **www.broadcasttools.com** for additional equipment and product manuals.*

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Sheffield

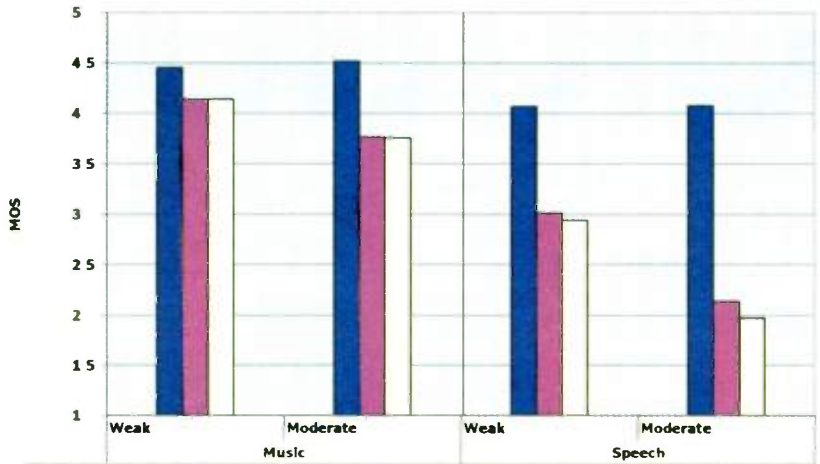
CONTINUED FROM PAGE 1

rushing to roll out "multicast" audio streams as part of their HD conversion, using bit rates that are comparable to Internet Web sites.

Dr. Ellyn G. Sheffield, Ph.D, helped lay the groundwork for the optimal use of digital audio compression and the selection of the 96 kbps IBOC data stream. She also conducted extensive consumer audio testing in conjunction with National Public Radio to determine if listeners could detect quality differences in the HD coder at low bit rates for multichannel uses and radio reading services.

Sheffield's research set the guidelines for data rates that IBOC stations can use and still maintain listener satisfaction. Her results are important for any station operating or planning to operate an IBOC digital carrier.

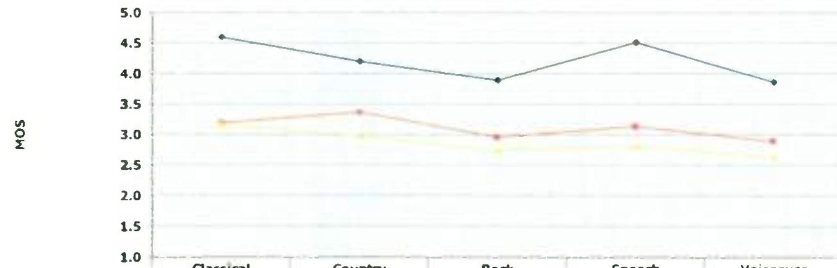
Sheffield received her Ph.D in cognitive psychology in 1997 from Rutgers, the State University of New Jersey. She has been an independent audio consultant and is an assistant professor in the psychology department at Salisbury University in Salisbury, Md. She has authored or co-authored numerous publications and papers. Radio World Engineering Extra talked to Sheffield about her work in digital radio studies.



	Music Weak	Music Moderate	Speech Weak	Speech Moderate
IBOC	4.5	4.5	4.1	4.1
Delphi	4.1	3.8	3	2.1
Pioneer	4.1	3.8	2.9	2

Figure 1 – Sample Mean Opinion Score Results Showing Digital and Analog Performance with Weak and Moderate First Adjacent Interference

Taken from: Report to the National Radio Systems Committee FM IBOC DAB Laboratory and Field Testing August 2001



	Classical	Country	Rock	Speech	Voiceover
Digital	4.6	4.2	3.9	4.5	3.9
OEM Analog	3.2	3.4	3.0	3.1	2.9
After Market	3.2	3.0	2.7	2.8	2.6

Figure 2 - Sample MOS Results Showing Performance of Digital and Analog Receivers Aggregating All Field Test Conditions

Taken from: Report to the National Radio Systems Committee FM IBOC DAB Laboratory and Field Testing, August 2001

Why is it important to conduct research studies on listener acceptance of digital compression?

Any time an industry makes such a significant change in their technology, it is imperative that they understand consumer reaction to and ultimate acceptance of the new product.

In the radio industry, when I first came on the scene, there were lots of interesting assumptions being made about how consumers would react to digital, but no real hard data to back up the theories. Furthermore these assumptions were based largely on objective data collected in different labs over many years.

Unfortunately, in the digital world, the objective data didn't necessarily correspond to the subjective listening experience. Consequently, we needed a way to verify these assumptions and, at the same time, satisfy the NRSC, FCC and other interested companies that converting to digital broadcasting was a good thing.

Subjective testing made it possible to directly obtain data on consumer reactions to a variety of important questions about digital compression. And as the engineers who have been involved will tell you, some of the answers were quite stunning. Of course this makes the whole testing process fascinating and fun, particularly for us psychologists who try hard not to assume anything about human behavior before collecting data!

You are a professor of psychology. Why did NPR select you to work on an audio testing program with HD Radio codecs?

A few years ago Mike Starling contacted me to help with some studies that NPR had great interest in. After graduating from Rutgers, I worked at Lucent Technologies for two years as a research scientist in the Multimedia Perception Assessment Center, Advanced Technologies. I helped create and implement an experimental lab that conducted perceptual tests for all sorts of telephony products. At one point Lucent Digital Radio came to our team and I was assigned to work with them on their subjective test program.

A little while later I was hired by LDR, which subsequently became Ibiquity Digital Radio. I worked with Ibiquity for two more years, before returning to my first love, academia. Of course I couldn't leave the audio industry for good, so I consulted whenever I got the chance.

I guess Starling knew of my work with Ibiquity, as he had read some reports that I authored. We started e-mailing back and forth about the Ibiquity report, and he called me a few weeks later to help design two subjective experiments.

When designing a multi-faceted study like your work on HD codecs, where do you start?

Great question. Designing a multifaceted study is an iterative process.

The first thing I do is converse at great length with the people who are intimately involved with the technology. I ask lots of questions about what they are trying to find out, what their hypotheses might be, why they are interested in public opinion about the topic, etc. Test design is relatively simple once you understand what the big questions are. So, in essence, I work backwards; understanding the large goals first, and then deciding what test methodology may answer these goals.

Of course, I may not be 100 percent correct on my first design, so after creating a proposal I again visit with the experts and discuss the testing methodology further. Normally after a few rounds everyone is satisfied that the design will work. Occasionally when there are areas that need further exploration, I suggest running a pilot study.

I think the biggest mistake a designer can make is suggesting a methodology before understanding the end goal. Unfortunately, this leads to trying to squeeze study questions into a prescribed methodological format, instead of having the methodology work to explore the important questions.

For the HD coder test that NPR ran during the summer of 2004, we needed to conduct a pilot study in order to narrow the number of sound sample comparisons we were ultimately going to give to general public listeners. For a variety of reasons, the biggest one being listener fatigue, we do not like subjective tests to run over two hours, including a screening test, instructions and breaks.

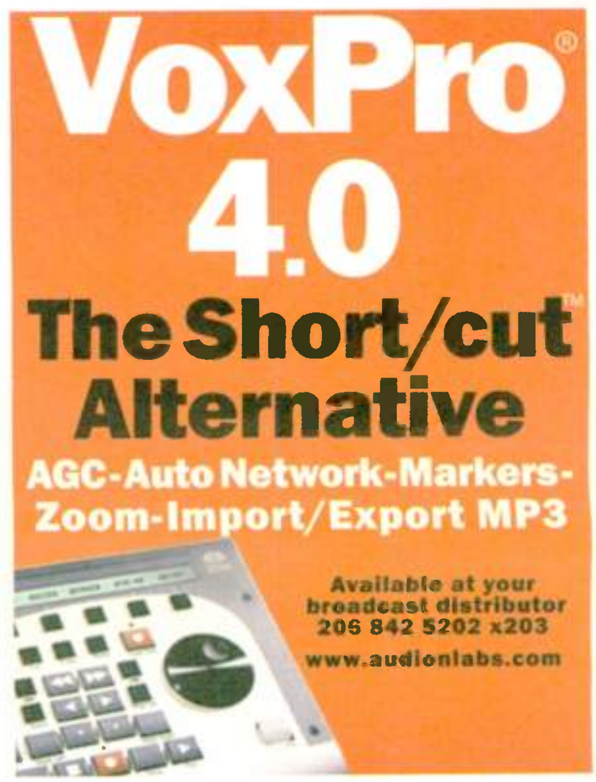
When we have more samples than we can administer in a two-hour block, we use experts to give us critical information that allows us to eliminate samples and make the test length "reasonable." The pilot was conducted with NPR engineers and employees. It was a huge success, and gave us crucial information that helped shape the consumer test we performed later that summer.

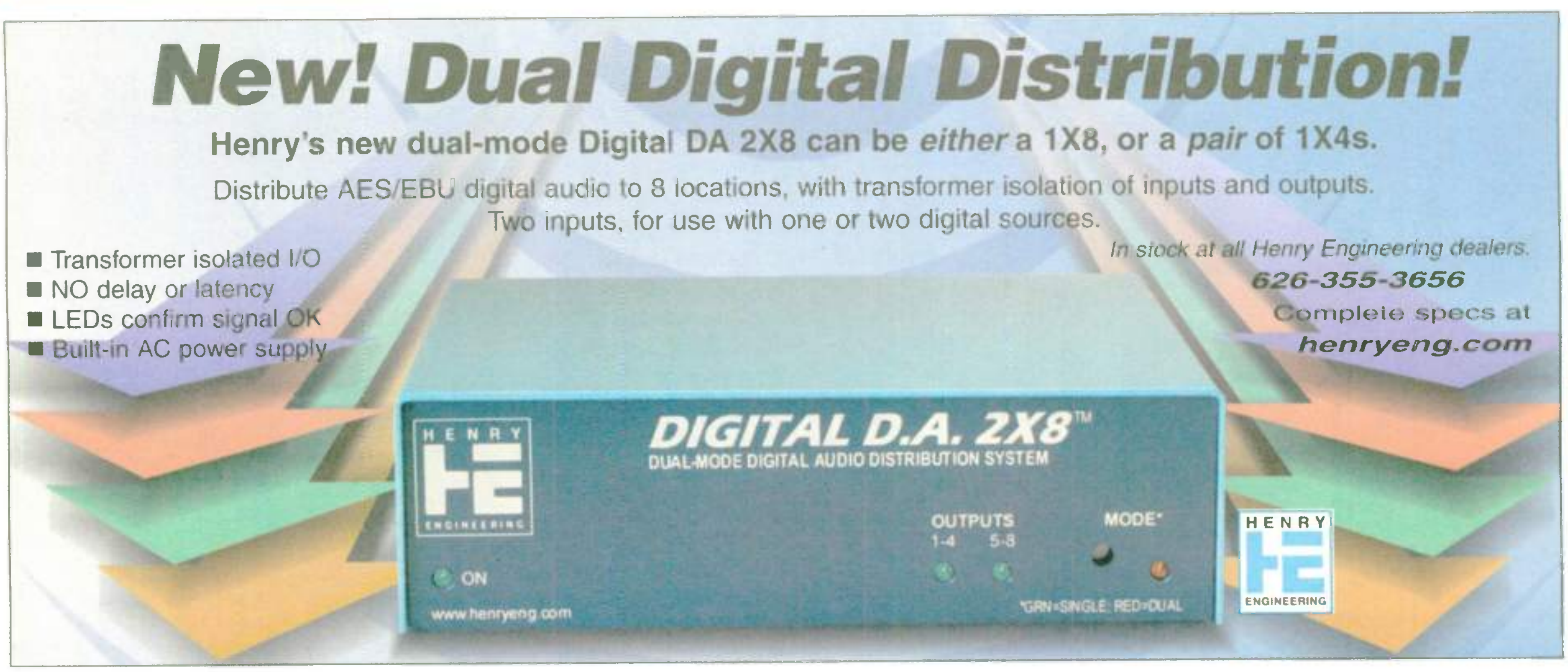
What assumptions can you make before such a test?

I don't make many assumptions before any test. I have learned over the years that

SEE SHEFFIELD, PAGE 24

Products & Services
SHOWCASE





Sheffield

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in audio testing, making assumptions about what listeners find important and relevant can be a big mistake.

The problem with assumptions is that they tend to color the design of the experiment. If you want to be scientific in your research, you must be fair and neutral while you design the methodology. Only then can you get really good answers.

In a nutshell, how does Mean Opinion Score testing work?

MOS testing was designed to allow listeners to rate individual samples on a five-point scale, from Bad to Excellent. Participants listen to samples, one-by-one, and rate them on this scale.

MOS has been used extensively in the telephony and audio industries. There are many good reasons to use MOS, including obtaining lots of data for each participant and quick and easy administration of the test. MOS testing doesn't work in every situation, but is excellent for comparing mid- to large-sized differences between sound samples.

Is such testing "repeatable" in other regions, and is that repeatability valid?

I would hope that all the testing we have done in the past is repeatable and valid. I do many things to make our results reliable.

These include using a large subject population, usually 40 subjects per test; using a diverse sample population, including both genders and all ages; using double-blind experimental techniques, which means both the listener and the experimenter have no knowledge of what is being tested; randomizing sample presentation order; performing screening tests on listeners, both prior to testing and after data collection has concluded; and running valid statistical analyses on the data.

With regard to data, as a general rule, I try to present data without collapsing or overly interpreting it — in other words, I let the data speak for itself.

The HD coder test showed some very interesting results concerning listeners' opinions about various bit-rates. Because these tests were conducted in a laboratory, and not in the field, these results should be interpreted with caution — and in my opinion they should be verified with field-testing before final management decisions are made.

The bottom line is that subjective testing in the laboratory gives us lots of excellent data, but it also needs to be interpreted carefully. Because of financial and time considerations, we tend not to "replicate" tests [or] repeat the same test in another part of the country or with another group of people.

Unfortunately this may leave us less confident than we might like when making crucial business decisions. The alternative, however, is even bleaker — making decisions without the benefit of collecting any data at all.

Is it possible to get "too much" data in such a study?

No. Too much data is an experimenter's dream come true. Of course, as I mentioned already, collecting data is expensive and time-consuming, so we tend to test the optimal number of people necessary for statistical analyses, which often turns out to be around 30-40, depending on the test.

In the NPR HD Coder Perceptual tests, there were many variables like gender, age, race, etc. How do these variables affect the testing process?

For the majority of studies, we want to include as diverse a group of listeners as possible. This practice allows us to project results from our sample to the population at large with confidence. In some cases, an evenly distributed sample population doesn't suit our needs.

For example, in the "reading services for the sight impaired" study conducted for NPR last summer, we were careful to test visually impaired people as well as non-visually impaired people of all ages. By including visually impaired people, we were able to focus on people who were directly affected by the service we were testing. By including non-visually impaired people, we were able to make comparisons between the two groups, which ultimately proved very interesting.

For most other tests, we test half males and half females in four age groups, usually 18–29; 30–39; 40–49; 50–65. After the data is collected, I statistically analyze it for differences in groups. So, for example, I will perform an analysis that examines whether females' and males' results differ significantly from a statistical point of view.

Over the years, I haven't found many differences between groups, except that young people consistently rate sound samples more critically than old people. However this is a global effect, meaning that young people are more critical of all samples, including CD source material, digital, analog, etc., not just one type of sample.

'I have learned over the years that in audio testing, making assumptions about what listeners find important and relevant can be a big mistake.'

Are you using this same methodology on other non-radio projects?

Yes and no. I have used the same or similar methodologies in telephony and television testing before, but the methodology is always tailored to the specific problem at hand.

The international standards setting body, the International Telecommunication Union-Radio Communication Sector recommends a variety of methodologies for audio testing, but also recommends that experimentalists carefully design studies to fulfill individual test needs.

The methodology I designed for the HD coder test was specifically geared for the questions NPR and the radio industry wanted to answer. Some of the methodology followed ITU-R recommendations, while other parts were constructed independently of prescribed methodology.

What other audio research projects have you worked on?

I have worked on many, many subjective test projects for Lucent, Ibiquity, ATTC and NPR. I am currently working on one audio project for the NRSC-AMSTG and another for NPR focusing on surround sound.

Would you like to work on any future NPR or broadcasting projects?

I continue to be interested in the audio and broadcasting field, and am always happy to work on radio projects.

As a professor at Salisbury University, working for the audio industry has afforded me a wonderful opportunity to conduct applied research that I enjoy greatly and may otherwise not get a chance to do. Being part of such a vital industry and getting to know so many great people has been a real treat for me.

Steve Callahan is the assistant chief engineer for the WBUR Group in Boston. ■

FMeXtra

As FMeXtra gains in popularity during the next few years, the push will come from the small station owners, not from the big corporations that have invested millions of dollars in HD Radio. The equipment manufacturers and big owners will continue to promote HD, but it will eventually go the way of AM stereo.

IBOC is a misnomer. It is in-band, but not on-channel.

AM HD is already in big trouble. IBOC is a misnomer. It is in-band, but not on-channel. The only equipment manufacturers currently promoting FMeXtra are Bext and Energy-Onix. If other manufacturers sell it, it is only so they will have something to offer in foreign markets that do not permit HD Radio.

Hopefully the FCC will eventually admit that HD was a technical error.

Stanley Swanson
KYRM(FM)
Yuma, Ariz.

Fresh Format

I read with great interest "Can Radio Fight the Internet Revolution?" (Dec. 14).

I have owned an AM station in Greenville, S.C., for 18 years and we have never sold one advertisement. We play music 24/7, totally uninterrupted except for our legal ID. We get rave reviews and our listeners prefer us over any other media because we have 10 million songs in our format that they cannot find elsewhere. We recently went digital, and we will just see how soon radio and even AM radio withers away.

People tell me the formats on satellite radio grow old after continued listening. Their formats are small compared to ours. We have listeners of all walks of life that call, write, come visit, e-mail and stop us on the street to thank us for what we offer them. They don't have to work at it at all. They simply turn up the radio and there we are.

We are not local, not loud, not funny and not trying to be the biggie. We play a very interesting mix. You never know what comes next. Listeners keep telling us they like it just fine. Try us at *www.papercuttersinc.com*. Click on the horn, sit back and watch us wither to the top of the tree. We have had our format on the air for over five years speechless and it has not yet gotten stale.

Randy Mathena
Owner and Programmer
Paper Cutters Inc.
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HD Tabletop Received With Eagerness

Clean HD Multicast Performance in a Table Radio Finally Arrives With the Boston Acoustics Recepter

Guy Wire is the pseudonym for a veteran radio broadcast engineer.

It seemed like it would take forever, but consumers finally got their first taste of a real tabletop HD Radio just in time for Christmas. Like so many of my radio-engineering brethren, I just had to spring for my own Boston Acoustics HD Recepter at the lofty retail price of $500. (The retail price was lowered to $299 subsequent to my review.)

I got a $20 rebate from the Crutchfield online deal, but never in my wildest dreams did I actually think I would ever pay that kind of money for a small, bedside radio. I easily rationalized the purchase by reminding myself the radio industry only invents and adopts a totally new transmission mode about once every 50 years. I really wanted to be at the starting gate of radio's third new era.

BA has stepped up to establish a milestone of sorts by delivering the first full-featured tabletop HD radio to market. No doubt those that follow from other manufacturers will be measured by the performance of the Recepter.

A MILESTONE FOR HD RADIO

Most important, perhaps, this radio features multicasting capability. Its introduction coincides with the recent unveiling of HD-2 format assignments by the HD Radio Alliance. Most of the early HD-2 services will initially be commercial-free, just like satellite. The national press is finally taking note by touting multicasting HD technology as terrestrial radio's counter-punch to satellite competition.

Getting easy-to-use multicast HD radios — which offer the consumer something more than just "digital sound" — onto store shelves is a crucial turning point in the HD rollout. Many articles showcase the Recepter as the first radio the public will see as a viable alternative to satellite for more variety and more choices.

I've spent considerable time playing with several Kenwood and Panasonic aftermarket HD car radios purchased soon after a few of our stations installed HD transmitters. But after-market car radios are only a tiny fraction of radio sales. Without HD OEM car radios, portables or tabletops, there has been little for the average buyer who might be interested in HD. The Recepter puts an end to that frustration.

I certainly didn't want to inflict a first-generation aftermarket HD radio on my almost new SUV complete with an integrated navigation display, so I decided to wait for the Recepter and put it through the paces during Christmas vacation.

BIG SOUND IN A SMALL PACKAGE

First impressions are lasting ones. BA radios deliver impressive bass and sport a digitally equalized amplifier with BassTrac circuitry. In fact, the Recepter may be a bit bass-heavy, especially on AM stations. Fortunately BA includes a bass trim control in the setup menu to allow for contouring the bass response for room conditions and personal taste.

Stereo performance is an added feature in the HD version of the Recepter via a separate second speaker, including an extension cord for wide separation installations. The overall frequency response is excellent with plenty of crisp highs to balance the big bass. Stations jump right out of the Recepter and fill up an average-sized room.

The bright blue LED tuning and text display features an HD indicator that blinks when HD content is first recognized. It blinks for about five seconds until HD lock is acquired and then stays constant when HD decoding replaces the analog program.

It's easy to spot HD stations that are not precisely time-aligned with their analog. I found a few that must still be tweaking their installations and had no analog delay, which of course presents the listener with an eight-second out-of-synch train wreck every time blending occurs. A few others appeared to be slightly out-of-synch, which causes a brief overlapping echo during blending transitions.

ANTENNA CONSIDERATIONS

BA includes a simple detachable wire on an F connector for the inside FM antenna as well as an external AM loop that includes a built-in plastic stand to position it for best reception. For fringe area and shadowed locations, an external antenna of some kind may be needed, especially for reliable HD and FM supplemental channel performance.

When signal levels are adequate and stable, HD and supplemental channel reception is solid as a rock. Even if the analog signal is contaminated by multi-path or noise, HD will lock and perform well as long as there is sufficient signal strength. This is the most common reception condition where HD shines. Hearing noisy analog blend to crystal clear HD does elicit a "wow" experience. It is awesome to hear the difference on AM.

On the other hand, when the analog signal seems relatively clean but attenuated because of shadowing or distance, receiving HD can be a bit challenging when using the short wire antenna. I found myself trying all kinds of various orientations for the wire to find a "sweet spot" where HD would lock on weaker stations, but discovered HD would not remain reliably stable. The Recepter could use enhanced front-end sensitivity to better cope with these conditions.

The obvious solution in such compromised signal environments is to employ an outside antenna. That has always been an option for those who want to go the extra mile to improve reception for a favorite station of any kind. But I fear that the majority of consumers who buy this kind of radio will not work that hard to make marginal HD signals work well, especially if there are enough other choices on the dial that do perform adequately.

Without blend to analog, supplemental channel HD programs simply mute when the HD lock is lost. This is a serious disadvantage when using HD in marginal receiving locations, but just like many satellite radio listeners who want to use their portable sets inside, buying and installing a suitable external antenna nails the problem.

HD TUNING MADE EASY

Tuning in supplemental channel HD programs is a snap and works the same as most HD car radios. Once the analog channel is set and the main HD-1 program locks, the user can merely jog the tuning dial one click up frequency to find an HD-2 program. If an HD-3 program is present, one more click will tune that in.

When a supplemental channel is established and locked, and the user then tunes in other stations, that same supplemental channel reappears first when the user returns to the main carrier frequency, instead of its main HD-1 program channel. If the user wants another supplemental channel or the main channel of that station, he merely clicks down frequency to tune it in.

There is no delay in capturing supplemental channel audio as long as HD-1 remains locked. This is an improvement over most of the HD car radios that revert to a muted "linking" period every time the channel knob is changed. If any of these channels contains program-associated data, like song title and artist, it will be conveyed on the text display.

CHALLENGING CHANNEL LABELS

While the industry is still trying to decide how supplemental channel programs should be labeled or identified on HD radio displays, receiver manufacturers with product in the market already seem to have settled on FM-1, FM-2, FM-3; or HD-1, HD-2, HD-3, etc.

Either format seems reasonable as a starting point, but this will undoubtedly cause some confusion for many HD users who encounter supplemental channel programs that have nothing to do with the main HD-1 station. Quite a few HD stations owned by multi-station clusters are simulcasting a sister AM station with completely different call letters and format on one of their HD-2 channels.

What needs to be incorporated in the standard bit-stream HD format is the ability for stations to be able to label their supplemental channels as they see fit. Perhaps an eight-character alphanumeric identifier, similar to the RDS ID format that can be user-programmed and modified as necessary into their HD exciter for any supplemental channel, would be feasible.

One nice feature of the Recepter, not offered in other HD radios I've test-driven, is the ability to display RDS data. Unfortunately it only works on non-HD stations. If you transmit HD with or without PAD, as well as RDS, you won't get to see the RDS data.

REDUCING INTERFERENCE

Selectivity and adjacent-channel suppression performance of the HD Recepter is impressive, perhaps the best in any consumer radio I've used. While those specs

SEE GUY WIRE, PAGE 29

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Guy Wire

CONTINUED FROM PAGE 27

are not published in the owners manual or available online, the design folks at BA obviously understand the challenges HD creates in the way of sideband interference energy. The company has done a nice job of minimizing its effects.

I was able to listen to a fringe area second-adjacent FM analog station next to a local FM HD signal with no problems. I could listen to a short-spaced first-adjacent analog signal next to an FM HD signal by optimizing the antenna orientation with no more interference than one might expect, even with the presence of HD sideband energy.

Selectivity performance on AM also is impressive. When tuning to a first-adjacent channel, sideband splatter seems to be gated or squelched off so that it is not nearly as annoying. Even while listening to an AM HD station on its first-adjacent channel, the level of digital hash is significantly attenuated. This suggests that new design tricks can be used in the new generation HD receivers to minimize digital noise interference to neighboring channels.

Unfortunately as of this writing we were not able to find a good example of receiving a station that was adjacent to an AM HD station during critical hours to better evaluate HD noise interference performance. Many engineers and observers fear this problem could wipe out large areas of nighttime coverage for such stations, even inside their protected contours.

IN PURSUIT OF AUDIO EXCELLENCE

For stations that have intelligently applied appropriate processing and bit-rate choices, the transition from analog to HD reveals strikingly cleaner and detailed high frequency response. Not having to cope with high-frequency pre-emphasis limiting restores the ability to hear unmolested high-frequency performance on an FM radio station for the first time.

The realism of more faithfully reproduced source material, including dynamic punch and clarity, is breathtaking on HD for anyone who appreciates good audio. Obviously HD stations that have no supplemental channels and use the full 96 kbps rate for HD-1 perform the best in this regard. But at 64 or 48 kbps, the minor degradation of nuance coding artifacts is generally less annoying than that caused by over-processed analog with a heavily limited or clipped high end.

Even on stations transmitting both HD-2 and HD-3 channels that typically use bitrates at 32 kbps and lower, the audio quality is remarkably decent if the source quality is maintained. The few HD-3 channels we could evaluate all belonged to NPR stations that are placing long-form talk formats or BBC news service rebroadcasts on that channel.

Coding artifacts are certainly noticeable at such aggressively reduced rates and are manifest primarily in a more unnatural, metallic sounding high end. But those are minor compared to the degradation of noise and low fidelity on AM and shortwave broadcasts. Supplemental HD will be a big breath of clean, fresh air for the typical consumer who uses these services.

Observing how HD stations are processing their analog vs. HD during this early rollout period reveals two schools of strategic thinking. Some stations have carefully matched the loudness, texture and EQ, making the transition during blend almost unnoticeable, except perhaps for the more open and detailed high end on HD.

Others seem to have purposefully made HD louder, brighter and "bigger" to apparently attract attention to the difference and convey the impression that HD is "better." The temptation to over-process HD is a slippery slope that can quickly become counter-productive. I prefer to hear HD sound as natural and unaltered as possible so it can deliver a comparable CD or iPod listening experience.

PROMOTING THE LOVE

Stations embarking on promotional activities showcasing HD will no doubt want to acquire quantities of Recepters for contest giveaways. Any fan of radio will love this unit. But perhaps the most important feature on it for some will be the aux input connector for their iPods. Oh well.

The only major complaints I can register about the Recepter are the lack of sensitivity to keep HD reliably locked using the wire antenna on weaker stations, and of course the price, which the manufacturer subsequently adjusted downwards by 40 percent.

Got a beef or compliment for Guy? Write to gwire@imaspub.com. ■

Blesser

CONTINUED FROM PAGE 30

It has often been said that success arises from the ability to notice luck when it presents itself, and the wisdom to take advantage of it when it appears.

This principle applies equally to managing a radio network or a professional career. Focusing only on the problem and its proposed solution ignores unintended consequences, which may be good or bad. Unintended consequences are always opportunities: — "the tail wags the dog."

Second, because we cannot know the properties of an unexpected event, having broad skills and knowledge raises the likelihood that we can take advantage of an opportunity when it appears. This immediately couples back to an earlier column ("The Paradoxes of Learning," Oct. 19) on the importance of having an efficient mechanism for learning, especially tangential topics. The more you know, the higher the likelihood of detecting and then taking advantage of an opportunity.

Third, every long-term problem should have a plan A, plan B and plan C. Not only do multiple plans provide flexibility, but they also reinforce the awareness of unexpected opportunities. An ideal plan will always need to be modified because it will have chaotic components. Having contingencies is not a reflection of pessimism or inadequate analysis; rather it is the explicit acceptance of randomness.

With these three conclusions, we have unified earlier Last Word themes into a holistic viewpoint: analysis, planning, learning and randomness. This is my version of wisdom: a way of living, not an academic subject.

Dr. Barry Blesser is director of engineering for 25-Seven Systems, a former associate professor at MIT and past president of the AES. Contact him with your comments at barryblesser@25-seven.com. ■

MARKETPLACE

Titus Offers 3DRX Repeater, Switcher

The Titus Technological Laboratories 3DRX is a digital audio repeater and cross-point switcher. It has three inputs and one output.

The first two inputs are digital audio inputs. They can be either AES-3 balanced, XLR-type inputs or AES-3ID BNC-type inputs. They also can be either terminating or bridging inputs allowing sampling of an existing serial digital audio data channel. The third input to the 3DRX is a stereo analog audio input. The serial digital output conforms to AES-3 and AES-3ID standards.

The 3DRX automatically switches to the secondary digital input if the primary digital source fails, such as in the case of loss of lock or data error. A tertiary stereo analog source will be switched to if the two stereo digital streams fail. The stereo analog inputs are digitized in the AES/EBU format at 24 bits and either 32 kHz, 44.1 kHz or 48 kHz sample rate.

The 3DRX also can be controlled by remote control, allowing the user to select the active source remotely or from the front-panel push button controls in either manual or automatic mode.

The company says the 3DRX is suitable for use as a digital serial audio repeater when used in a "transparent" operating mode (AES in to AES out). The digital signal in this mode is not decoded into analog but remains in the digital domain. When the 3DRX is operating in this mode and inserted into a digital serial audio cable run the length of the cable can be lengthened. This in an active repeater and not a direct "hardwire switch" path so it can be used to extend cable lengths.

The 3DRX operates up to 24 bit, 96 kHz stereo digital audio for digital inputs and outputs.

For more information, contact Titus Technological Labs in Connecticut at (860) 633-5472 or visit www.tituslabs.com.

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by Barry Blesser

Chaos Theory and Radio's Possibilities

In my *Last Word* column on stakeholder analysis ("Tools for Analyzing Your Future," Dec. 14), I ended the discussion by teasing readers with a reference to Chaos Theory. It deserves more than a casual mention because it is one of the more profound discoveries of the 20th century, with applications to a wide range of phenomena.

Chaos Theory, as one part of Complexity Theory, analyzes the properties of unpredictable systems. The radio industry is an example of an unpredictable system embedded within an even larger complex system. These theories help explain the limits of what we can understand about our industry. No amount of thinking will make sense out of the random attributes of a system.

The story of how chaos was discovered illustrates its implications. In 1961, Edward Lorentz was doing weather simulation with computers. When he restarted a simulation with data that was only insignificantly modified by numeric truncation, the simulation then produced completely different results. Storms appeared where previously there had been sunshine, and vice versa.

The phenomenon of chaos, which would later be called the butterfly effect, also applies to social and technology trajectories. Consider that a butterfly could change history if it distracts a hunter who then accidentally kills the scientist who would have developed a new type of modulation that would change how audio is distributed. Theoretically, a butterfly can dramatically influence the future of radio.

Several examples illustrate how minor events came to be seen as the birth of revolution.

A FATEFUL CHOICE

After developing the PC computer in 1980, IBM attempted to license the CP/M operating system from Digital Research, but negotiations failed because the owner's wife, Dorothy McEwen, would not sign the non-disclosure agreement. As IBM's second choice, Bill Gates licensed QDOS (quick and dirty operating system) to IBM after having just acquired rights to it from Tim Paterson at Seattle Computer Products.

The way in which computers changed broadcasting was clearly the result of a sequence of unpredictable events by these seemingly insignificant individuals. Any of the parties involved in starting the personal computer industry easily could have selected a slightly different path, like truncating data in a weather simulation.

Similarly, Linus Torvalds was indulging his software hobby when he created a primitive — but free — operating system, called Minix, which would later become Linux. And his contribution was itself a response to the earlier work of Brian Kernighan and Dennis Ritchie at Bell Laboratories, who developed Unix and the C programming language because executive management was phasing out Multics.

These individuals wanted a new computer playground to replace their mainframe time-sharing system. They had not set out to change the world, but they did.

The Internet also is a perfect example of Chaos Theory in action. The current Internet is the result of minor choices made by thousands of contributors.

In contrast to the failure of numerous dot.com companies, Jeff Bezos's Amazon and Larry Page's Google are unique. Not only did they survive and thrive, but they also created a new business model: low overhead selling and advertising. These individuals did not set out to change the relationship of advertisers and listeners to radio broadcasting — but they did, through a long series of seemingly unrelated event trajectories.

VIDEO GAME ALTERS HISTORY

A personal story about the origins of digital audio illustrates the degree to which trivial events changed the world.

While sitting in an MIT laboratory at 3 a.m. playing a video game in 1970, I noticed a colleague, Francis Lee, who specialized in computer memory. We discussed how the combination of audio and computers might produce a commercial product. Our conversation eventually led to the founding of Lexicon with the first commercial digital delay line, and that event induced EMT to invest in the first digital reverberator.

Observing the success of these two small-scale endeavors, Sony and Philips began the development of the digital audio CD, which provided a market for high-quality, low-cost digital-to-analog converters.

> **Only by looking backward can we identify those events that would later prove to be significant. But at the time events are happening, we cannot identify those that will have 'revolutionary' consequences.**

Had I not just lost that video game, and had I not stopped to chat with Francis, the history of digital audio would have taken a different trajectory. Our unplanned interaction was just one of many possibilities.

Digital audio would certainly have come about had I gone to bed earlier, but its properties would have evolved differently because different people would have reacted to different contexts at different times. A decade later, the Audicy workstation for radio production resulted from a friend breaking his leg while jogging over an ice patch.

If these events had not happened, my career would have certainly taken a different path, and there never would have been any Last Word columns. And without these articles, some reader would have made a different decision managing radio technology. Like the butterfly, losing that video game changed the radio industry.

There is, however, an asymmetry between trivial events and the profound changes that they cause. While all changes arise from the minor actions of a few individuals, only a tiny fraction of such actions actually have consequences. Only by looking backward can we identify those events that would later prove to be significant. But at the time events are happening, we cannot identify those that will have "revolutionary" consequences.

Chaos Theory, a mainstream intellectual discipline that explores the properties of unpredictable systems, considers the implications of minor perturbations. How is it relevant to the radio industry? Like all large systems, our industry has a chaotic (random) component. No amount of analysis by the most brilliant minds can accurately predict the consequence of minor choices made by thousands of individuals.

In other words, stakeholder analysis, like predicting the weather, works best in the short term but fails miserably in the long term. When a seemingly random event eventually changes the paradigm of a stakeholder group, its relationship to other stakeholder groups also changes, and so on.

In 2006, there also are thousands of trivial events occurring, and a few of them will have a profound effect on the broadcast industry of 2016. But nobody can determine which events will have what effect; analysis can predict the equivalent of the weather for the broadcasting industry over a year or two, but not a decade.

EXPECT THE UNEXPECTED

What are the implications for individuals living in a chaotic system without any means of predicting how that system will evolve? The initial step is to accept the premise that there are limits to analysis and knowledge; there is a random component. We are then left with several important conclusions.

First, because events follow the law of unintended consequences, we should always be on the lookout for the sudden appearance of an unexpected opportunity.

BLESSER, PAGE 29

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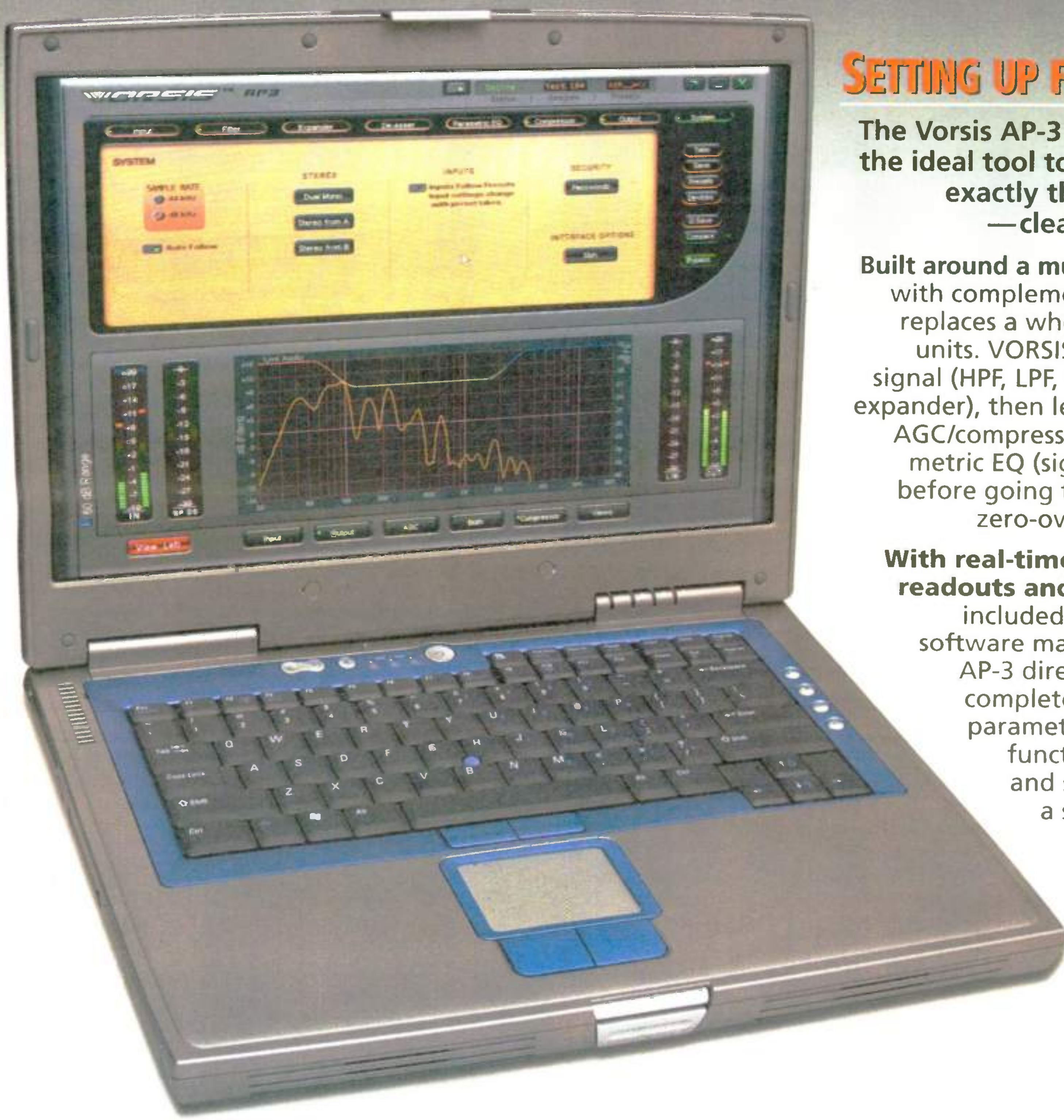

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