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DESIGNER INTERVIEW

Richard Schrag

The Art and Science of Studio Sound

Richard Schrag of Russ Berger Design Group Explains How to Make Studio Acoustics Work Right the First Time

By Michael LeClair

A common belief about acoustics is that it is more of a black art than a science. Yet the final acoustics of a studio are determined by factors that are largely controllable in the design process. A well-designed studio can

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Evaluation and Improvement of AM Antenna Characteristics

By Ronald D. Rackley, P.E.

The author is vice president of du Treil, Lundin & Rackley Inc., Sarasota, Fla.

INTRODUCTION

AM digital transmission places high demands on antenna system bandwidth, both from the standpoint of the input impedance at the transmitter load, which can cause noisy analog reception, and the antenna's far-field radiation characteristics, which can erode the error-correction capability of the digital signal and make its reception inconsistent. Directional antenna systems present the additional complication of sideband phase and amplitude errors resulting from changes in pattern shape with frequency, which can render digital modulation undecodable in areas with satisfactory analog reception.

AM IBOC transmission technology is currently moving from the realm of the theoretical into the "real world," and, although the body of on-the-air experience with it is limited, it is rapidly expanding. Measurements can be made to evaluate the important aspects of antenna performance and, in most cases where performance is found to be lacking, relatively simple measures can be taken to improve matters.

MODULATION BASICS

As the ultimate goal in digital transmission is to have the signal pass through the antenna system and arrive at the receiver with its components in their correct relationships, it is best to start with an understanding of the natures of those components and how they might be affected by the transmission system.

Digital modulation is accomplished by varying both the amplitude and the angle of the RF carrier simultaneously to transmit the digital bits that are decoded in the receiver. In the case of in-band, on-channel (IBOC) transmission such as is being introduced in the United States, this is done while still transmitting the conventional AM signal by sending the digital information at a low level so that the resulting occupied spectrum fits within the "RF mask" of an AM channel as defined in the FCC Rules.

A conventional AM signal is modulated in amplitude only, and has an uncomplicated spectral display that is directly related to the spectral content of the modulating audio. For single-frequency (sinusoidal) audio modulation, the RF envelope waveform is produced by two sidebands separated from the RF carrier frequency by the modulating signal's frequency. For 100 percent modulation, each sideband is one-half the amplitude of the carrier (see Fig. 1).

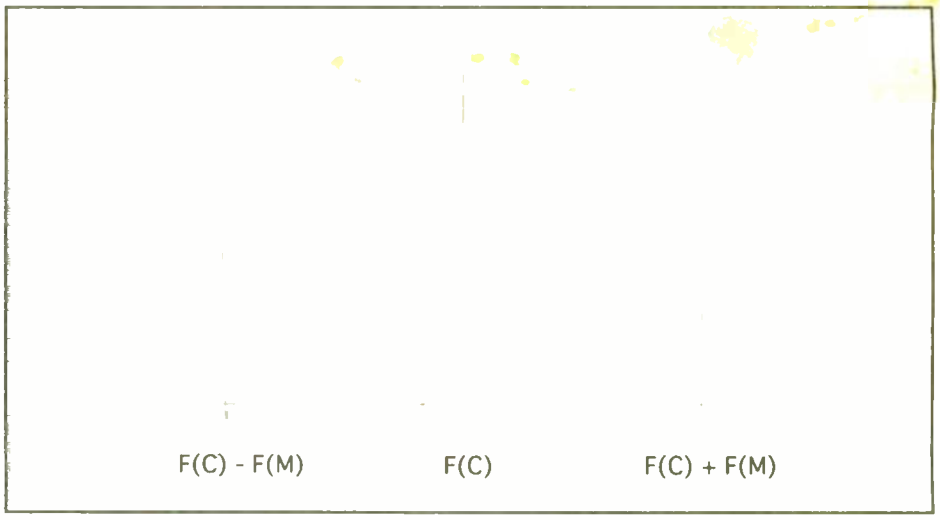

Fig. 1: AM Sinusoidal Modulation Spectrum (Frequency Domain)

A spectrum analyzer screen showing such modulation contains the three components of the signal plotted in terms of frequency (in the frequency domain).

It is very difficult to visualize the components of even the simplest form of modulation in the time domain, since the

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Thoughts on Digital Radio

by Michael LeClair

Michael LeClair

Just the other night, I commissioned my first HD Radio transmitter, on an FM station. It was a surprisingly simple operation compared to the changes required in the transmitter plant to prepare for it.

We chose to go with high-level combining, which meant upgrades to the electrical and HVAC systems. We also needed to change the studio-to-transmitter link from composite to digital. After all that work, the final installation and modification to the transmission lines took only a few hours.

Of course, one of my priorities the next day was to arrange some listening time to our new signal in the studio. We have one of the early car radios installed as an HD Radio monitor, so we can compare three audio versions in the same reference environment: the studio line output, the FM air monitor and the HD Radio signal.

The results are impressive on a radio well inside our coverage contours. Despite the requirement of a high ratio of lossy data compression, the digital radio signal sounds as good as the FM version, if slightly different. I suspect the latter effect is due to the lack of pre-emphasis in the audio processing for the digital signal. The alterations to the signal caused by the final audio processing are far more audible than any artifacts caused by the digital compression.

WORRISOME

It seems the FM digital system has delivered most of what it promised in terms of audio quality, which leads me to think about the AM IBOC digital system and the potential improvements it offers.

I am old enough to have grown up when AM radio was still in its prime; and my first radio memories are of listening to the giant clear-channel signals. To me, AM radio remains a great medium, one that I love. Its current state worries me.

The AM band stands to benefit the most from the audio quality available with digital conversion. The advances in audio coding over the last five years have made low bit-rate digital audio streams quite good. Applying these advances to create a digital AM will give the quality improvement that has been needed for many years to counteract the increasing noise and interference that limit AM coverage.

My first experience of Internet audio streaming at low bit rates via telephone modem was a complete turn-off. The swishing artifacts and strange echoing caused by the compression algorithms were unlistenable to my ears, although to the pioneers of Web streaming it sounded wonderful.

I remember having a "discussion" with a Web enthusiast about the relative merits of AM radio vs. Internet streaming; we just couldn't come around to understanding each other's point of view. Audio artifacts didn't bother him at all; he was enthralled by the overall frequency response compared to telephone sound. I couldn't stop hearing the artifacts; no amount of frequency response could compensate for the distraction of hearing strange noises underneath the audio.

The good news is that the technology for digital radio on AM is out there and that it works.

We were both partly right. Digital compression had great potential, even if it hadn't yet delivered on its promise.

LAGGING AM

However, as digital compression technology has improved, and the understanding of how to process digital audio has increased, it is no longer a contest.

Internet audio streaming now is superior in quality to AM radio, at least as heard by the majority of listeners on consumer radios. With the rapid advance in wireless technology, it won't be many more years before terrestrial cell-type networks can deliver a mobile audio stream with similar coverage to even the best AM station, at higher fidelity and with practically infinite program choices.

Unfortunately, the cost of converting AM to digital is much higher than in the case of FM. Most important, the overcrowded band does not painlessly accommodate new digital sideband carriers. A number of AM signals will take new interference and lose coverage due to the implementation of a digital system that operates in adjacent channels, (although certainly not the majority of stations). The worst of these interference effects occur at night.

Imagine the plight of a mighty daytimer whose night power is reduced to 10 watts due to international treaties. New interference from an adjacent high-power station could potentially swamp out this tiny signal, even though the digital sidebands are allowed only 1 percent of licensed power. Different methods of resolving this interference have been proposed, but so far no solution has been adopted.

The economic cost of digital conversion may also be somewhat higher for AM, particularly for directional arrays. While some stations have reported a need for only minimal improvements in their AM transmission systems, I suspect that many AMs would be facing wholesale replacement of older antenna systems from the 1950s and 1960s.

Additionally, the studio facilities of many AM stations are not as prepared for digital conversion as the typical FM. To take full advantage of digital quality improvements, the audio chain will need to be upgraded as well.

However, there is another economic cost that is more painful. In our market-driven economy it is not possible to stand still as other broadcast media surpass the performance of an older technology. The eventual result for AM radio, should stations decide not to pursue improvement, will be the loss of new revenue and a slow decline into irrelevance. A solution that permits the upgrading of AM to digital has to be found if it is going to survive in the long term.

OPTIMIZATION

The good news is that the technology for digital radio on AM is out there and that it works. New developments for the AM band continue at a regular pace.

In this issue we have a couple of papers relevant to AM broadcasting and digital conversion.

Ron Rackley offers us a look at how to optimize AM antenna systems for digital modulation, using a simple vector approach to analyze amplitude modulation. We also have a paper on a new high-power AM transmitter design from Broadcast Electronics that is ready for digital operation.

I am also happy to note that masked engineering columnist Guy Wire, long a denizen of the virtual world, will be joining us in print on a regular basis as well. His provocative and insightful comments are intended to get you thinking — and talking back.

We invite your comments and suggestions about anything you read in Radio World Engineering Extra. In large part we look to our readers to help define the direction of this paper, and we want to know how we are doing so far.

Please send an e-mail to me at *mlrwee@verizon.net* with your thoughts and opinions.■

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CONTINUED FROM PAGE 1

carrier and sidebands are all varying sinusoidally over time — although at different rates. Normally, complex mathematical expressions are used to explain modulation in the time domain using phasor algebra. Fortunately, it is possible to show the relative "motion" of the carrier and sideband phasors using vector analysis by referencing the degree of rotation of the sidebands to a fixed point in the rotation of the carrier phasor.

The two sidebands of a conventional single-frequency modulated AM signal appear as equal-length vectors that rotate in opposite directions relative to the stationary carrier vector — and are always displaced from it by equal but opposite angles. At the point in the modulated RF envelope where its amplitude is equal to that of the carrier alone, the two sidebands oppose each other to completely cancel in the resultant of the vector addition (Fig. 2).

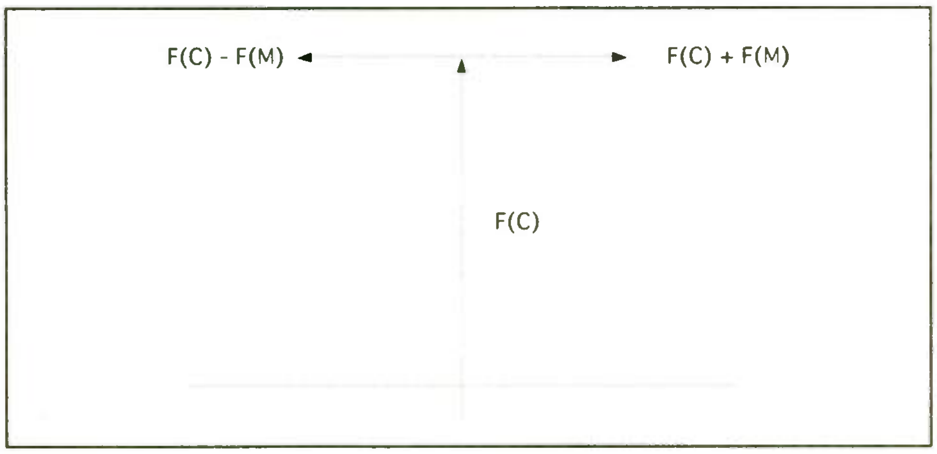

Fig. 2: At Zero Crossing of Audio Waveform

As the sideband vectors rotate so that the resultant is greater than the amplitude alone, the waveform progresses toward a positive modulation peak (Fig. 3).

As the two sideband vectors fall in line with each other in line with the carrier vector, the resultant approaches twice the carrier magnitude to form a 100 percent positive modulation peak (Fig. 4).

Similarly, when enough time has passed for the two sideband vectors to rotate to the opposite direction, the resultant will approach zero to form a 100 percent negative modulation peak (not shown).

Except when testing with tone modulation, AM signals are much more complicated. The complex waveforms of normal programming have many frequency components. In a perfect system, however, the components will always be divided into equal, symmetrical sideband components (Fig. 5).

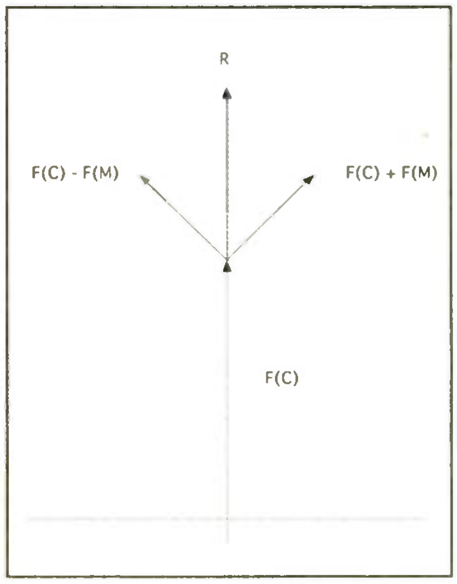

Fig. 3: Positive Modulation

A spectrum analyzer shows the range of frequencies occupied by the various sideband components for normal program modulation (Fig. 6).

Suppose the sideband vectors are symmetrical about an imaginary line running perpendicular to the carrier vector instead of to the carrier vector itself — in other words, the sidebands of AM transmission are rotated 90 degrees relative to the carrier. The resultant will not always fall in line with the carrier, but will vary in angle from it over time (see Fig. 7).

Fundamentally, this will be angle modulation, which, in analog transmission, may be either phase modulation or frequency modulation depending on the process applied to the audio to produce it. Careful observation reveals that angle modulation accomplished with only two sidebands for the single-frequency, sinusoidal case would also produce secondary amplitude modulation, since the resultant follows a line tangent to the circle that would be traced by the carrier vector if only its angle were changing. That is why an FM signal has many sidebands even if only a single-frequency tone is being transmitted — the infinite number of harmonically related sidebands that trail off in amplitude and vary in phase are necessary in order to have the resultant trace the circle rather than the tangent line.

Fig. 5: AM Complex Modulation (Vector Representation)

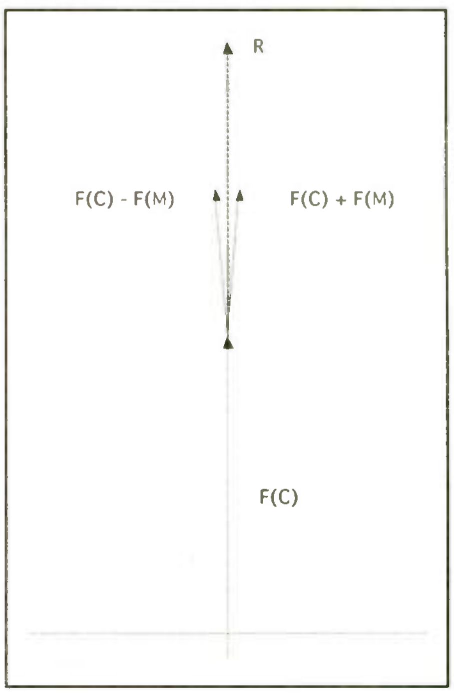

Fig. 4: Approaching 100 Percent Positive Modulation

DIGITAL TRANSMISSION

Digital transmission utilizes a combination of amplitude and angle modulation to send the bits of information that are decoded by receivers. Such modulation generally is referred to as I-Q modulation, as each bit has a unique combination of I (in phase with the carrier) and Q (in quadrature relative to the carrier) components assigned for it. Fig. 8 depicts the basic concepts involved in sending a 16-bit digital "constellation" with I-Q modulation superimposed on a conventional AM signal. It is a conceptual sketch only, and does not represent any actual transmission system. It is not to scale.

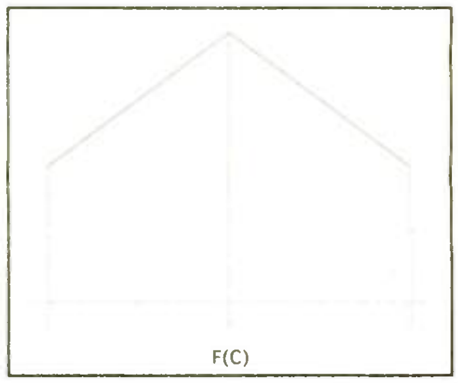

Fig. 6: AM Complex Modulation Spectrum (Frequency Domain)

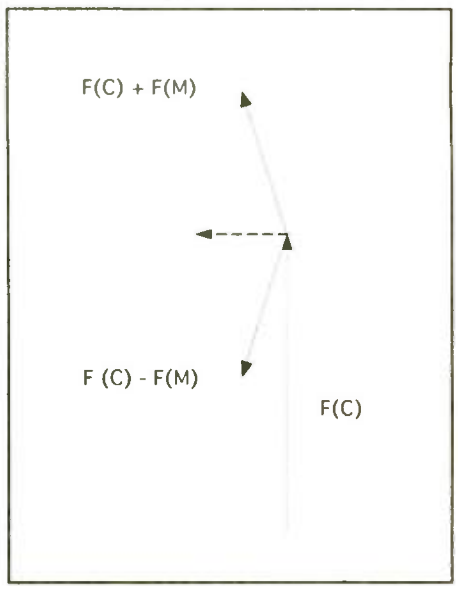

Fig. 7: FM-PM Modulation First Order Sidebands

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It can be seen that, for the bits to arrive in the correct relationships to be decoded by a receiver, the transmission system must maintain their correct locations in terms of I and Q and they must remain isolated from the AM signal.

IBOC digital transmission occupies spectrum in addition to that of a conventional AM signal (Fig. 9).

TRANSMITTER LOAD OPTIMIZATION

When a transmitter with a perfect modulation system is connected to a perfect load (one that does not change impedance with frequency), perfect transmission of the digital signal will result. Modern transmitters that have been designed to carry digital modulation approach perfection, at least for all practical purposes. Real-world antennas, however, do not.

Since it is not realistic to expect antennas to have input impedances that are constant with frequency, the goal of having the effects of changing impedance at least be symmetrical must be pursued. If the load impedances at sideband frequencies differ from the carrier frequency in symmetrical fashion, crosstalk between the digital and analog signals can be minimized.

It is important to note that the symmetry must be provided to the actual active components of the final amplifier, where the process may be simulated most closely by a perfect source (Fig. 10). The circuitry that provides impedance matching and bandpass filtering between the final amplifier and the transmitter output port must be considered if it is impossible to make impedance measurements at the final amplifier. Assuming that the output network of a transmitter is adjusted to provide the correct load impedance for the final amplifier at carrier frequency, the most important thing to know about it for sideband symmetry analysis is its phase delay.

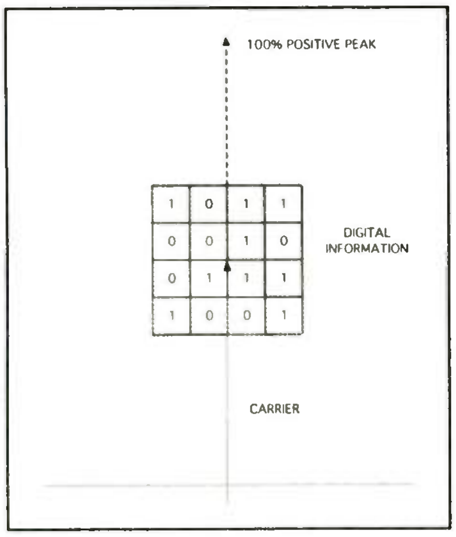

Fig. 8: AM + Digital IQ Modulation (Not to Scale)

It is most convenient to view impedances as they appear on a Smith Chart when analyzing sideband symmetry. This is because it is possible to see how they vary from the carrier-frequency impedance at sideband frequencies without regard to what the carrier-frequency impedance and reactance might happen to be.

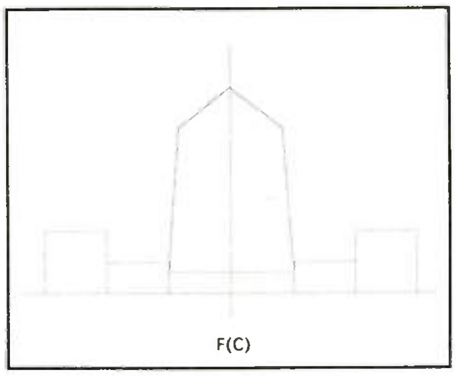

Fig. 9: AM IBOC Modulation Spectrum (Frequency Domain- Not to Scale)

Transmitters are generally designed to operate into 50-ohm loads. Solid-state transmitters sometimes have amplifiers that operate into much lower impedances, like 8 or 10 ohms. Confusion can be avoided if the analysis is done using impedances normalized to the carrier frequency value on a per-unit Smith Chart, having a value of 1.0 instead of an actual transmission-line characteristic impedance at its center (see Fig. 11).

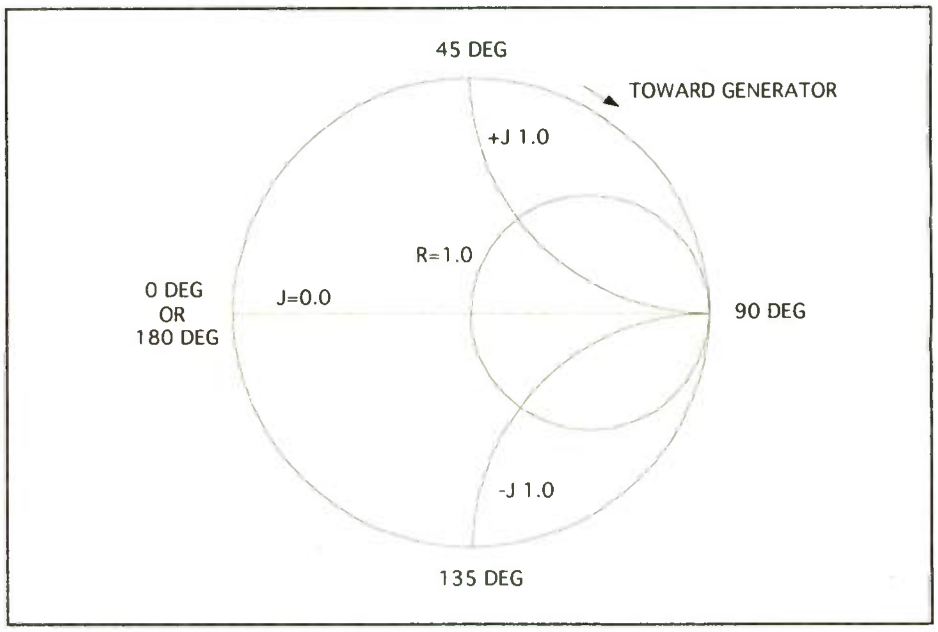

Fig. 11: Smith Chart Basics

The process of converting an impedance sweep for per-unit analysis is known as normalizing to the carrier-frequency impedance. Dividing the resistance at a sideband frequency by the carrier-frequency resistance gives the sideband per-unit normalized resistance if the carrier-frequency resistance is assigned a value of unity. Dividing the difference in reactance between a sideband frequency and the carrier frequency by the carrier resistance gives the sideband per-unit normalized reactance if the carrier-frequency reactance is assigned a value of zero (Fig. 12).

A symmetrical impedance sweep will appear on the Smith Chart with equal normalized resistances and equal normalized reactances of opposite sign for equal frequency deviations from the carrier frequency (Fig. 13).

For this symmetrical case, the sideband resistances are equal — but lower than the carrier resistance — at both sideband frequencies and the sideband reactances are symmetrical. If the final amplifier functions as a voltage source, higher power will be delivered to both sidebands than would be the case with a flat load (meaning longer sideband vectors) — resulting in pre-emphasis, but not harmonic distortion in the demodulated waveform (see Fig. 14; please note that for space purposes, several figures appear on pages 8-12).

In addition, the opposite reactances will cause both sidebands to be offset by equal-but-opposite angles relative to the carrier,

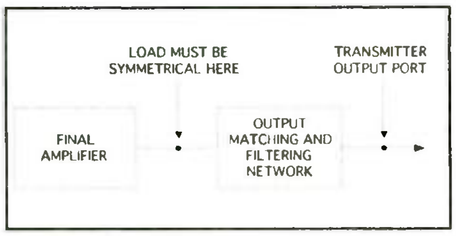

Fig. 10: AM Transmitter Output Stages

NORMALIZING PER-UNIT VALUES TO THE CARRIER IMPEDANCE

1. DIVIDE EACH SIDEBAND RESISTANCE BY THE CARRIER RESISTANCE
2. DIVIDE THE DIFFERENCE BETWEEN EACH SIDEBAND REACTANCE AND THE CARRIER REACTANCE BY THE CARRIER RESISTANCE

EXAMPLES

FREQUENCY	C.P. RESISTANCE	C.P. REACTANCE	PER-UNIT RESISTANCE	PER-UNIT REACTANCE
- 15 KHZ	45.0	-j 8.0	0.90	-j 0.16
CARRIER	50.0	0.0	1.00	j 0.00
+ 15 KHZ	57.0	+j 10.0	1.14	+j 0.20
-15 KHZ	45.0	-j 8.0	0.86	-j 0.09
CARRIER	52.5	-j 3.5	1.00	j 0.00
+ 15 KHZ	57.0	+ j 10.0	1.09	+j 0.12

Fig. 12: Normalizing Per-Unit Values to the Carrier Impedance

producing envelope delay, but not harmonic distortion, in the demodulated waveform. Both effects are perfectly acceptable for analog transmission, as long as the transmitter is not driven into nonlinearity to deliver the required sideband power. For digital transmission, the effects on the I-Q constellation will be more easily compensated than would be the case with a nonsymmetrical load. Importantly for IBOC transmission, the symmetry will prevent crosstalk between the digital and analog signals.

Digital-to-analog crosstalk may be the area of most concern for stations initially converting to IBOC transmission. Although a certain amount of hiss-type noise must be tolerated under the carrier of an AM station transmitting the IBOC signal, a very prominent "bacon frying" noise has been noted in certain cases. Thus far, experience indicates that this noise may be significantly decreased with load symmetry.

The following case of asymmetrical sideband load impedances shows how the two sideband vectors are altered differently in length and offset differently in angle due to the sideband loads presented to the final amplifier (Fig. 15). It is obvious that the resultant will follow an ellipse over a cycle of the analog modulating frequency rather than the straight line that is traced by a pure AM signal (Fig. 16).

Nonsymmetrical load impedance characteristics, in addition to causing crosstalk between the digital and analog signals in IBOC transmission, also complicate the error correction process for the digital I-Q constellation, leading to higher bit error rates that erode the reliability of the digital transmission channel. For the same reasons, the analog signal sidebands are altered in such a way as to cause harmonic distortion in the demodulated waveform.

AM IBOC technology is moving into the 'real world.' Measurements can be made to evaluate the important aspects of antenna performance and, in most cases where performance is found to be lacking, relatively simple measures can be taken to improve matters.

The asymmetrical load in this case can be made to perform identically to the symmetrical load that was considered in the first case by the addition of a -45 degree phase shifting network between the transmitter and the antenna (see Fig. 17). As can be seen from the notations on the blank Smith Chart of Fig. 11, this means rotating the asymmetrical transmitter final amplifier load impedance plot of Fig. 15 by one-quarter turn in a clockwise direction to become identical to the symmetrical plot

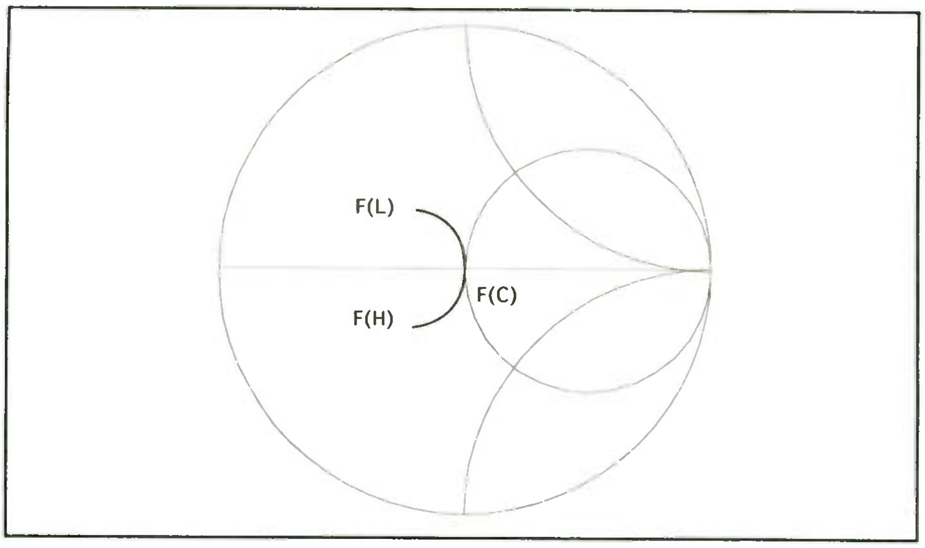

Fig. 13: Final Amplifier Load Impedance Symmetry

of Fig. 13.

Some loads may not be made symmetrical for the transmitter's final amplifier by simple phase rotation (see Fig. 18). One method to evaluate the potential for symmetry is to plot the normalized per-unit antenna input impedance on transparent overlay paper over a Smith Chart so that it can be rotated about the reference point to simulate the effects of added phase shift.

It is obvious by inspection that this load will require more than the addition of a phase-shifting network to have final amplifier load symmetry.

IBOC 'DESIRED CHARACTERISTICS'

Although there are no universally accepted criteria for evaluating the effects

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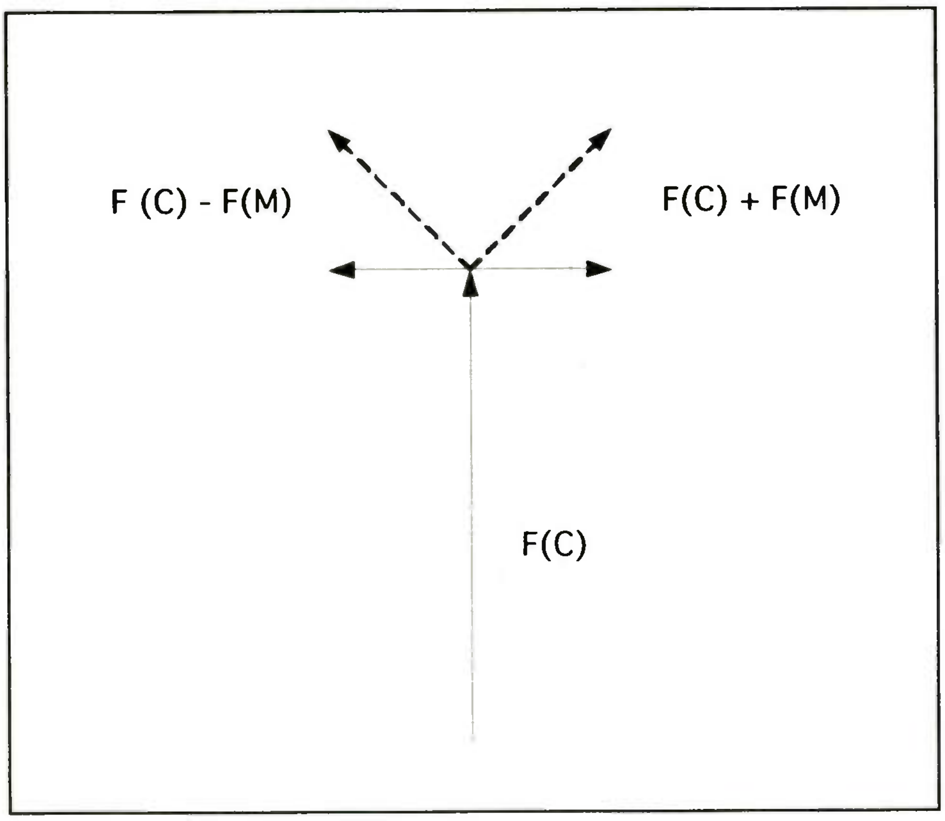

Fig. 14: Sideband Alteration from Symmetrical Load

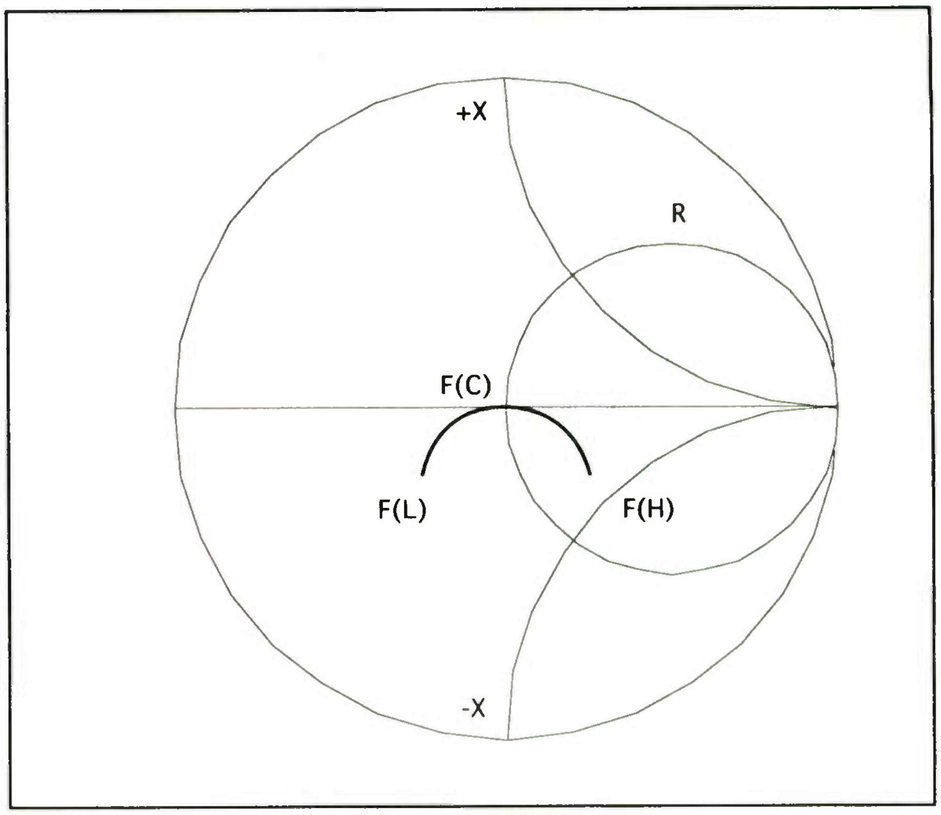

Fig. 15: Final Amplifier Load Impedance Asymmetry

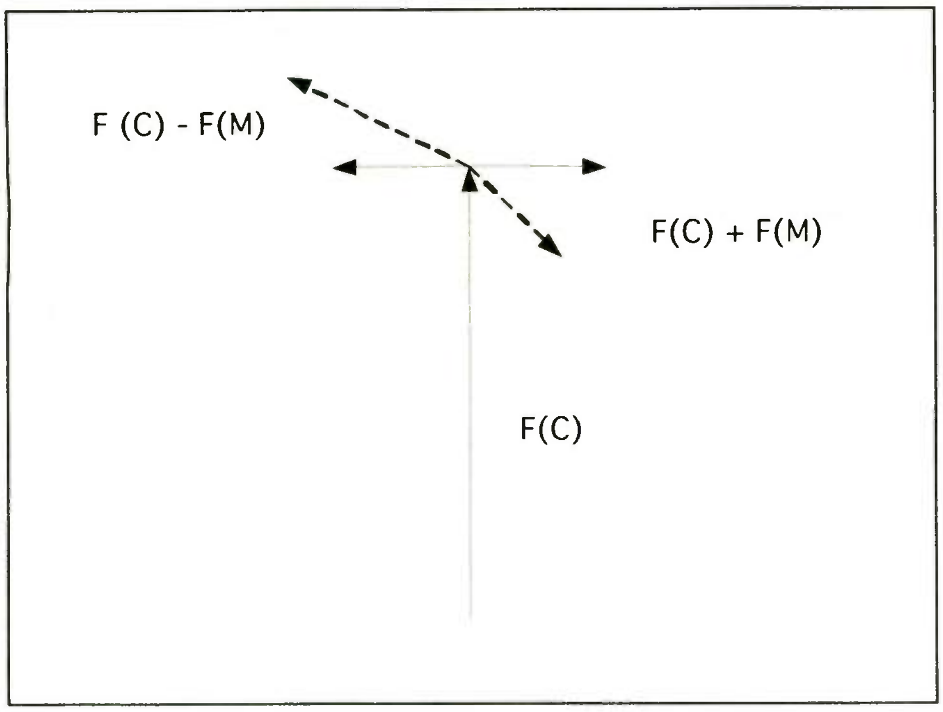

Fig. 16: Sideband Alteration from Asymmetrical Load

on digital transmission of finite-bandwidth antenna systems, the developer of the digital transmission system that is in use in the United States, Ibiquity Digital, has recommended the following "desired characteristics" for antenna systems that transmit with their system:

I. *Load Impedance Bandwidth*

a. +/- 5 kHz — symmetry of the load impedance presented to the final RF amplifier within the transmitter such that the VSWR calculated for one sideband impedance, when normalized to the complex conjugate of the corresponding sideband impedance on the other side of carrier frequency, does not exceed 1.035:1 (see Fig. 19).

b. +/- 10 kHz — the VSWR of the load impedance presented to the final RF amplifier within the transmitter should not exceed 1.20:1 when normalized to the carrier frequency impedance (Fig. 20).

c. +/- 15 kHz — the VSWR of the load impedance presented to the final RF amplifier within the transmitter should not exceed 1.40:1 when normalized to the carrier frequency impedance (Fig. 20).

II. *Far-Field Radiation Pattern Bandwidth*

a. +/- 15 kHz — Response within +/- 2 dB

b. +/- 15 kHz — Group delay constant within +/- 5.0 microseconds

These numbers are preliminary guidelines and are subject to change as more experience is gained with digital transmission. Thus far, experience has shown that the recommendations for sideband VSWR at 10 and 15 kHz from the carrier frequency are very conservative. Loads with higher sideband VSWRs have provided satisfactory performance both in the lab and in the field when the sideband impedances are made near-symmetrical about the carrier frequency (where perfect symmetry would have equal resistance and equal-but-opposite reactance excursions on either side of the carrier frequency).

As explained earlier, the symmetry of the sideband impedances at +/- 5 kHz of carrier frequency has proven to be important for minimizing digital-to-analog crosstalk. Such crosstalk causes noise that is sometimes described as "bacon frying" underneath the analog signal when digital transmission is underway.

NETWORK ANALYZER MEASUREMENTS

It has been found to be convenient to use a network analyzer to make measurements necessary for evaluating antenna system performance for digital transmission. A power-amplified system for making measurements with interference immunity has been developed for this purpose (see Figs. 21, 22 and 23).

Another configuration may be used to measure the phase and ratio-vs.-frequency characteristics of a directional antenna system's elements. The common point is driven by the directional coupler, which derives the signal for the analyzer's reference receiver input (Fig. 24). The antenna monitor sampling line of the reference tower is connected to the B receiver's input. The other sampling lines are switched to the A receiver's input and the analyzer is alternately set to measure the magnitude and phase relationships of the A and B inputs for each tower (Figs. 25 and 26).

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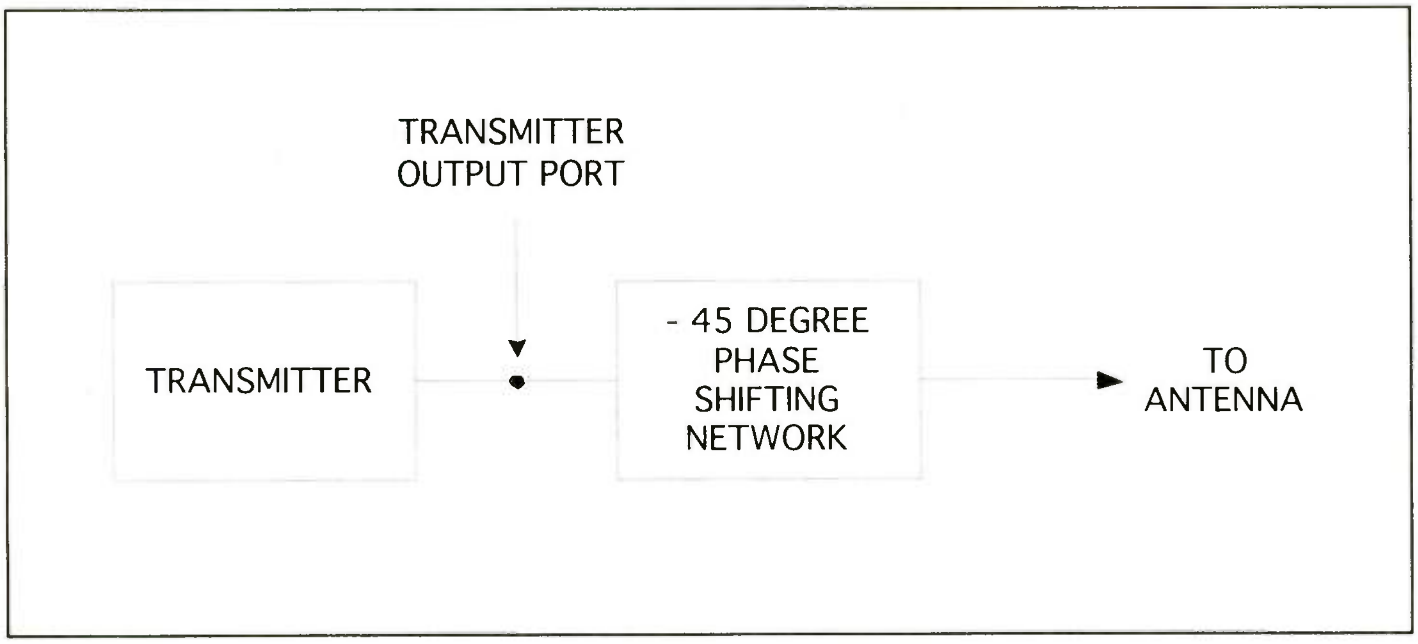

Fig. 17: Insertion of Phase Shift Network to Achieve Load Symmetry

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- VU + PPM level meter for the embedded audio
- Monitor DA converter and headphone/speaker amp

NEW! AL1 Acoustilyzer
Acoustics & Intelligibility analyzer

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

NTI Americas Inc

PO Box 231027
Tigard, Oregon 97281 USA
503-639-3737
www.nt-instruments.com
americas@nt-instruments.com

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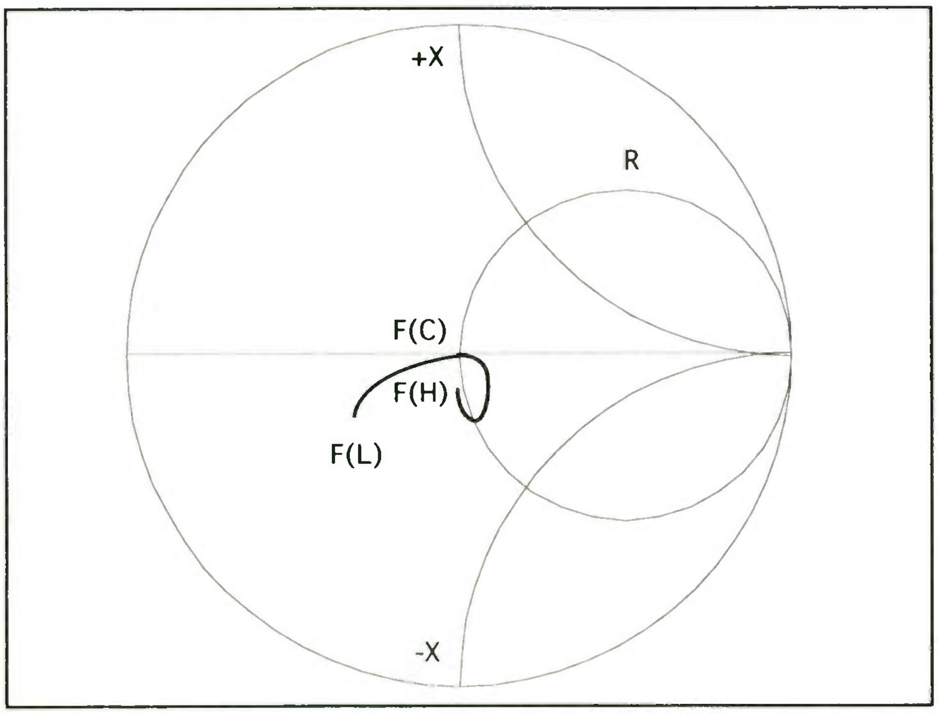


Fig. 18: Antenna Impedance Asymmetry

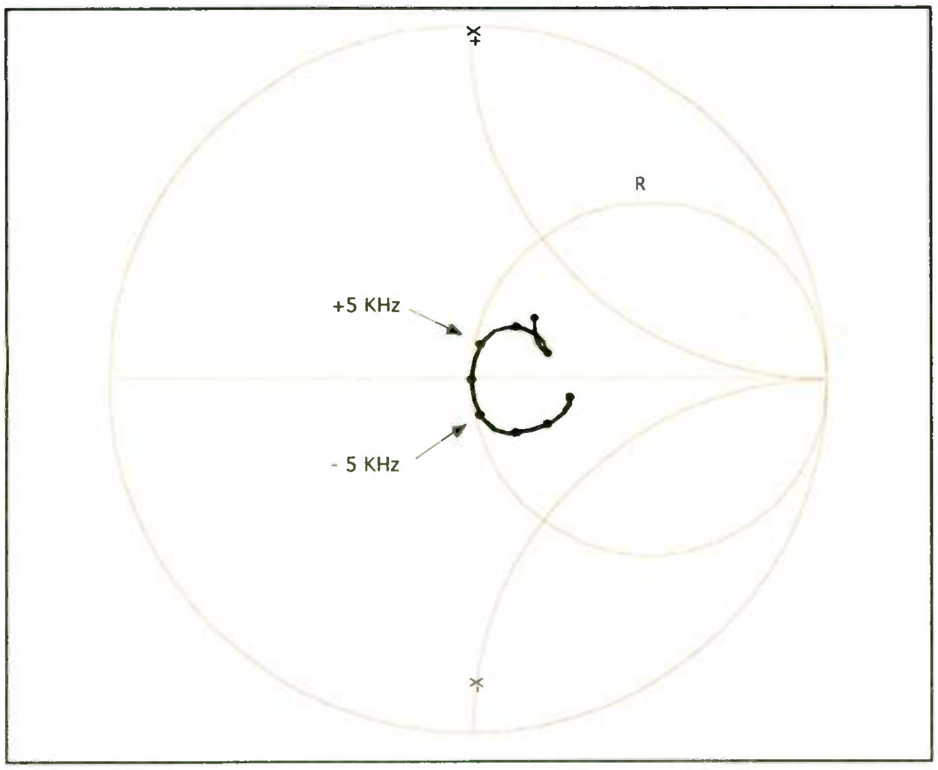


Fig. 19: Final Amplifier Symmetry at +/- 5 kHz

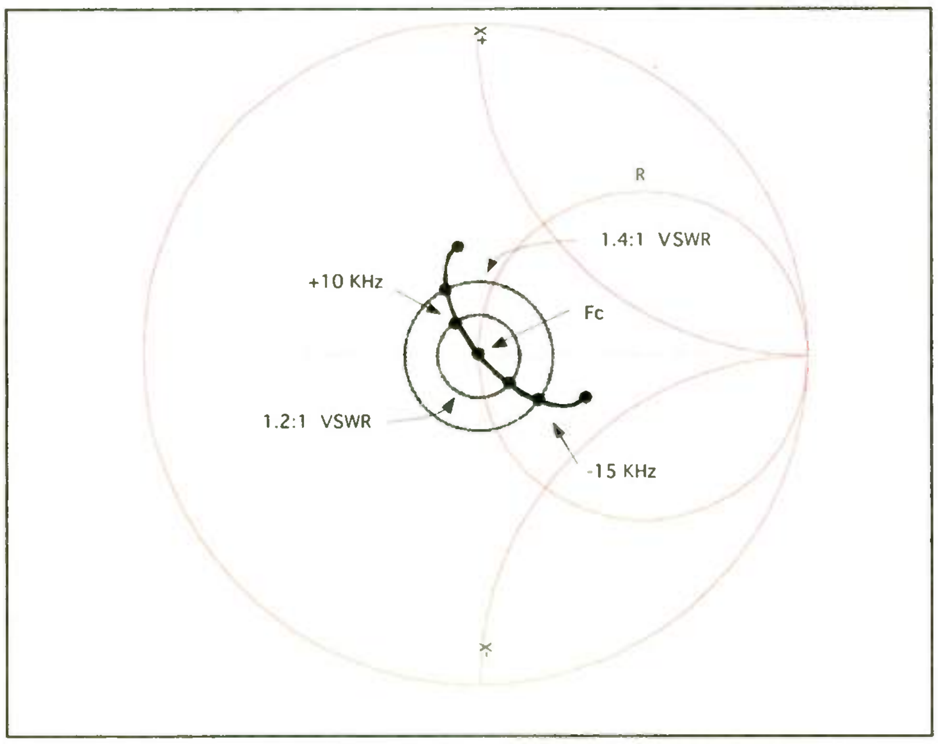


Fig. 20 Symmetry Requirements for IBOC

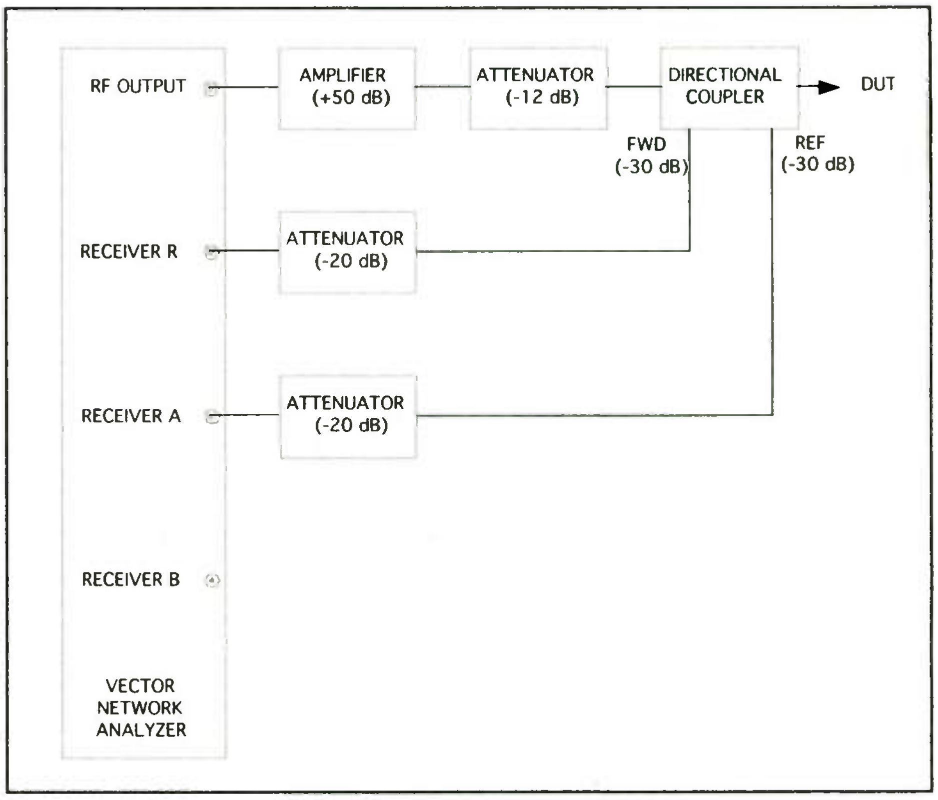


Fig. 21: Impedance Measurement System

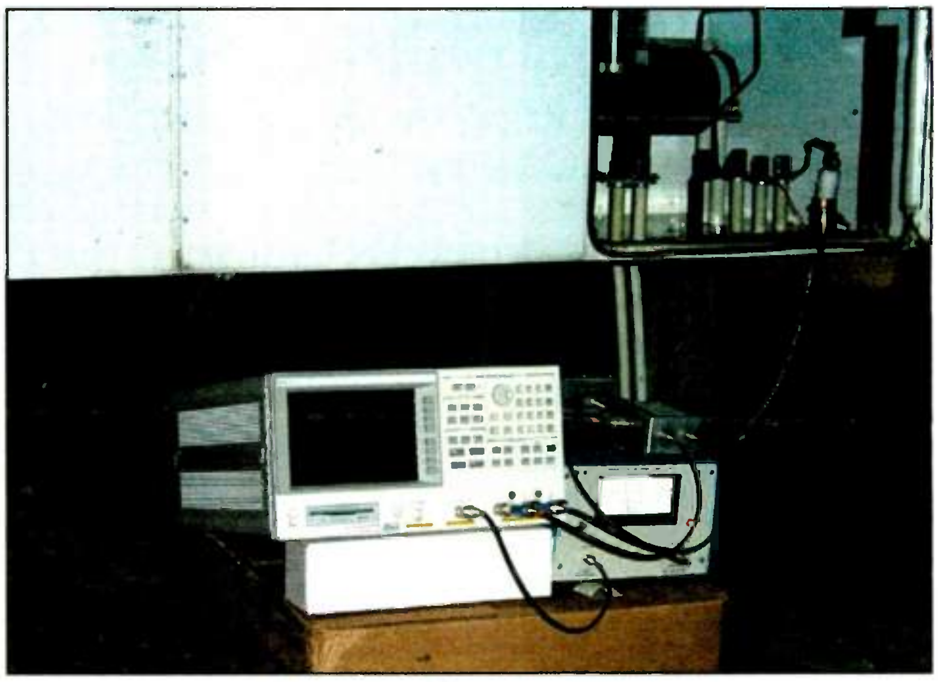
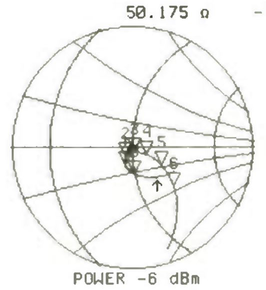


Fig. 22: Impedance Measurements at ATU Input

CH1 A/R FScl 500 mU 50.175 Ω -11.32 Ω 25.797 nF
545 kHz

Cor
Avg
16



IF BW 300 Hz POWER -6 dBm SWP 2.967 sec
START 535 kHz STOP 585 kHz

SELECT LETTER
SPACE
BACK SPACE
ERASE TITLE
DONE
STOR DEV [DISK]
CANCEL

N	SWP PARAM	VAL	AUX	EXTRA
0	545 kHz	50.175 Ω	-11.32 Ω	25.797 nF
1	550 kHz	47.155 Ω	-7.2862 Ω	39.715 nF
2	555 kHz	47.463 Ω	-3.8503 Ω	74.48 nF
3	560 kHz	50.826 Ω	-2.1784 Ω	130.46 nF
4	565 kHz	56.277 Ω	-3.3165 Ω	84.935 nF
5	570 kHz	63.194 Ω	-9.5886 Ω	29.12 nF
6	575 kHz	66.6 Ω	-22.454 Ω	12.327 nF

Fig. 23: Impedance Measurement Results

Information on how the tower ratios and phases vary with frequency may be used to evaluate pattern bandwidth. The measured antenna element parameters at carrier and sideband frequencies may be used to calculate the changes in far-field magnitude and phase that occur at different azimuths, using computer software such as the Mininec Broadcast Professional package. The far-field response may then be calculated directly from the magnitude excursions. The delay characteristics may be calculated from the phase characteristics at the azimuths of interest.

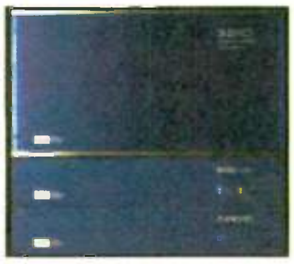
Experience to date indicates that the

AM ANTENNAS, PAGE 12

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

Fact is, SAS packs so much sophistication and capability into the depths of the new **Rubicon**™ control surface that even the most intensive major market programmer or board operator will swoon. Yet Rubicon is so intuitive, so comfortable, so easy to use, the weekend intern is sure to sound like a pro.

Here's why:



SAS Connected Digital Network™
Rubicon Control Surface
32KD Digital Audio Router
RIOLink™ Remote I/O

Frequently used controls are always right at the operator's fingertips. And for the power-user, the multi-function "dynamic control matrix" provides quick access to deeper capabilities. In other words, Rubicon has a bucket load of features for the simplest or most complex of broadcast-related tasks.

And should you think form to precede function, you'll find Rubicon's clean, easy-to-understand interface wrapped up within a custom-configured, drop-dead gorgeous frame.

Best of all, Rubicon is engineered by the brand synonymous with the finest in digital audio routing and network design. When it comes to quality and reliability, our name is all over it.

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AM ANTENNAS

CONTINUED FROM PAGE 10

methods described herein are worthwhile for evaluating existing antenna systems for HD-Radio compatibility and, in most cases, for determining what simple steps can be taken to optimize their performance. As more experience is gained in this area, knowledge about the optimum sideband-impedance characteristics for various HD Radio-compatible transmitter models can be expected to evolve, as well as the level of understanding with regard to real-world directional antenna pattern bandwidth and propagation issues.

Ultimately, it would be desirable to have some form of through-the-system performance evaluation method, based on a standard HD Radio test signal and a special receiver with diagnostic capabilities, which might even make some new form of precorrection for transmitter and antenna characteristics possible. Even then, work to improve a system's internal performance will be desired, as it can provide improvement in areas that do not lend themselves to precorrection. These methods should prove valuable both now and in the future.

Radio World Engineering Extra welcomes your comments and suggestions. Write to mlrwee@verizon.net. ■

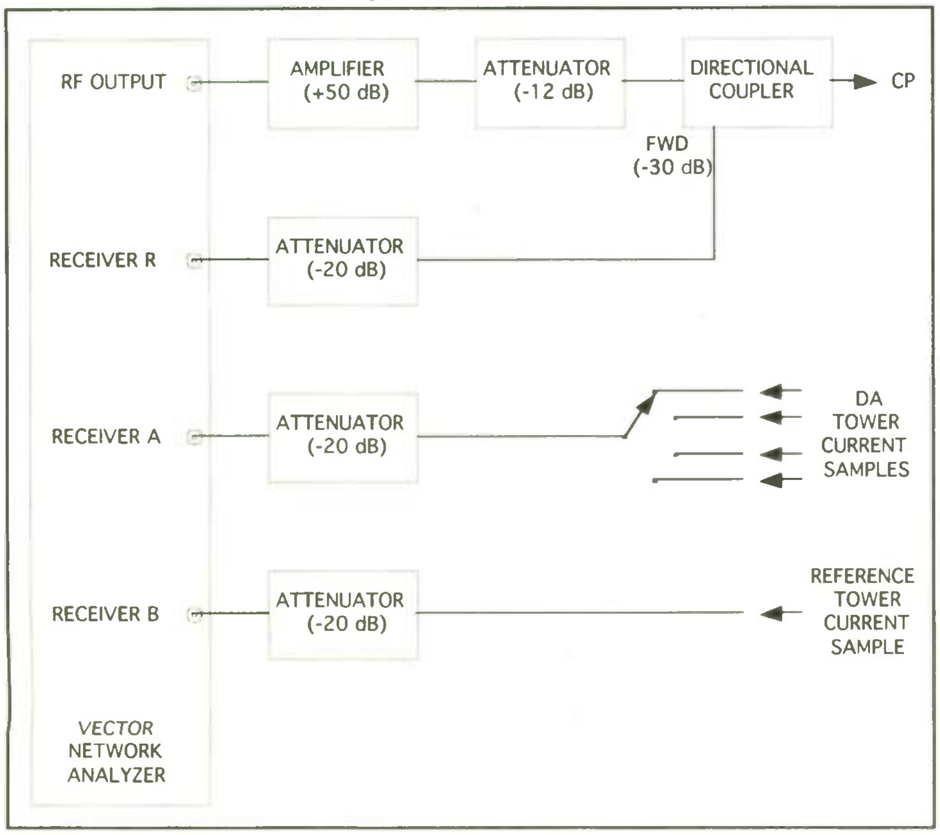

Fig. 24: DA Phase and Ratio Measurements

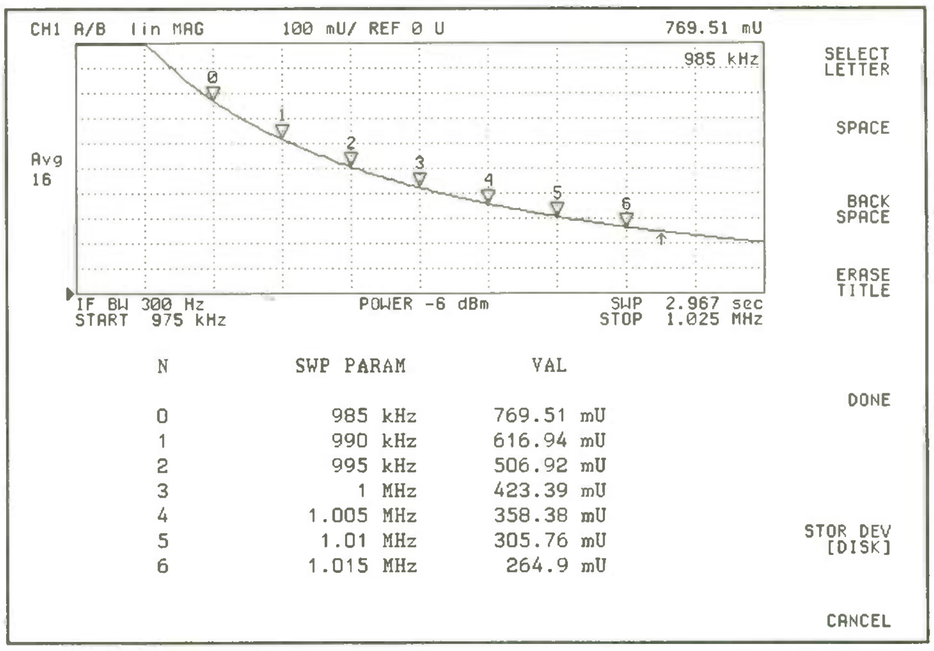

Fig. 25: Measured Radio Sweep

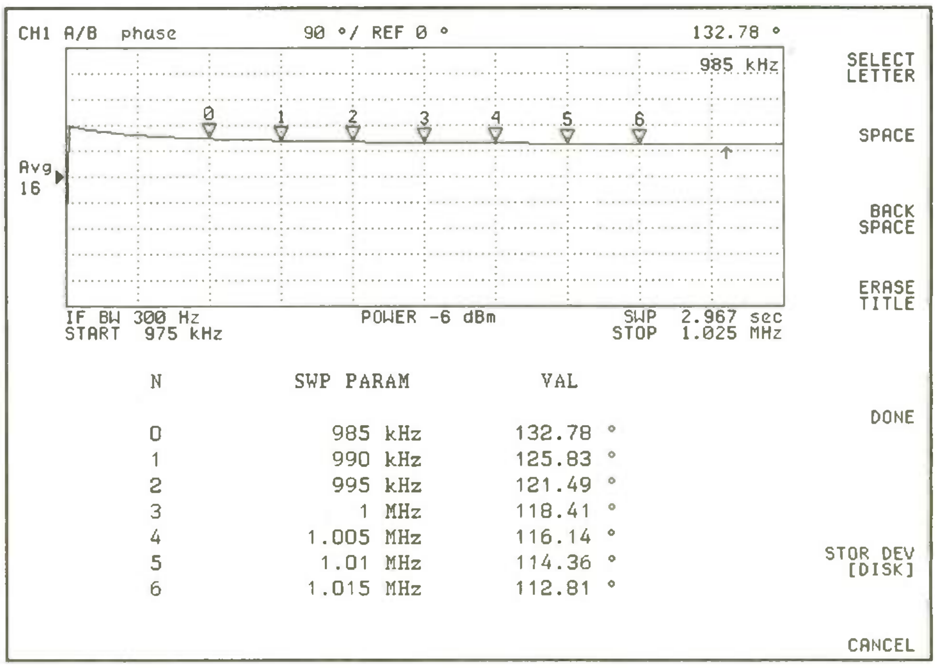

Fig. 26: Measured Phase Sweep

Schrag

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not only sound good, it can work efficiently and be interesting and attractive to the eye.

For this issue's Designer Interview, we spoke with Richard Schrag, principal of Russ Berger Design Group, an acoustic consulting firm.

What is the role of an acoustic consultant in a studio project?

Radio is all about creating sound. So much attention is paid to the equipment and microphone techniques, transmission algorithms and processing gear that it's easy to forget how critical the acoustical environment is in fashioning the right sound. That control room or studio or edit booth isn't just a place that houses the broadcast stuff; it's one of the tools that you use to "make radio."

That means radio stations have specialized requirements for acoustical performance, and that's where an acoustical consultant comes in. Architectural acoustics generally boils down to three basic concerns: sound isolation, noise and vibration control and room acoustics.

Why is it important to involve a consultant early in the project?

It's a common misconception that acoustics can be added on once the basic design is complete. We refer to that kind of thinking as "Band-Aids and perfume," because if you're dealing with the acoustical design too late in the process, you're pretty much limited to patching up mistakes and doing what you can to conceal the flaws that can't be fixed.

The acoustical problems we encounter most frequently arise when someone has already made some poor choices — a building with inadequate ceiling height, a site that's got environmental noise problems, or a lease space that's too close to an elevator or a noisy air handler.

Even if you're just considering the acoustics inside the room — which is what most people think of as the acoustical design — there are really only three things you have to work with: the volume of the space; the

Fig. 1: Studio Hallway of WPLN, Nashville Public Radio

basic dimensions and shape of the space; and the interior finishes.

If the first two are constrained by the inherent limitations of the building or a layout that didn't address the acoustical requirements, finishes alone won't always provide a satisfactory solution. On the other hand, if an acoustical consultant is involved at the beginning of a project, these same rooms can be designed to have appropriate sizes and shapes, making the rest of the design efforts much easier and much less expensive.

And it's not just about the acoustics. The technical spaces within a radio station have lots of unique requirements that generally don't apply to offices or other architectural projects. We have to simultaneously satisfy special requirements for adjacencies, sightlines, orientation of operators and equipment, connectivity, equipment ergonomics, lighting — all of this while making sure that the acoustics are right so it sounds good, and the interior design is right so it looks good.

That's why RBDG offers architecture and interior design under the same roof as acoustical consulting, and why we address a lot of issues that fall outside the realm of "normal" acoustical consulting.

What kinds of problems are specific to acoustics that can be avoided by proper design?

We often receive calls from stations that go something like this: "The real estate broker has already found a 'great' lease space. It's on the top floor of an existing office building, overlooking the airport runway and directly beneath the mechanical penthouse that serves all 20 floors. The developer has an architect who has already designed the space, and who thinks it would be 'fun' to have L-shaped control rooms arranged around a circular studio in the middle. Did I mention that the building air-conditioning shuts off at 6 p.m. every afternoon?"

This could be a real recipe for disaster. In this kind of situation, having an acoustical consultant involved in the early planning could have helped avoid a number of potential problems — environment noise, building mechanical noise, upper-floor structural vibration, poor room shapes and layout, and a variety of other functional limitations.

On the other end of the spectrum is a project like WPLN, Nashville Public Radio (Fig.1). They brought us on early in the process, before they had even decided on the scope of what they wanted to build. We helped them develop an architectural program, which defines how many rooms there will be, their sizes, their functions and the relationships among them. After also looking ahead to their likely future expansion, WPLN was able to go shopping for buildings that were well-suited to their specific needs. When they identified a potential candidate, we helped them evaluate its acoustical and architectural suitability. Ultimately they found a piece of land that allowed them to build from the ground up, and together we came up with a building that was tailored to exactly what they wanted the station to be.

Planning ahead like this not only avoids the pitfalls of bad site selection, but also leads to a more efficient use of space and more cost-effective construction, because you're doing a better job of matching the building to the radio station.

What are the different kinds of studio construction materials and how do they affect studio acoustics?

Well, that's a whole interview in itself, but the basics are pretty straightforward.

The acoustical tools we have on our belt include 1) absorbers — broadband types, selective low-frequency types, and materials that are effective only at high frequencies, 2) acoustical diffusers — in a variety of diffusion patterns and bandwidths, and 3) reflective surfaces — which include by default most windows, doors and any untreated wall surfaces.

The goal is to select the best assortment of these materials and deploy them in such a way that we create a good balance of the frequency response in the room, and at the same time control specific reflection paths and the way that sound propagates through the space.

It's a common misconception that acoustics can be added on once the basic design is complete.

Many of the treatments available commercially and marketed for their "acoustical" benefits are very effective at absorbing sound in the middle of the speech frequencies and above, but unfortunately that effectiveness starts to fall off rapidly below 500 Hz or so. The result can be a room that has had all of its acoustical life sucked out of the high end, but is still boomy thanks to limited low-frequency absorption. It's better to use less coverage of absorption, but select materials with a wider bandwidth, and instead incorporate adequate diffusion with enough hard surfaces left over to preserve the room's basic character. Carefully placing the materials where they will do the most good is often more successful than blanketing the entire room, and less expensive as well. (See Fig. 2.)

How would you approach a typical studio project with a live DJ running combo?

A combo-style control room by itself may be the easiest "acoustically critical" type of space to design for a radio station. Making a room sound good for monitoring and also for a voice talent on mic is pretty straightforward. Now, when you add in the possibility

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Products & Services SHOWCASE

Schrag

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of interviewing guests in that same control room, it gets a little trickier — not because of the room acoustics, but because it's difficult to situate the operator and guests where they can have a comfortable conversation, and still have the operator in a good audio monitoring environment. That's why you see control rooms where the monitor speakers are up near the ceiling or where the guests have to peek between the video monitors and the copy stand to catch a glimpse of the host. Unfortunately, good acoustics are often sacrificed when there's an apparent conflict with the ergonomics or the equipment — but it doesn't have to be that way.

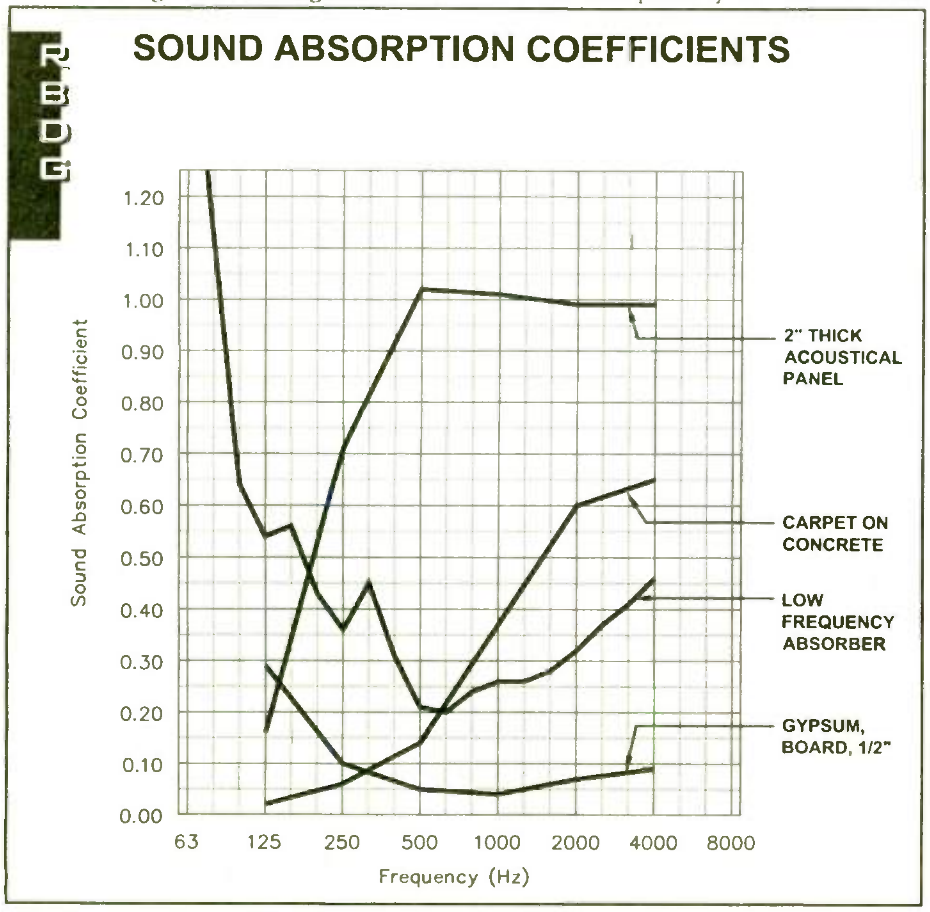

Fig. 2: Material Absorption Values

The goals for a combo control room are much like those for any space designed for audio monitoring: You want to control the short reflection paths between the sources and listeners, maintain left-right symmetry, provide sufficient diffusion and hard surfaces to keep some life in the room and make it sound like a real space, place the speakers in the right relationship to the listeners and the room boundaries, properly orient the surround speakers. …

By the way, you are monitoring in surround, right? Because there are people out there who will be, and you need to know whether they're hearing what you intend.

How about for a talk studio with live guests?

A talk studio is one of those spaces where the acoustical design is mostly a matter of establishing the right balance. Too reverberant, and what comes across on the air is a sense that the people are talking to each other in a stairwell or a locker room. Too dead, and the voices become disembodied with no sense of the space around them. Too boomy, and articulation suffers. Too bright, and there's no sense of warmth.

Having a pleasant-sounding studio where the acoustics have a controlled character without obvious peculiarities not only affects the sound that the mic captures, it also affects the people who are speaking. Both the host and the guests will be more at ease when they are in a room that has a natural, neutral sound (see Fig. 3).

What about for a live performance room for small bands or ensembles?

Performance studios for live radio have the same need for balance as talk studios, but are even more complicated. This kind of studio really has to be tailored to the station and to the specific types of performances and events that are likely to be produced.

Is the room primarily for music or for the spoken word, such as town hall meetings or political debates? Are there usually acoustic instruments — where the musicians will be looking for the room to give them some natural reinforcement and a return of their own sound? Or are there usually amplified instruments — where the challenge might be to keep from overloading the acoustic volume or to control the low end? Will there be an audience? Will you be trying to achieve some degree of isolation between mics on different instruments for mixing purposes, or taking a more "set it and forget it" approach?

We would use the station's answers to these questions to determine the best overall reverberant characteristic of the space, and to help place absorptive and diffusive treatments where they will enhance the sound behavior in each part of the room. That would include making sure there's good acoustical communication between musicians, and between the performers and the audience, if the room will be used that way.

Fig. 3: WBUR Talk Studio

Planning ahead leads to more efficient use of space and more cost-effective construction.

The real key to a performance studio is to have enough volume to begin with. If the room's too small or has a nine-foot ceiling, it will generally sound that way. Our ears and brains do a remarkable job of judging the size of a space from acoustical clues — deciphering from the arrival times of reflections and their frequency content just how far away the floor, walls and ceiling are from the microphone. For the most part, if the volume doesn't exist, there's not much you can do to "fake it." But when the room is an appropriate size and shape, the interior finishes become a lot easier to design and, actually, a lot less critical to the kind of results you can achieve (see Fig. 4).

Fig. 4: KUOW Performance Studio

The terms STC and NC are used to describe acoustic specifications, but are often misunderstood. Explain how each is important when designing a studio complex.

STC stands for "Sound Transmission Class," and it deals with sound isolation. It is a way of describing, with a single number, the ability of a construction element — a door, a window, a wall or an entire room assembly — to stop airborne sound from getting in or getting out. You determine it by making noise on one side of the element and measuring the sound levels in that space as well as the space on the other side. The difference between the two noise levels is the noise reduction. Once you correct for the physical conditions — the size of the partition between the two spaces, the volume and amount of sound absorption — the noise reduction can be normalized to a sound transmission loss value. Do that for the 16 third-octave frequency bands that range from 125 to 4000 Hz, then compare the data to a standardized contour using a complicated set of rules about how far the individual data points can be below the curve, and eventually a single number drops out.

The bad news is that STC is a metric best suited for evaluating speech, not music; it doesn't even consider the noise below 125 Hz, and it is heavily weighted to the frequencies that occur in speech. Because it's a single number, two different materials or assemblies can have the same STC rating but perform very differently — one might actually be much better at stopping low-frequency sound, for example.

The other rating, NC or "Noise Criterion," tells you something about the amount of noise that exists within a given space. In this case the noise is measured at full-octave bands ranging from 63 Hz to 8000 Hz, then compared to a standardized set of curves to determine a single-number rating.

The NC rating system has some shortcomings. It ignores extreme low-frequency noise, which can be disturbing in its own right; and it fails to take into account the uniformity or balance of the noise with respect to frequency. Other metrics have been developed that are more descriptive of some of the characteristics of the noise or correlate better to human responses, but NC ratings have a foothold in the testing and literature of thousands of air delivery devices from hundreds of manufacturers, so it continues to be the most widely used rating system.

No matter what metric you use, no single number can tell the whole story with respect to an acceptable background noise level in a radio environment. NC ratings are really meant for evaluating HVAC (heating, ventilating and air-conditioning) noise, so when they're used to discuss extraneous noise from other sources, like the office next door, they may not be a good predictor of just how quiet is quiet enough. Even when the noise in a room meets the specified NC rating, it can still be too loud if it contains strong tonal components, is intermittent over time or can easily be localized to a specific source.

How important is HVAC design in achieving studio specifications? How can this affect the final studio performance?

Along with noise transmitted from adjacent rooms or outside the building, the most prevalent source of background noise is the HVAC systems. You can't avoid bringing conditioned air to the room, and the mechanical equipment that moves that air through the ducts as well as the air movement itself invariably generates some amount of noise.

What you can do, however, is minimize

SCHRAG, PAGE 25

The world's best* IP-Audi

*Okay, you caught us. It's also the world's *only* IP-

Everybody needs to share audio. Sometimes just a few signals — sometimes a few hundred. Across the hall, between floors, now and then across campus. Routing switchers are a convenient way to manage and share your audio, but will your GM really let you buy a router that costs more than his dream car? Unlikely.

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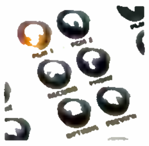
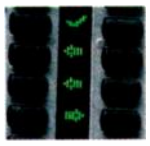
Axia SmartSurface provides the perfect blend of flexible features and intuitive control. Easy to learn and easy to use, it's tailor-made for talent-intensive formats.

Programmable soft keys and recording device transport control buttons give instant control of all audio functions.

Ergonomically designed channel start and stop buttons, with guards that prevent accidental activation.



Each channel's main, special-purpose, phone and preview assignments are quickly accessible. Automatic mix-minus for each fader!



Available Telos Console Director panel with Status Symbols® provides easy, intuitive control of phone-based segments.

"This sounds expensive." Just the opposite, really. Axia saves money by eliminating distribution amps, line selectors, sound cards, patch bays, multi-pair cables, and tons of discrete wiring — not to mention the installation and maintenance time you'll recover. And those are just side benefits: our hardware is about half the cost of those big mainframe routers. That's right... *half*.. Once you experience the benefits of networked audio, you will never want to go back.

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Routers are OK... but a network is so much more modern. With Axia, your ins and outs are next to the audio, where they belong. No frame, no cards, no sweat.

Put an Axia Microphone Node next to your mics and send preamplified audio anywhere you need it, over Ethernet — with no line loss or signal degradation.

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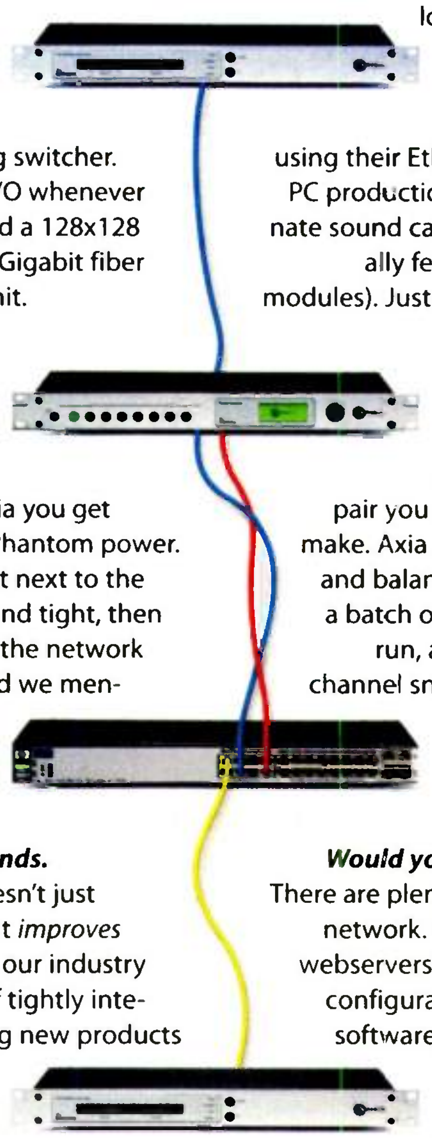
Scalable, flexible, reliable... pick any three.
An expensive proprietary router isn't practical for smaller facilities. In fact, it doesn't scale all that well for larger ones. Here's where an expandable network really shines. Connect eight Axia 8x8 Audio Nodes using Cat-6 cable and an Ethernet switch, and you've got a 64x64 routing switcher. And you can easily add more I/O whenever and wherever you need it. Build a 128x128 system... or 1024x1024... use a Gigabit fiber backbone and the sky's the limit.

Put your preamps where your mics are.
Most mainframe routers have no mic inputs, so you need to buy preamps. With Axia you get ultra-low-noise preamps with Phantom power. Put a node in each studio, right next to the mics, to keep mic cables nice and tight, then send multiple mic channels to the network on a single Cat-6 cable. And did we mention that each Mic Node has eight stereo line outputs for headphones? Nice bonus.

With a little help from our friends.
A networked audio system doesn't just replace a traditional router — it *improves* upon it. Already, companies in our industry are realizing the advantages of tightly integrated systems, and are making new products that reap those benefits.
Working with our partners, Axia Audio is bringing new thinking and ideas to audio distribution, machine control, Program Associated Data (PAD), and even wiring convenience.



Are you still using PC sound cards?
Even the best sound cards are compromised by PC noise, inconvenient output connectors, poor headroom, and other gremlins. Instead, load the Axia IP-Audio Driver for Windows® on your workstations and connect *directly* to the Axia audio network using their Ethernet ports. Not only will your PC productions sound fantastic, you'll eliminate sound cards and the hardware they usually feed (like router or console input modules). Just think of all the cash you'll save.

Put your snake on a diet.
Nobody loves cable snakes. Besides soldering a jillion connectors, just try finding the pair you want when there's a change to make. Axia Audio Nodes come in AES/EBU and balanced stereo analog flavors. Put a batch of Nodes on each end of a Cat-6 run, and BAM! a bi-directional multi-channel snake. Use media converters and a fiber link for extra-long runs between studios — or between buildings.

Would you like some control with that?
There are plenty of ways to control your Axia network. For instance, you'll find built-in webservers on all Axia equipment for easy configuration via browser. PathfinderPC® software for Windows gives you central control of every audio path in your plant. Router Selector nodes allow quick local source selection, and intelligent studio control surfaces let talent easily access and mix any source in your networked facility.

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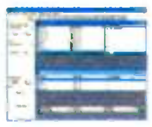
There's a better way to get audio out of your PC. No more 1/8" connectors – with Axia your digital audio stays clean and pristine.

An Axia digital audio snake can carry hundreds of channels of digital audio on one skinny CAT-6 cable. We know you're not going to miss soldering all that multi-pair...



Control freaks, rejoice: PathFinderPC software for Windows® gives you systemwide control of all routing functions with just a click of your mouse.

Advantages of 4M Modulation

Broadcast Electronics Describes New Modulation Scheme for AM

by Richard Hinkle and Jerry Westberg

The authors are vice president of engineering and senior engineer for Broadcast Electronics.

The digital age has arrived for the radio broadcasting industry, with the adoption of such standards as HD Radio and Digital Radio Mondiale. These modulation standards impose new demands on both FM and AM transmitters, requiring more linear reproduction than current analog FM and AM transmission techniques can provide.

Legacy modulation designs in AM products have been, and continue to be, of major interest during development, testing and implementation of these emerging digital transmission standards. In particular, both pulse width modulation and digital modulation have been the subject of discussions and papers focusing on their spectral performance and efficiency in the face of new digital transmission schemes.

This paper presents the latest innovation in AM transmitter technology, 4M Modulation, developed by Broadcast Electronics, and the benefits that can be derived from this unique design.

4M Modulation provides many advantages in both spectral performance and efficiency for digital and analog transmission. The 4MX 50, Broadcast Electronics' first transmitter using this modulation scheme, is a fraction of the footprint and weight of comparable models due to the component-efficiency of 4M Modulation and associated technologies. It is the only 50 kW AM transmitter that, when tipped on its side, can be moved through a standard doorway.

In addition to these key performance criteria, 4M Modulation provides the advantages discussed here.

EXISTING MODULATION SCHEMES

To understand 4M Modulation and its advantages, it is important to review existing modulation schemes and understand the limitations and key features of each as they relate to standard analog AM, HD Radio and DRM. The two dominant modulation designs used today are pulse width modulation (PWM) and digital.

PWM OVERVIEW

Pulse width modulation is one of the most common modulation schemes and is the modulation design used in all AM transmitters produced by Broadcast Electronics before the advent of the 4MX product line. A block diagram of a PWM transmission system is shown in Fig. 1. The PWM transmitter design encompasses six major blocks:

1. audio low-pass filter
2. PWM generator
3. power supply
4. low-pass filte
5. RF amplifier
6. bandpass filter

Audio Low-Pass Filter — The audio input filter limits the higher-frequency audio of the incoming signal. The high-end limiting usually occurs between 10 kHz and 15 kHz. The filter also prevents unwanted high-frequency noise that might be injected into the system.

PWM Generator — The PWM generator is a comparator whose two inputs are the filtered audio input and a triangle wave that is generated internally. The triangle waveform has a frequency usually between 70 kHz and 150 kHz, known as the PWM frequency. These two signals are compared and produce an output signal whose duty cycle is proportional to the audio level (i.e., when audio amplitude is high, the duty cycle increases, and as the amplitude is lowered, the duty cycle decreases). This output, known as a pulse train, is shown in Fig. 2. The PWM generator also produces an amplified pulse train based on the DC voltage from the bulk power supply.

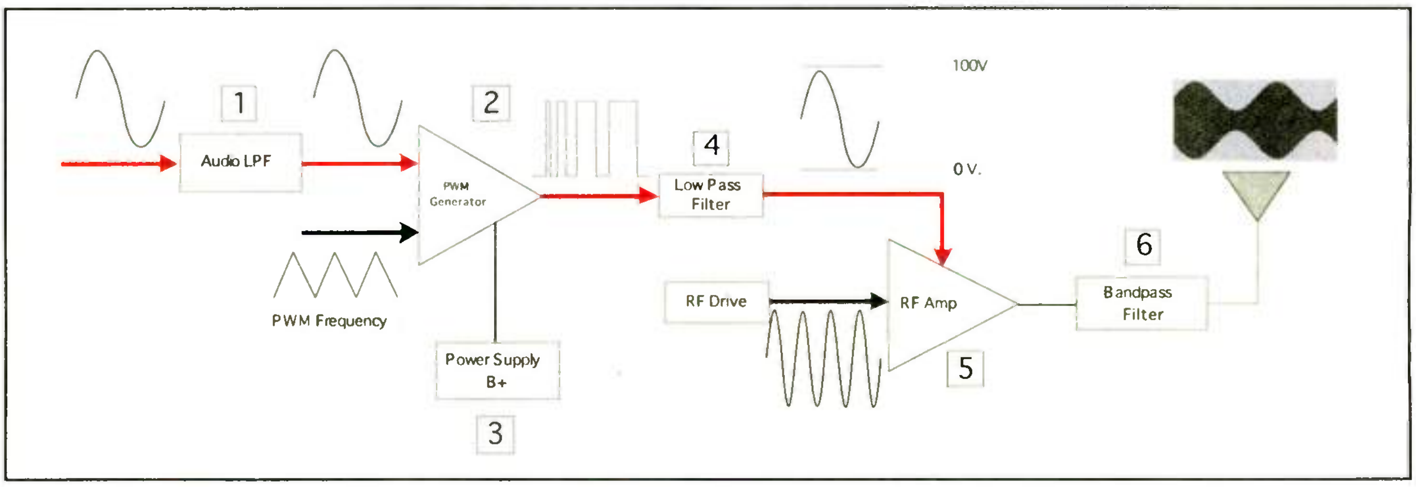


Fig. 1: PWM Transmitter Diagram

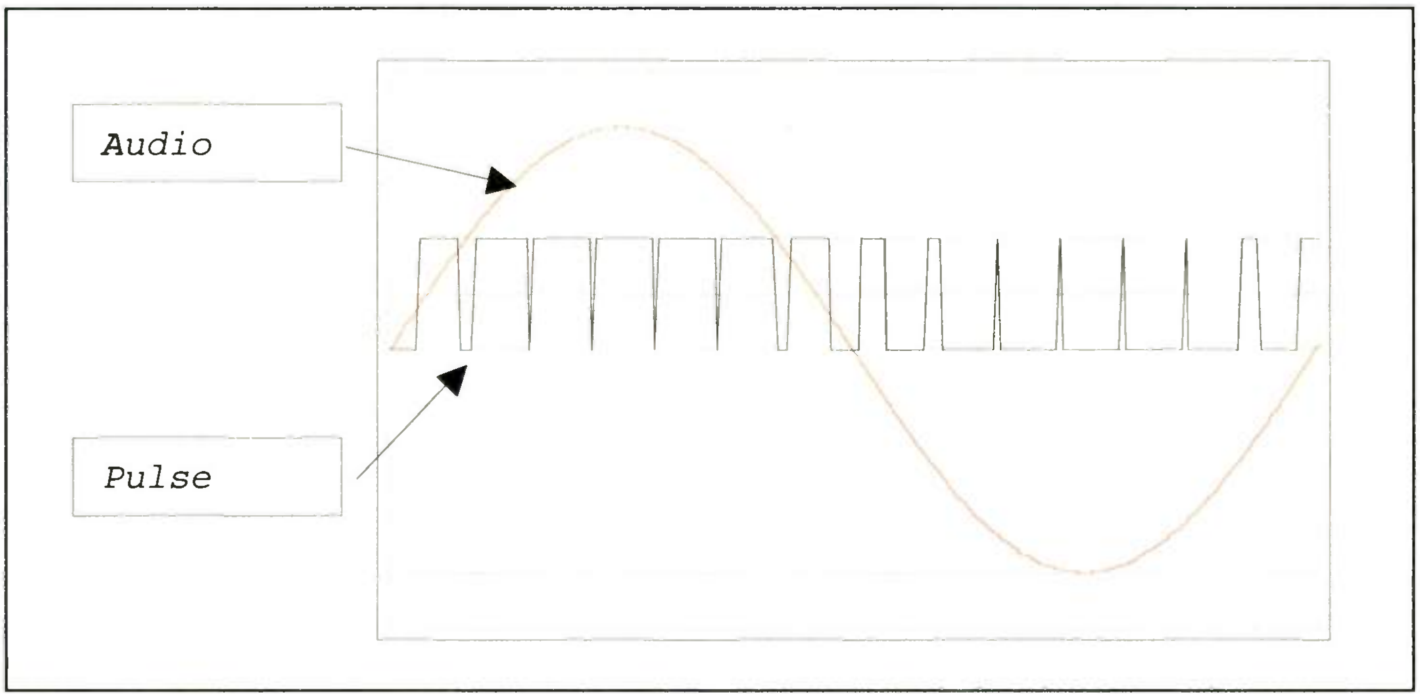


Fig. 2: PWM Generator Pulse Train

Power Supply — The power supply takes energy from the AC mains and converts it to the desired DC voltage. This voltage, typically around 100 V, is one of the inputs to the PWM modulator.

Low-Pass Filter (LPF) — The PWM generator output is then passed through a low-pass filter to regain the original audio signal at an amplified voltage. The DC output of this LPF has a voltage proportional to the audio input. The LPF must be capable of filtering the PWM switching frequency and harmonics to levels that comply with the regulatory emission mask requirements.

RF Amplifier — The RF amplifier switches at the carrier frequency, which is amplified by the voltage coming from the modulator LPF. This yields an RF signal whose amplitude varies in proportion to the original input audio signal.

Bandpass Filter — The bandpass filter then attenuates any harmonics of the carrier frequency to an acceptable level.

Effects on HD Radio

For HD Radio, the input filter is removed or modified to allow the higher-frequency amplitude signal to pass unobstructed into the PWM generator. The HD Radio audio bandwidth is 15 kHz. Therefore, this filter must be sufficiently wide and have a bandpass characteristic that limits any phase and amplitude distortions caused by the filtering. Widening of this filter can cause problems for the transmitter because it now passes higher-frequency components other than the audio.

The phase information for the HD Radio signal is injected directly into the exciter and sent to the RF amplifiers. The internal oscillator is disabled (usually by a relay), and the carrier signal with HD Radio phase information replaces it. The injection of the phase information in this manner poses no bandwidth limitations, since the RF amplifiers and the bandpass filter have more than acceptable bandwidth to pass HD Radio phase information.

However, the frequency of the PWM is critical to HD Radio performance. The PWM low-pass filter must limit the spurious products generated by the modulator. As the switching frequency gets lower, it becomes increasingly difficult to reduce these spurious emissions and still maintain enough bandwidth to pass the higher-frequency HD Radio signal without introducing phase and amplitude distortion. If the switching frequency is high, the LPF can be designed to minimize these distortions and still maintain acceptable spurious levels. It is important to note that increasing the PWM switching frequency has a negative impact on overall transmitter efficiency because of switching losses in the modulator. As the switching frequency increases, the losses in the field effect transistors in the modulator increase, thus lowering the overall efficiency of the system.

If the phase and amplitude of the signal passing through the PWM filter are distorted, then the resulting output signal will have intermodulation distortion products. These yield a spectrum that may not meet the required spectral emissions mask. Also, the receiver may not decode the resulting broadcast signal properly (i.e., with high bit-error rates).

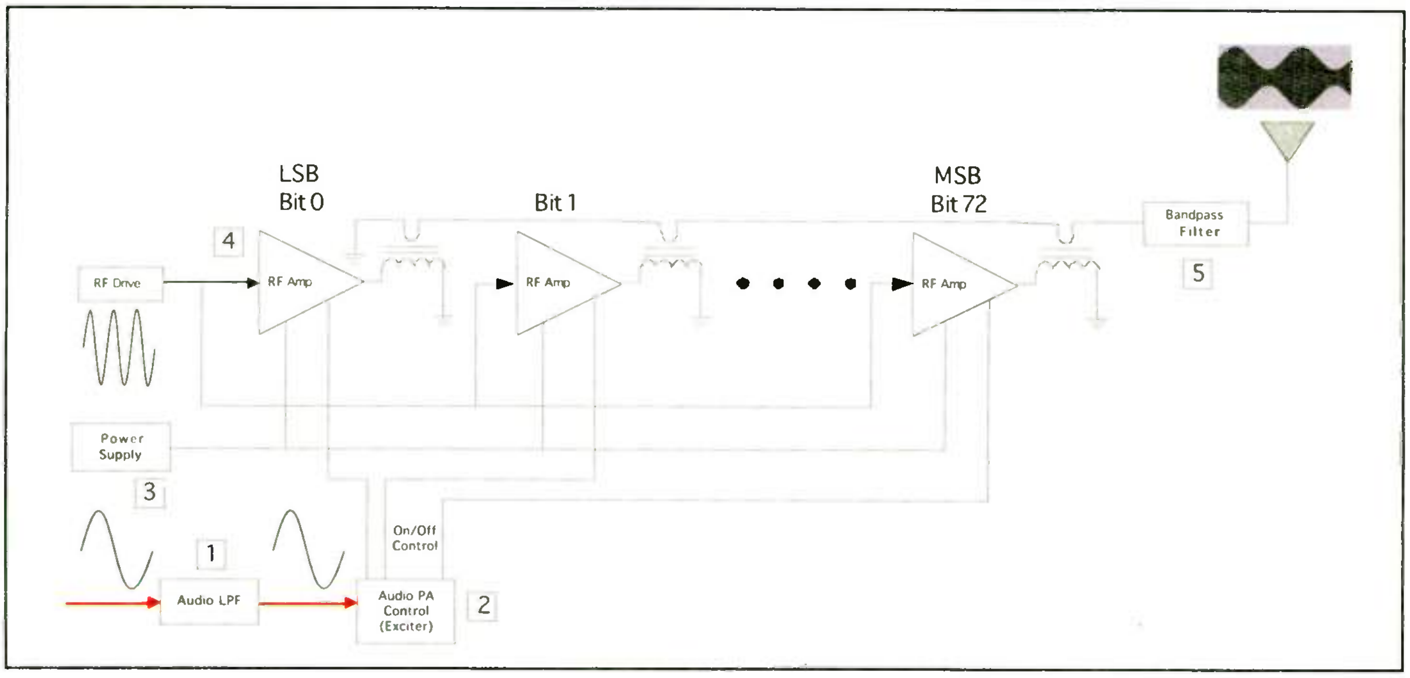


Fig. 3: Digital Modulation Transmitter Diagram

DIGITAL OVERVIEW

Digital modulation technology has been incorporated at high-power levels for many years. This technology works like a high-power digital-to-analog converter (DAC). As the audio input level increases, more modules are turned on, and, conversely, the number of modules turned on decreases as the audio input level decreases. The digital AM system (see Fig. 3) comprises five major blocks:

1. audio low-pass filter
2. audio/power-amplifier controller
3. power supply
4. series of RF amplifiers
5. bandpass filter

Audio Low-Pass Filter — The audio input filter provides the same function as it does for the PWM transmitter: it removes any unwanted high-frequency audio and noise before sending the audio to be amplified.

Audio/Power-Amplifier (PA) Controller — The audio/PA controller receives the audio signal and determines the number of amplifiers that are to be turned on to mirror the audio signal at the output of the bandpass filter. It also determines the gain of the variable gain amplifier used to eliminate any quantization noise resulting from the finite number of amplifier steps.

Power Supply — The power supply creates a voltage from the AC mains and feeds the RF amplifier directly. This is different from the PWM transmitter, where this voltage is used by the modulator stage.

RF Amplifiers — The digital system incorporates a series of amplifiers that can be turned on or off at the command

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4M MODULATION

CONTINUED FROM PAGE 19

of the audio/PA controller. As the amplitude of the audio rises, more amplifiers are turned on. As the amplitude of the audio lowers, amplifiers are turned off. This modulation scheme works well at power levels when there are enough modules to reproduce the desired audio waveform accurately. However, performance degrades at lower power levels due to limitations in the number of modules available to reproduce the audio signal. In addition, as amplifiers are turned on and off, quantization noise occurs, just as it does with a DAC — an additional cause of performance degradation.

Bandpass Filter — The bandpass filter provides much the same function as the filter used in the PWM design: it reduces the level of harmonics of the carrier to an acceptable level.

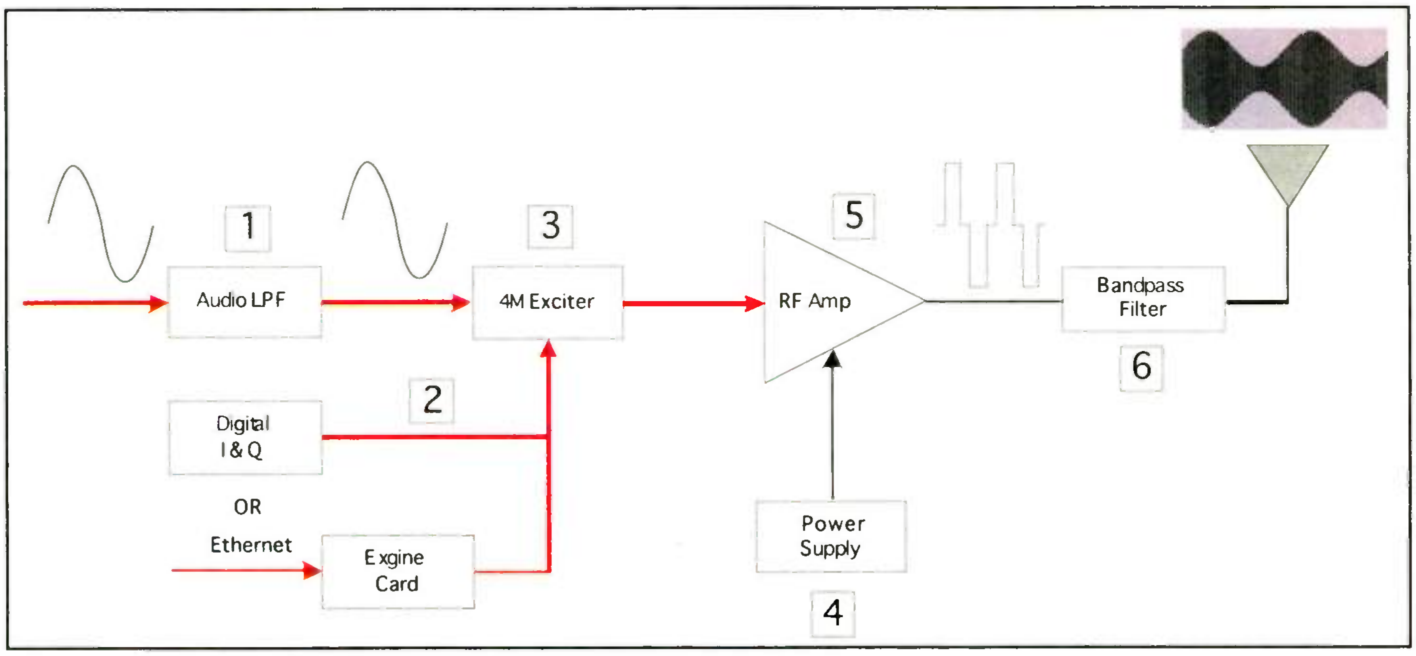

Fig. 4: 4MX Transmitter Diagram

Effects on HD Radio

The digital modulation scheme eliminates the need for the PWM generator, modulator and the associated LPF. Eliminating these parts causes an efficiency advantage over a PWM transmitter, with efficiencies in the high 80s. In addition to the efficiency advantage, eliminating the modulator and associated LPF is important to HD Radio broadcast because the LPF is the major bandwidth-limiting portion of the PWM transmitter system. In its place is a controller system that determines, from the audio input signal, which RF amplifiers are turned on. This modulation scheme is not bandwidth-limited.

For HD Radio, the input filter is removed or modified to allow the higher-frequency amplitude signal to pass unobstructed. This is the same as with the PWM transmitter.

Like the PWM transmitter system, the digital transmitter injects the phase information of the HD Radio signal directly into the exciter. The internal oscillator is disabled with a relay circuit, and the carrier signal with HD Radio phase information replaces it. Injection of phase information in this manner poses no bandwidth limitations. Because the system is not band-limited, the resulting spectrum fits well within Ibiquity's recommended mask.

4M MODULATION

The background information on the modulation techniques that have been incorporated in AM transmitters over the years assists in illustrating the distinctive technologies in 4MX transmitters and 4M Modulation. The 4MX transmitter provides advantages in efficiency, power range, analog performance and performance in the digital transmission era (see Fig. 4).

The 4MX transmitter consists of six major blocks:

1. audio low-pass filter for analog
2. I and Q or Ethernet data input for HD Radio operation
3. exciter
4. power supply
5. power amplifier
6. bandpass filter

Input Low-Pass Filter — The audio input filter for the analog signal limits the bandwidth for analog operation by protecting against unwanted high-frequency signals and noise.

HD Radio Data Input — The I and Q or Ethernet data input provides a path for the HD Radio signals to pass directly into the processor in front of the exciter to be summed with the analog signal. It is important to note that the usual path for entering the HD Radio amplitude signal is through the standard analog input. For PWM and digitally modulated transmitters, the analog filter has to be removed or modified in some way, which leaves the transmitter unprotected against undesirable audio or noise that may enter through this port during HD Radio operation.

Exciter — The exciter provides the drive signal to each PA to create the amplified output waveform. The drive characteristics determine the duty cycle of the amplified waveform.

Power Supply — The power supplies are an important feature of the 4MX transmitter design. They run as efficiently as the state of the art will allow at 400 VDC. A power-factor corrector supply creates this voltage easily and efficiently.

The supplies were over-designed for efficiency and reliability. Two input diode bridges are used, even though the design requires only one. The FET used in the power-factor corrector circuit delivers 2.85 kW maximum, even though it still operates at the stresses created by as much as a 6 kW output power. The power-factor correction inductor was designed for twice the actual current to keep losses at a minimum. The catch diode is the latest silicon-carbide diode, which improves the efficiency of the supply by half a percent. All these design implementations yield an efficient (97 percent) and reliable power supply.

The 4MX provides a one-to-one relationship between the power supplies and the power amplifiers. Each power supply operates only one PA with no busing of power supplies. This scheme provides optimum redundancy for the system.

Power Amplifier — Each power amplifier has an H-Bridge topology. Each runs independently of the other amplifiers in the transmitter and the output of each amplifier is in phase with all other modules. This is important because it allows the 4MX to operate on a single-supply and power-amplifier combination, resulting in excellent operation at very low powers. In addition, the same drive signal is applied to all PA modules. The

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The 4MX transmitter consists of six major blocks

Understand Ground Loop Problems: Part III

By Bill Whitlock

The author is president of Jensen Transformers Inc. in Van Nuys, Calif.

In the first two parts of this article, we explored the basics of audio signal interfaces. We also explained why National Electrical Code requires AC power wiring to be structured in a way that prevents shock, electrocution and fire. We learned that this structure unavoidably produces problems when electrical noise voltages or currents couple into audio signal paths. We explored balanced interfaces, their advantages and also their potential problems.

In this, the conclusion, we will learn how to troubleshoot a system — and, once we locate the point of noise coupling, how to cure it.

TROUBLESHOOTING

Under fortuitous conditions, systems may be acceptably quiet in spite of poor grounding techniques. However, logic and physics ultimately will rule, although we can sometimes get away with poor practices.

Troubleshooting noise problems can be frustrating and time-consuming. A methodical approach can make the experience much more rewarding. The simple procedure outlined here requires no test equipment (other than ears), yet it reveals not only the nature of the problem, but its exact location in the signal path.

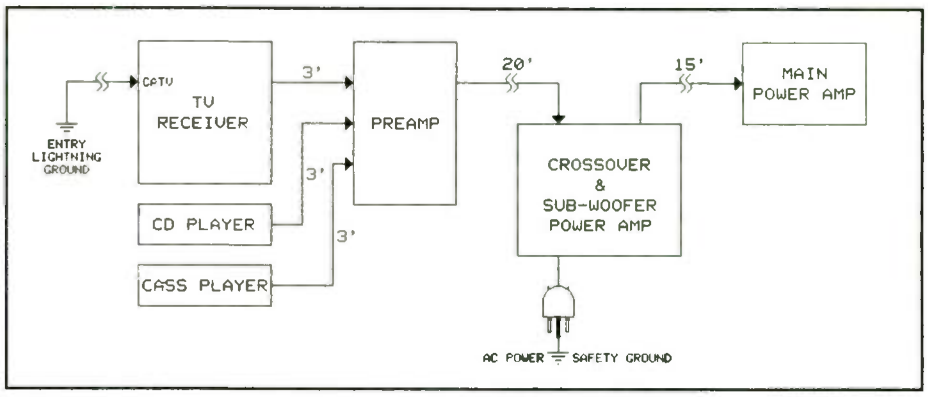

Fig. 1: Diagram of Simple Home Theater System

QUESTIONS, DIAGRAMS AND CLUES

Gather as many clues as possible before you try to solve the problem. Ask

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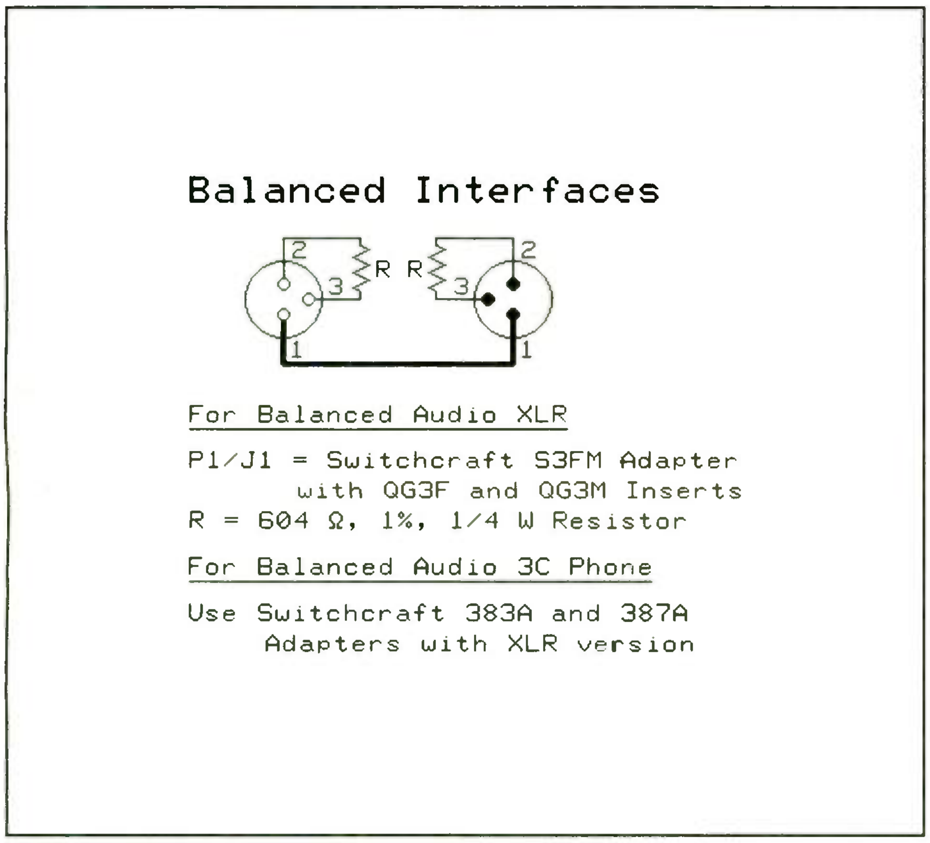


Fig. 2: Dummy for Unbalanced Interfaces

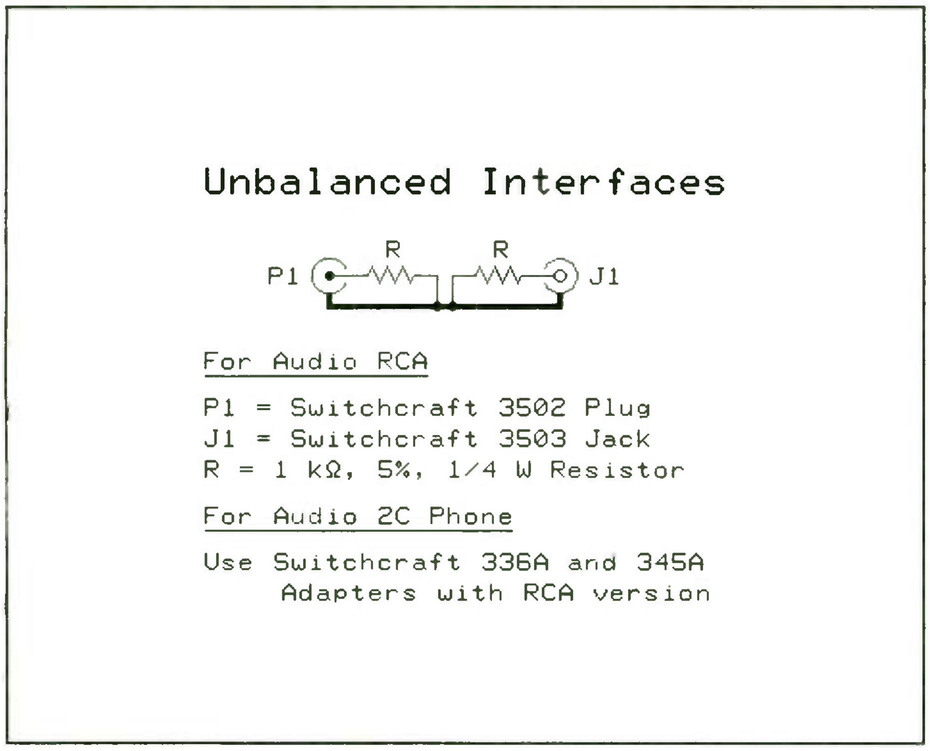


Fig. 3: Dummy for Balanced Interfaces

GROUNDING

CONTINUED FROM PAGE 21

questions: "Did it ever work right? What are all the symptoms? When did it start? What other symptoms showed up just before, just after, or at the same time?" Write everything down! Imperfect recall can waste lots of time.

Sketch a block diagram of the system. Show all signal interconnecting cables and indicate their approximate length. Mark inputs and outputs as balanced or unbalanced. Generally, stereo pairs can be indicated with a single line. Note any equipment that's grounded via its power cord or rack mounting. Note any other ground connections such as cable TV, satellite receiver or feed line from another facility. Fig. 1 is an example of such a drawing for a simple home theater system.

Operation of the equipment's controls, along with some simple logic, can provide valuable clues. For example, if the noise is unaffected by the setting of a volume control or selector, logic dictates that it must be entering the signal path after that control. If the noise can be eliminated by turning the volume down or selecting another input, it must be entering the signal path before that control.

THE DUMMY TEST

Special, easily constructed test adapters or dummies serve as test tools. Because they block the normal signal path but allow ground currents to flow, their temporary placement at strategic locations will reveal precise information about the nature of the problem:

✓ Common-impedance coupling in unbalanced cables
✓ Shield-current induced noise in balanced cables
✓ Pickup of magnetic or electric fields by cables
✓ Defective equipment with the Pin 1 Problem

The audio dummies are made from standard connectors wired as shown in Figs. 2 and 3. They do not pass signal, so make sure they're clearly marked and accounted for after testing!

Always work backwards toward the sources in the signal path. For example, unless clues suggest otherwise, begin at the inputs to the power amplifiers in an audio monitoring system. Each suspect interface is tested using a four-step procedure detailed in the drawings. It's generally a good idea to use dummies in both channels when dealing with stereo signals. This test has been adapted to video systems, too. See the image "Dummy Test Figures."

GROUND ISOLATION IN THE SIGNAL PATH

A device called a ground isolator solves the fundamental noise problem in unbalanced interfaces. Broadly defined, a ground isolator is a differential responding device with high common-mode rejection. It is not a filter that can selectively recognize and remove hum, buzz or other noises when simply placed at the end of a signal path. To eliminate noise coupling, a ground isolator must be installed at the interface where it actually occurs.

A transformer is a device that fits the definition of a ground isolator. A transformer magnetically transfers signal voltage from one circuit to another without any electrical connections between the two. As shown in Fig. 4, when a transformer is inserted in an unbalanced signal path, the connection between device grounds through the cable shield is broken. This stops the flow of power-line noise current in the grounded (shield) conductor, which eliminates the noise.

LOOPS CAN INVOLVE MULTIPLE INTERFACES

Ground loop current may flow through several interface cables and pieces of equipment to complete a path between points having a ground voltage difference. As shown in Fig. 5, this ground voltage exists between the CATV cable shield and the safety ground of the sub-woofer, causing a relatively large noise current in the shields of all the intervening signal cables. Common-impedance coupling at each cable will add noise to the signal, generally in direct proportion to the length of each cable. This system would exhibit a loud hum regardless of control settings because of coupling in the 20-foot cable.

This loop could be broken by: 1) mis-applying a three-to-two prong AC adapter (not recommended!); 2) installing an audio ground isolator in the path of the loop; or 3) installing a CATV ground isolator at its input to the TV. Of course, a CATV isolator must be installed downstream of the lightning ground connection!

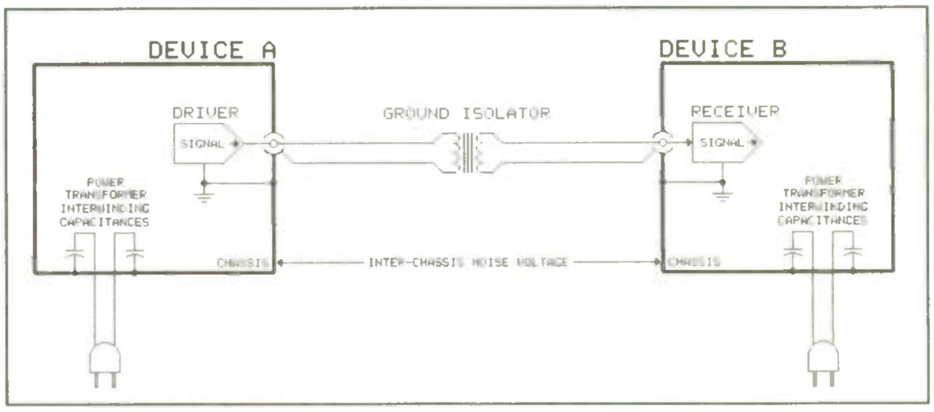


Fig. 4: Isolator stops noise current in shield.

GROUNDING FLOATING EQUIPMENT

Since most unbalanced interfaces are made to consumer equipment that is ungrounded (i.e., two-wire AC cord), isolating an interface may leave a device floating with no ground reference at all. This allows the voltage between the isolator input and output to approach 120 V AC. While not dangerous, it puts an extreme and unnecessary rejection burden on the isolator.

This problem is solved easily by simply adding a separate ground connection to the floating device, as shown in Fig. 5A. This can be done by replacing its two-prong plug with a three-prong type and adding a green wire from the safety ground of the new AC plug to a chassis ground. Use an ohmmeter to verify that the selected chassis tie point has continuity to the grounded contact of one of its audio connectors (which itself can be used if no other point is available).

GUIDELINES FOR UNBALANCED INTERFACES

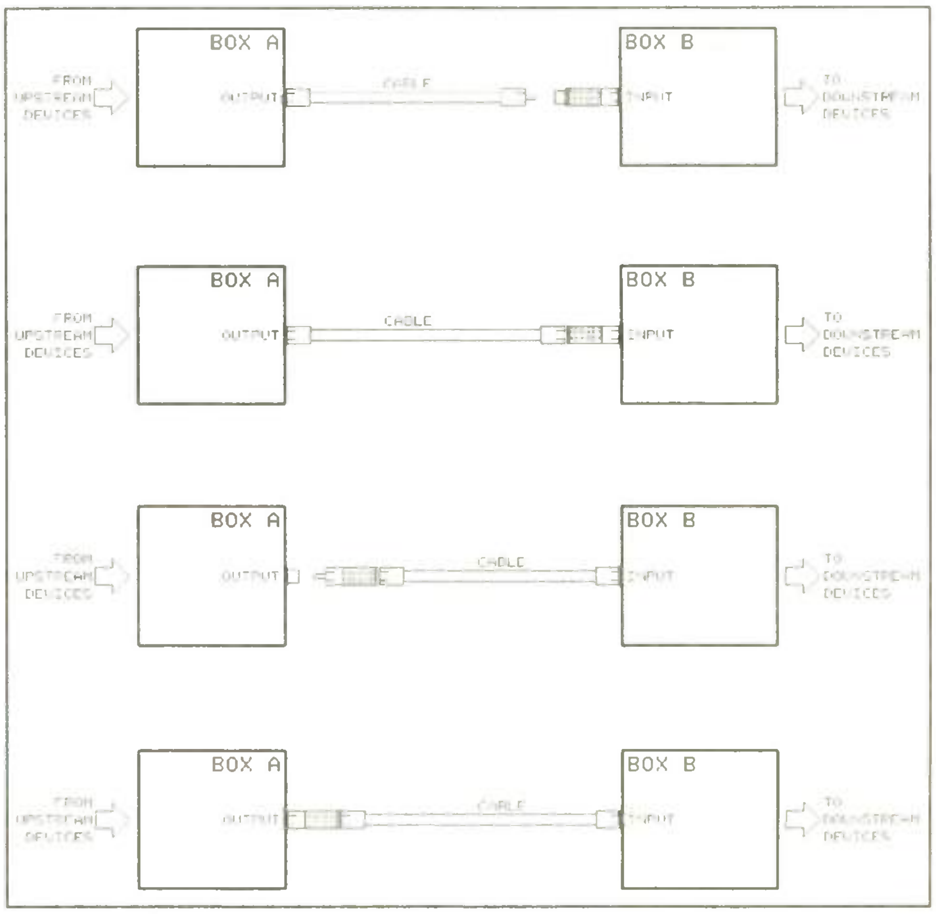
Keep cables as short as possible! Longer cables increase the coupling impedance. For the same reason, never coil excess cable length.

Use cables with low-resistance ground conductors! Since the only cable property that has any significant effect on audio frequency noise coupling is shiel d resistance, replace cables having foil shields and small drain wire with those having heavy braided copper shields. Especially for long cable runs, consider a low capacitance, low shield resistance cable such as Belden #8241F. Its low 17 pF per foot capacitance provides maximum bandwidth and its low 2.6 milliohm per foot shield resistance is equivalent to #14 gauge wire, reducing common-impedance coupled noise. It's also flexible and available in many "designer" colors.

Bundle signal cables! All signal cables between any two boxes should be bundled. For example, if the L and R cables of a stereo pair are separated, nearby AC magnetic fields will induce a current in the loop formed by the two shields, coupling hum into both signals.

Maintain good connections! Contact resistance is part of the coupling impedance. Hum or other interference that changes when the connector is wiggled indicates a poor contact. Use a good

Dummy Test Figures



contact fluid and/or gold plated connectors.

Don't add unnecessary grounds! Additional grounding of equipment generally will increase ground loop currents rather than reduce them.

Use ground isolators at problem interfaces! They solve the fundamental problem by eliminating common-impedance coupling. Commercial ground isolators are available in bandwidths suitable for audio, video and CATV signals.

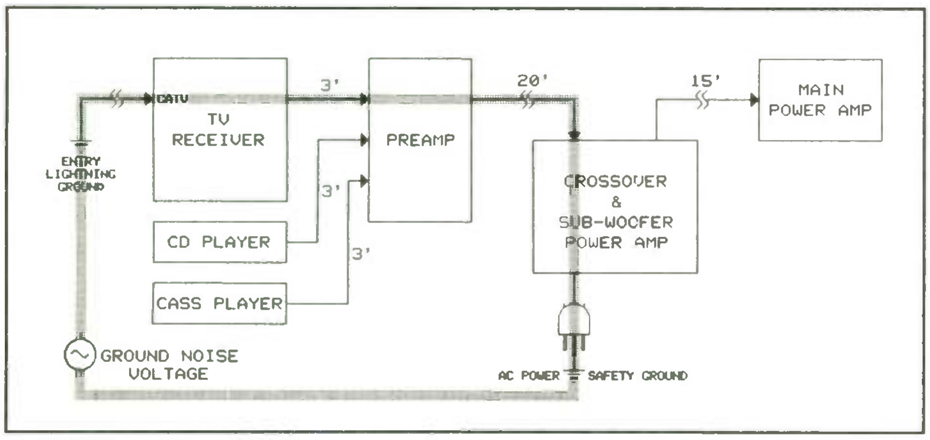


Fig. 5: Ground Loop in Simple Home Theater

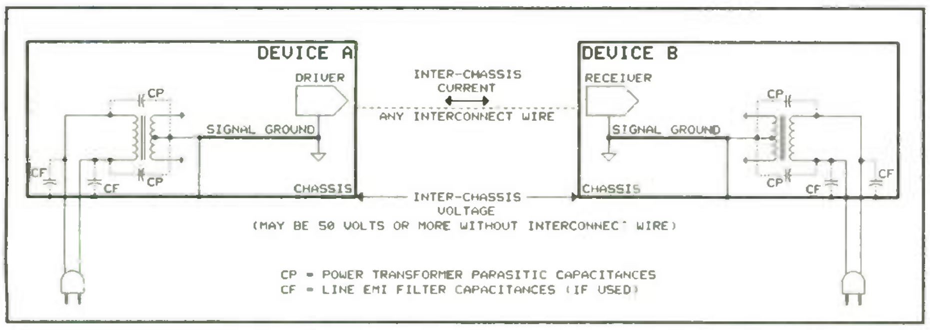


Fig. 5A: Sometimes an added ground is necessary.

SOLUTIONS FOR BALANCED INTERFACES

Beware of the Pin 1 Problem! Lots of commercial equipment has this designed-in problem, some from respected manufacturers. If disconnecting the shield at an input or output reduces a hum problem, the equipment at one end of that cable may be the culprit. Use the hummer test to be certain.

Add an isolator to improve CMRR! Widely used "active balanced" equipment inputs have generally unpredictable CMRR in real-world systems. Actual in-system CMRR may be quite low. A quality isolator using a Faraday-shielded input-style transformer can increase the CMRR of even the most mediocre balanced input to over 100 dB.

UNBALANCED OUTPUT TO BALANCED INPUT INTERFACES

Be wary of using adapters! Using shielded single-conductor cable and an RCA to XLR adapter as shown in Fig. 6 results in zero ground noise rejection. All the potential benefit of the balanced input is lost!

Fig. 7 shows the proper way to interface an unbalanced output to a balanced input. Simply using a shielded twisted-pair wired as shown, the smart connection, generally results in about 30 dB of rejection for typical active-balanced inputs.[3]

BALANCED OUTPUT TO UNBALANCED INPUT

Balanced equipment uses a wide variety of output circuits. Some, such as the one in Fig. 8, can be damaged when one output is grounded. Others, including most popular servo-balanced output stages, may become unstable unless the output is grounded directly at the driver end.[3] Unless the balanced output already has a built-in transformer, using an external ground isolator, such as the one shown in the figure, is the only method that will work with any output stage.

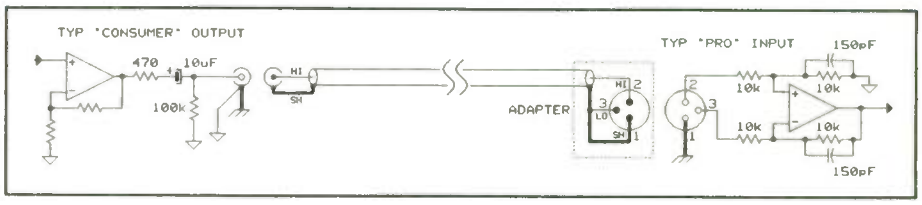


Fig. 6: Two-conductor cable wastes rejection of input.

SIGNAL QUALITY ISSUES

Ground isolators in signal lines solve ground-loop problems without resorting to unsafe and illegal tampering with safety grounds. However, isolators using poorly designed or undersized transformers can cause loss of deep bass, bass distortion and poor transient response. Beware of cheap products with scanty or nonexistent specs — they're often made with $2 data-com transformers. High-quality ground isolators have the bandwidth, phase response and harmonic distortion characteristics to make them audibly transparent, even to audiophile standards.

Whether used in balanced or unbalanced interfaces, high-quality transformer-based isolators have a number of other benefits:

The input to the isolator is truly universal; accepting signals from either balanced or unbalanced sources, without compromising very high noise rejection.

Those using transformers with internal Faraday shields, not to be confused with magnetic shields, provide not only the highest CMRR, but also inherent suppression of ultrasonic and RF interference. The resulting reduction of spectral contamination in downstream equipment is often described as a marvelous new sonic clarity.[4]

Some isolators, such as the ISO-MAX Pro units, can also solve equipment pin 1 problems.

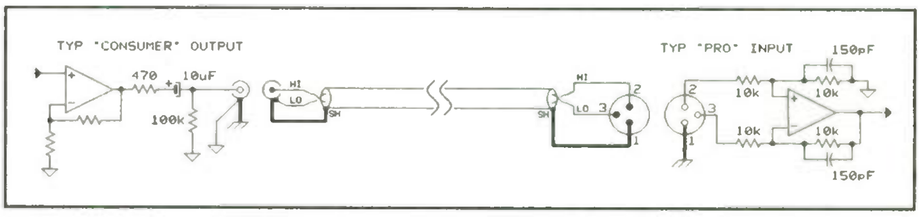


Fig. 7: Three-conductor cable uses rejection of input.

Several manufacturers make powered active boxes using ordinary differential amplifiers as ground isolators. Fig. 9 compares the 60 Hz hum rejection of a typical active ground isolator to an ISO-MAX transformer-based isolator. Note that over the 200 ohm to 1 k-ohm range of typical consumer source (output) impedances, the active isolator can achieve only 15 to 30 dB of hum rejection. Under the same typical operating conditions, the transformer isolator achieves 90 to 110 dB. Transformers require no power and are inherently robust. They are virtually immune to transient over-voltages that can spell disaster for active units.

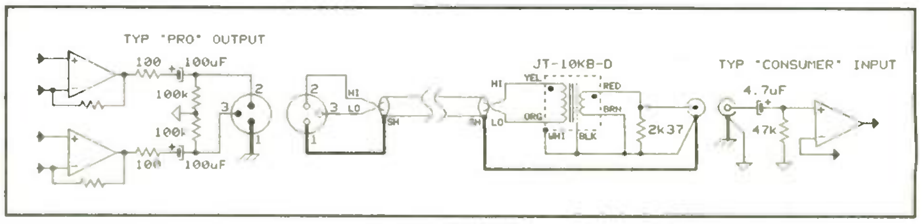


Fig. 8: Universal Balanced Output to Unbalanced Input

ALTERNATIVE TREATMENTS

Technical or Isolated Ground Systems

A so-called technical ground scheme can sometimes reduce electrical noise in the safety-ground system.

It is most applicable in situations where

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GROUNDING

CONTINUED FROM PAGE 23

conduit or junction boxes may come in contact with building steel, water pipes, gas pipes or other structures that may be grounded in other locations, thus causing additional noise current in the ground system. The scheme uses special insulated-ground or IG outlets (generally orange in color) that intentionally insulate their green safety-ground terminal from their mounting yokes or saddles. Therefore, safety grounding is not provided by the J box and conduit, but by a separate insulated green wire. This wire must be routed back to the electrical panel alongside the white and black circuit conductors to keep inductance low. In most cases, wiring is not daisy chained to other outlets on the same branch circuit.

These practices are covered by NEC Article 250-74 and its exceptions.

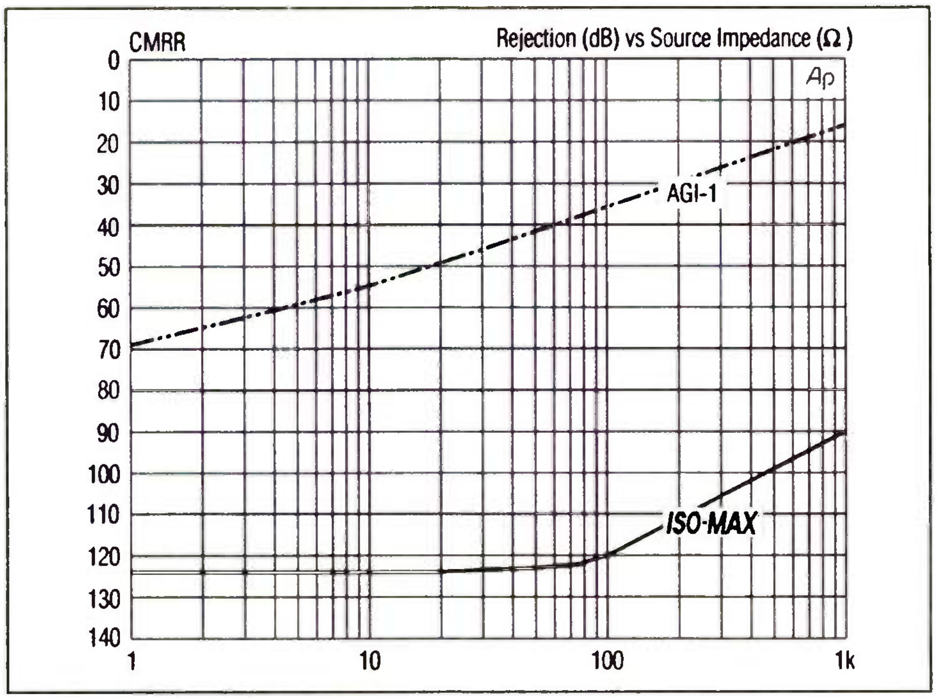

Fig. 9: Transformer far outperforms diff-amp.

BEEFING UP THE GROUND SYSTEM

Because of the effects of inductance (see Grounding Myths), attempts to lower the impedance of a ground system by adding lots of copper generally result in small improvements at high cost, especially if the cost of installation labor and downtime are considered.

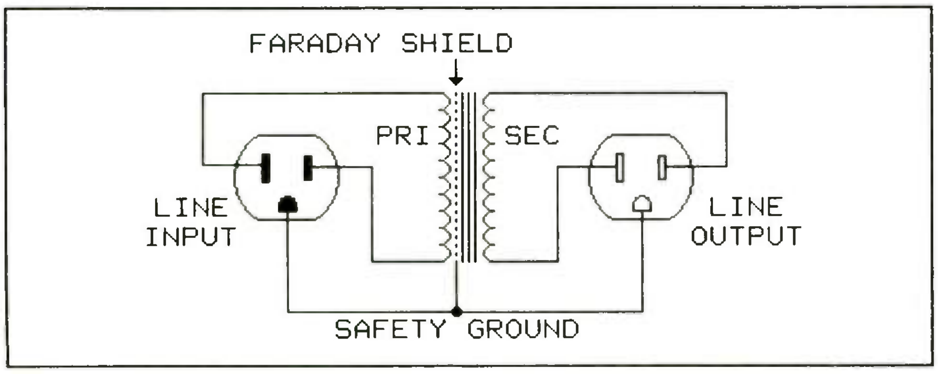

Fig. 10: Isolation Transformer for AC Power

Remember that an earth ground is not a substitute for safety grounding! Code allows additional ground rods to be added (bonded) to a properly implemented safety-ground system, but doesn't allow them to replace it. Soil resistance is simply too high and unstable to be relied on to divert fault currents.[5]

Equipment grounding via the standard power-cord safety ground is logical, easy to implement and safe. It's recommended highly for all systems and is the only practical method for portable or often reconfigured systems.

POWER LINE CONDITIONING

Over the last few years, fear, horror stories and junk science have been used to sell a host of power-line conditioning and protection devices. Because system noises are most frequently coupled from the power line, solutions that somehow cleanse or purify it have great intuitive appeal.

However, in this author's experience, such treatments generally produce only marginal improvements. Usually the most cost-effective solution is to identify and eliminate the ground loops or other problems that allow the noise coupling in the first place. This approach solves the real problem. Treating the power line to get rid of noise is like using a shotgun instead of a silver bullet to target the problem!

First, when any line filter, conditioner or isolation transformer is used, Code requires that the device, as well as its load, still be connected to safety ground as shown in Fig. 10. Because these devices divert additional 60 Hz and high-frequency noise currents into this safety-ground system, they frequently aggravate the very problem they claim to solve.

Second, the advertised noise attenuation properties for virtually all these power-line filters and transformers are unrealistic. Measurements are made with the equipment (generator, detector and device under test) mounted on a large metal ground plane. Although the resulting specs are impressive, they do not apply to performance in a real-world situation where grounding is via safety-ground wires or conduit. However, such devices can be effective if installed at a power panel and feed an entire branch circuit, sometimes called a technical branch.

Balanced power — or more properly, symmetrical power — is a seductively appealing concept, illustrated in Fig. 11. However, many of its proponents mistakenly assume that each piece of equipment has precisely matched capacitances from each leg of the power line to chassis when they explain how it cancels ground noise. Of course, if this were true, capacitive noise currents from each 60 V leg would then be of equal magnitude and opposite polarity, and would completely cancel each other. But this assumption isn't true for the overwhelming majority of real-world equipment, where ratios of capacitances C1:C2 or C3:C4 are often 3:1 or 4:1.

It's unlikely that equipment manufacturers will ever adopt costly power transformers having capacitively balanced windings or RFI filters having precision capacitors. Even balanced power proponents admit that actual noise reduction is usually less than 10 dB and rarely exceeds 15 dB. For audio, a 10 dB improvement will rarely make the difference between unacceptable and acceptable system performance. In fact, many of the benefits ascribed to balanced power and other conditioning are due to simply plugging all system equipment into the same outlet strip or dedicated branch circuit. Whenever possible, this is always a good idea!

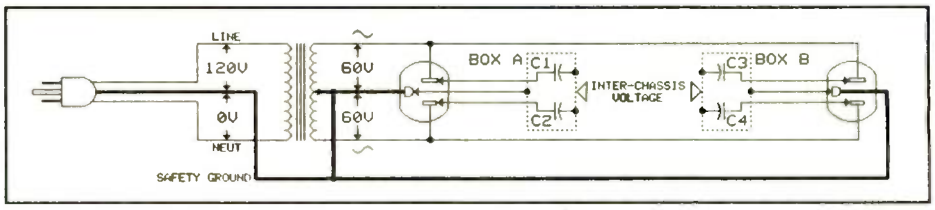

Fig. 11: Concept of balanced power is seductive.

GROUNDS AND SURGE SUPPRESSORS

The haphazard placement of conventional surge protectors at point-of-use outlets can actually increase the risk of equipment damage when signal or data lines bridge devices on different branch circuits.[6] Conventional MOV-based suppressors divert surge energy into the safety-ground system, creating very high-ground voltage differences during an actual surge or spike event, as shown in Fig. 12. Therefore, for protecting sub-circuits or individual groups of equipment, I recommend series-mode suppressors, such as those by Surge-X, which don't create noise or potentially damaging voltage differences on the safety-ground system. Series-mode protection offers high impedance to the surge energy rather than trying to shunt it to ground.[7]

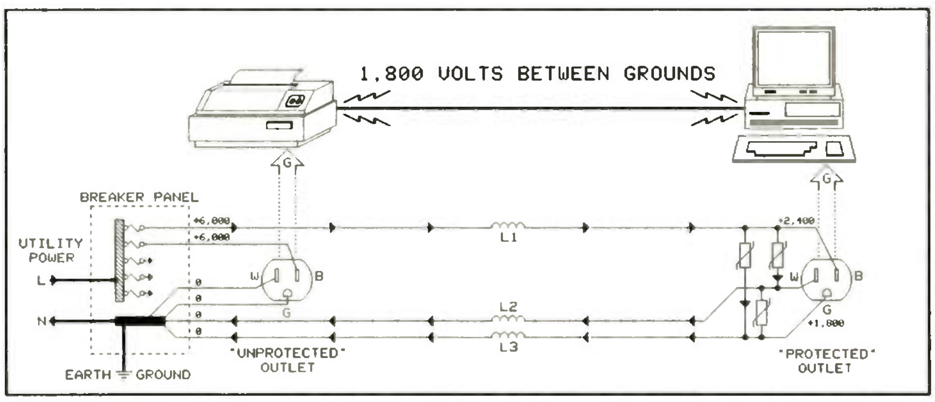

Fig. 12: Equipment Damage Caused by Surge Protector

AUDIO CABLES OF 'EXOTIC' DESIGN

No other product is as shrouded in pseudo-science and hype as the audio cable. In reality, there is no mystery surrounding genuinely audible differences among cables.

For example, the physical design of a cable is known to affect its coupling of ultrasonic power-line interference. Even low levels of such interference will cause audible spectral contamination in downstream amplifiers.[8] In this case, the real solution is to prevent the coupling in the first place (with a ground isolator), instead of agonizing over which "designer cable" makes the most pleasing improvement.

Audio cables are not transmission lines. Hype for expensive cables often invokes classic transmission-line theory and implies that nanosecond response is somehow important. Real physics reminds us that audio cables do not begin to exhibit transmission-line effects in the engineering sense until they reach about 4,000 feet in physical length. Expensive and exotic cables, even if double-or triple-shielded, made of 100 percent pure unobtainium, and hand-braided by Peruvian virgins, will have no significant effect on hum and buzz problems.

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Schrag

CONTINUED FROM PAGE 15

that noise — by keeping the airflow velocities slow enough, by choosing ductwork geometry that avoids turbulence, by selecting equipment designed for quiet operation, by placing the supply and return devices within a room to promote uniform distribution with less air movement and by installing vibration isolation devices to keep noise from traveling through the building structure.

Sometimes that's easier said than done. We had a project recently where the floor-to-floor height was extremely limited. There wasn't even enough room to route ductwork above the control rooms and studios without lowering the ceilings below the minimum required to meet code, let alone what we would have liked acoustically. Worse than that, ductwork couldn't be routed above the corridors either, and the floors above and below were off limits, so centralized air handlers were out of the question.

The only viable solution was to create an HVAC closet next to each of the rooms with a small local fan-coil unit, and route the ducts through soffits at the perimeter of the room. Prefabricated sound attenuators were used on the supply duct, with a labyrinth of ductwork running vertically inside the room for the return path. The resulting system was quiet enough, but a great deal of time and money and creativity were expended trying to overcome a shortcoming in the building that should have been obvious from the very beginning. Had we been involved before the building was selected, we would have raised a big red flag about the hidden costs of getting such a "good deal."

What is the importance of wire management systems in studio spaces and why is this perhaps a design function best done by an acoustic consultant?

Well, actually, I don't think that the acoustical consultant is always the one who's best suited to handle wire management issues, even though those systems definitely have implications for the sound isolation between rooms. What is essential is that the wire management system be regarded as an integral part of the design — not an afterthought or something that's considered separately.

That means its design should be in the domain of someone who understands how the system needs to function and what alternatives exist for different ways to get cables from point A to point B — someone who knows how those pathways can create sound leaks between rooms and what to do about it — and someone who can integrate those systems into the architectural design without having them look like something tacked on at the last minute.

Oh, yeah, and it had better be someone who is sensitive to the plight of the engineers or integrators who have to come in after everyone else has finished up to pull the wires — and then open it up again after a few months or so to pull that one new cable ... to a live control room ... during a 90-second break ... in the middle of election night coverage.

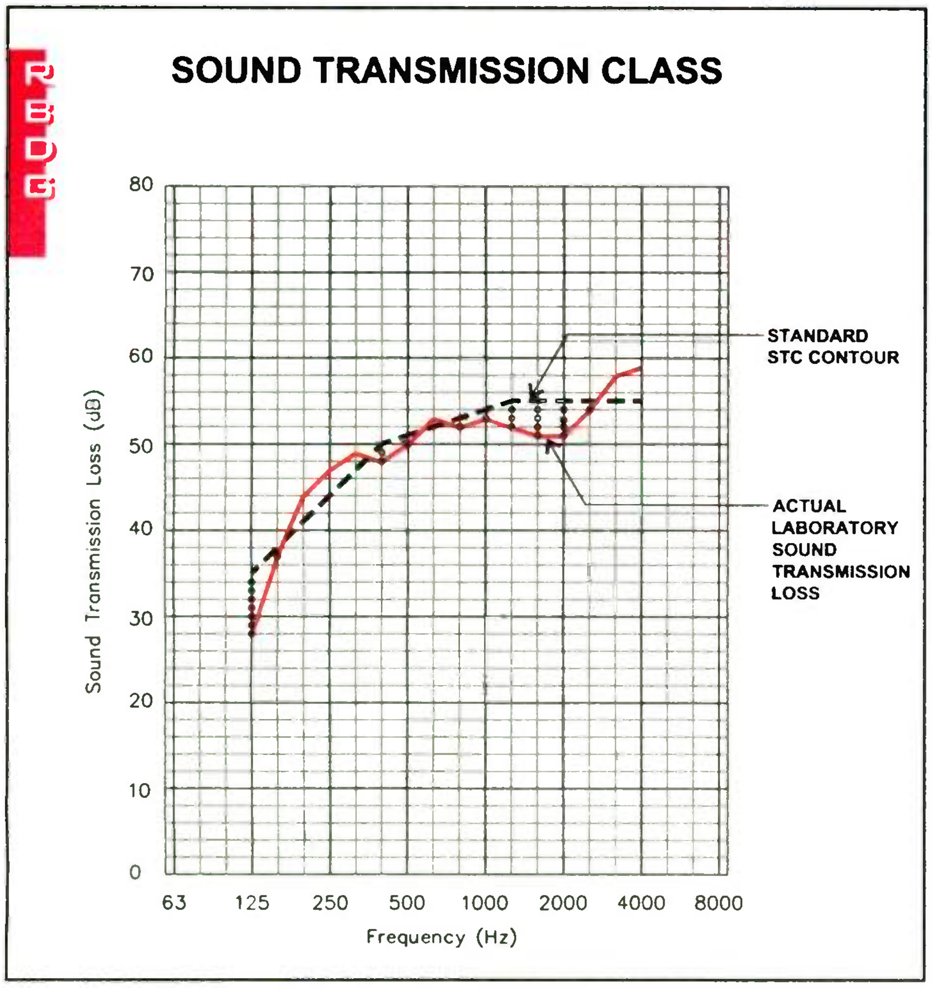

Fig. 5: Sound Transmission Class Chart

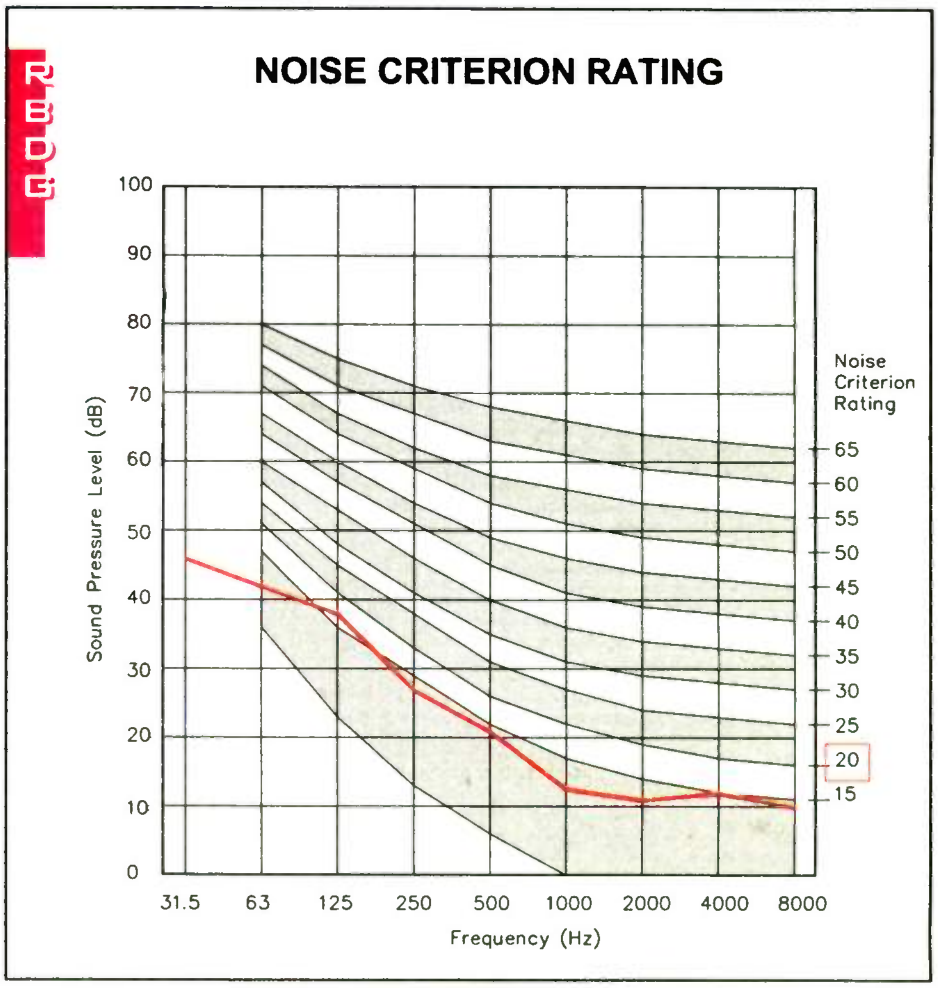

Fig. 6: Noise Criterion Chart

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4M MODULATION

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drive to the power amplifier is key to the 4MX operation. The details of this drive circuitry are in the patent process and cannot be described in detail here; however, the system's results and benefits are described in the conclusion.

Bandpass Filter — The bandpass filter was designed to reduce the harmonics of the carrier to an acceptable level. The 4MX transmitter uses a bandpass filter designed to be broadband, with approximately 180 degrees of phase shift between devices and the output port. The bandpass filter also matches the ideal PA load impedance of 10 ohms to the 50 ohms output impedance.

EFFECTS ON HD RADIO

The 4M Modulation and 4MX technologies eliminate the need for a modulator stage and the LPF. This offers a tremendous efficiency advantage without limiting the bandwidth of the system. Unlike all other transmitter designs, the 4MX transmitters allow the I and Q or Ethernet data signal for HD Radio to be fed directly into the processor. Feeding the HD Radio signal via a separate path allows the analog input audio to be passed through its protective and noise-reducing filter. The I and Q or Ethernet data signals can then be added at the processor stage. This means that no modifications are needed to operate in the HD Radio mode. The Ethernet data signals are applied directly to the exciter via the Exgine card.

Modulation Method	Efficiency	Power Range	HD Radio Spectrum	Spurious/Noise	Modulation Capability	Audio	Size	Weight
4M	89%	High and low	Excellent	Excellent	Excellent: 145% to -100%	Excellent	45"W x 25"D x 87"H	Less than 1,000 lbs
Digital	87%	High	Excellent	Good; quantization and switching products associated with frequency that modules are turned off and on	Excellent: 145% to -100%	Excellent	102"W x 54"D x 78" H	More than 3,500 lbs
PWM	75%	Low, due to efficiency limitations	Good; only if PWM switching frequency is high (> 120 kHz)	Good: dependent on PWM switching frequency	Limitations at negative modulation due to nipple effect	Good; limited by PWM frequency	108"W x 48"D x 73"H	More than 3,500 lbs

Table 1: Performance Comparison (available 50 kW models)

WHY THE NAME '4M MODULATION'?

Rarely are engineers allowed input on how the product model is identified. Using 4M Modulation to designate the name of the modulation scheme developed by Broadcast Electronics has a meaning related to how the system works. The "4" in 4M is a play on the mathematical term "Fourier."

In 4M Modulation, the amplitude of the filtered output carrier is determined by calculating the coefficient of the first term of the Fourier expansion of the waveform presented to the combiner transformer. This is very different from the waveform produced by filtering a PWM signal, where the amplitude of the filtered output carrier is a linear function.

CONCLUSIONS

The information below compares the 4MX transmitter to those using other modulation techniques.

- In a PWM transmitter, the modulator uses a duty-cycle modulated waveform to create a DC voltage proportional to the audio input signal. This audio-modulated DC is then sent to the PA to produce the final AM signal. In a 4MX transmitter, the duty cycle of the RF waveform is modulated directly, without the use of a modulator.

- The PWM transmitter performs modulation at some intermediate frequency (70 kHz to 150 kHz); the 4MX transmitter modulates at the carrier frequency.

- For HD Radio, the PWM transmitter is band-limited; the 4MX transmitter is not.

- The PWM transmitter has one audio input for the amplitude portion of the HD Radio signal. This requires modification or removal of the audio input circuit during HD Radio operation. The 4MX transmitter has the Ethernet data sent directly to the processor, so no modification of the audio input filter is required.

- The digitally modulated transmitter requires many PAs and turns them off and on to create modulation. The 4MX transmitter requires only one PA to achieve modulation. This allows the 4MX transmitter to produce high-quality audio at low power.

- A transmitter using digital modulation has one audio input for the amplitude portion of the HD Radio signal. This requires modification or removal of the audio input circuit during HD Radio operation. The 4MX transmitter has the Ethernet data signals sent directly to the processor, so no modification of the audio input filter is required.

- The 4M Modulation design results in a more compact design that is a fraction of the footprint and weight of comparable designs.

The table accompanying this article summarizes the advantages of the 4MX transmitter over other AM systems. ■

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Why do low-frequency sounds present a special challenge in studio design? What are possible techniques for controlling them? How can these concepts be integrated into the studio design?

The behavior of low-frequency sound is the least understood part of room acoustics and is certainly the most difficult aspect to tame. First of all, it's important to recognize that, to a great degree, the low end is dependent on the volume, size and shape of room. If you make poor choices with those aspects of the design, it's sometimes impossible to fix them with any kind of finishes or special devices inside the room.

There are a lot of products on the market to deal specifically with controlling low-frequency sound, and one or two of them actually work. (Laughs)

No, really, the challenge is in selecting materials that work in the frequency range that you want them to, without having such a narrow Q that they create as many problems as they solve. Many "traps" — both commercially available products and homebrew solutions — are like that.

Also, as I said earlier, the acoustical finishes have to be chosen to maintain balance, and it's sometimes difficult to selectively treat just the low end without disrupting what's going on in the mids and highs. Another maddening aspect of low-frequency treatments is that where you place them is often just as important as what you use, and it's not always intuitively clear what the most effective location will be.

We've developed a lot of custom solutions to handle low-frequency sound in the studios we've designed, but we've also focused on blending different treatments that complement each other. For example, it's easy to forget that acoustical diffusers have bandwidth characteristics, just like absorbers, so we often try to provide low-frequency components that pick up about where the diffusers leave off. When you see the different pieces in combination as part of a system, you can mount them in such a way that low-frequency diffusion is created below what the diffusers themselves can do, and then transition smoothly to a very controlled bottom octave (see Fig. 7).

Tell us your studio design tips. Write to Radio World Engineering Extra Technical Editor Michael LeClair at mlrwee@verizon.net.

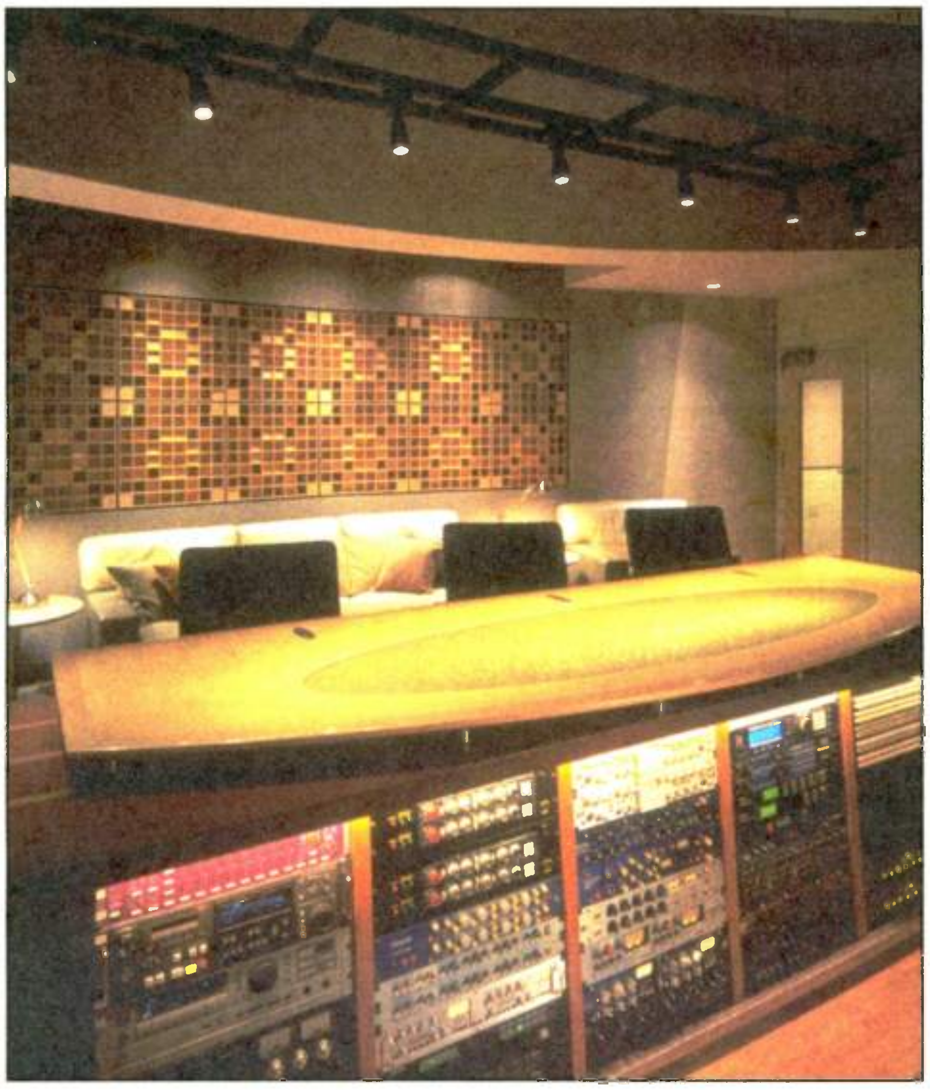

Fig. 7: Sound Diffusion Combined With Absorption

You can reach Richard Schrag of Russ Berger Design Group at richard@rbdg.com. ■

Keeping Score on Satellite Radio

by Guy Wire

Guy Wire is the pseudonym of a well-known veteran radio engineer who prefers to remain anonymous.

Two years ago I predicted XM and Sirius would have a tough time surviving in the midst of a hard recession while using business models based on tons of red ink far into the future. Surely shareholders would run out of patience and hold them accountable for more rational balance sheets and business practices before rewarding them with higher stock prices.

I think I ignored the lessons of the dot-com bubble and investors' willingness to ride a hot pony with irrational exuberance. Or perhaps I just didn't give them enough time before reality would set in.

The market has leveled off and pulled back into a choppy trading range and appears to be taking a more sober look at what the prospects for these companies really are. Numerous investment advisors are dropping their calls to neutral or outright sell. Of course, satellite stock prices have jumped around a lot, and probably will continue to be volatile in the near future.

Back in 2002 it seemed likely the much smaller Sirius would falter and merge or morph into something else. That hasn't happened. Yet. After leaving Viacom and losing interest in traditional media, Mel Karmazin has ridden in on a white horse with sidekick Howard Stern to keep the Dog Star burning for now. But even Mel may have lost his magic touch with Wall Street. After his recent press conference touting the latest Sirius video add-on feature, for example, the stock immediately sold off 5 percent.

The economics of both Sirius and XM still seem way out of balance at present subscription rates and income levels.

HEARING FOOTSTEPS

The dot-com boom and bust cycle lasted about eight years. Could satellite radio be following in the same footsteps? There are many parallels and similarities. I only regret that I didn't buy them two years ago and then sell a few months ago to reap profits approaching 1,000 percent.

Instead of shoring up their business plans by restructuring and scaling back risk like any typical young company struggling to become profitable, they're going in the opposite direction. Both have tried to enhance their content offerings and appeal by increasing programming commitments and debt while diluting their share values by issuing more stock. The only thing saving them for now is a steadily growing subscription base. That's the key difference between these companies and many dot-coms that had nifty ideas and a few products but not enough buyers. Wall Street eventually lost its patience with the dot-com bubble businesses and many bit the dust.

The obvious challenge for satellite radio is to keep growing fast enough to convince investors they will turn the corner and become profitable over the long haul. As happened after cable and satellite TV were launched and grew rapidly, subscription growth eventually leveled off and churn rates increased. XM and Sirius hope to cross the threshold of profitability before that happens. As of January, XM counted more than 3 million and Sirius had 1 million subscribers. Both have a very long way to go.

MAKING THE CASE

Radio isn't television. Even in large markets, the selection of over-the-air TV programming was very limited compared to what could be delivered over cable and satellite. Those services have succeeded because most American households have been willing to pay to get a dramatic increase in viewing choices, all with near-perfect pictures and reception.

According to Leichtman Research, the total number of cable and satellite TV subscribers is about 90 million in the United States. First we heard the sat radio boys thought they could eventually count half that many who will buy their service. Now, Karmazin is saying

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Guy

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they might even equal that many. Nobody ever accused these guys of not being supremely optimistic.

The case for pay radio is nowhere near as compelling as pay TV's. The most important and valuable radio listening is mobile during drive times. Coverage of local news, weather, traffic, sports and civic events is still the most powerful reason folks turn on the radio. Even in average-sized cities, the number of viable radio-station choices is 20 or more, with many major formats available. CD players, Walkmans, i-PODS and portable MP3 players augment the majority of listening needs and choices for music aficionados. The universe of those who are willing to pay for more radio programming and music choices beyond these traditional methods will most certainly prove to be a small fraction of that enjoyed by pay TV and video.

Sirius is basing much of its future growth on Howard Stern and NFL football, while XM has high hopes for its Major League Baseball deal. Practically all industry observers have noted the prices paid for such marquee content were ridiculously high. The Wall Street Journal has said Sirius is "crushing its own windpipe" with such deals. XM has also added the infamous Opie and Anthony (O&A), plus popular talkers Dr. Laura and Gordon Liddy. The price of good talent honed in traditional radio is being bid way up and a few are crossing over.

Both companies, but especially Sirius, are betting that new subscribers will pay the extra freight and sometime in the future a profit will be made. They keep upping the ante in spite of the odds their bet may not pay off. They're also hoping subscription sales will just keep on rolling after consumers who really want the service buy it separately by including the cost of new subscriptions in the purchase of new cars. Dealers can easily bury the sat radio charges in the monthly car payments in hopes buyers won't really notice or remember they agreed to pay for the service.

GOING OVER THE LINE

In its zeal to enforce indecency standards, the FCC effectively has forced several bad-boy jocks off regulated free radio onto satellite. Opie and Anthony are on XM doing their shtick in unexpurgated fashion. Citadel has chosen to replace Stern in some of their markets with O&A as fill-ins. The dump buttons and post editors must be getting a real workout. Presumably, they're doing this as punishment for Howard spending too much time promoting Sirius and his new show that starts there unfettered by FCC regulations next year.

I'm surprised neither Mel nor Howard have thought about why his show was so popular with indecency restrictions, but why it might not be so attractive without them. Those very limits force such talent to create suggestive and titillating radio bits that "come very close to the line" without going over it.

Both Howard's and O&A's fans are largely folks who enjoy hearing how far the creative genius will go and what they will say next without dropping over the edge. Without limits, this "art form" quickly degenerates. Most everyone values some level of decency and the power of imagination. That's why well-produced soft porn attracts much larger audiences than no-holes-barred XXX. It won't take too long to find out if these guys justify their huge salaries as bona-fide stars or just morph into dirty old men being disgusting all too often.

When XM and Sirius first launched, both CEOs said they could compete and coexist like Coke and Pepsi. If that's the case, traditional radio must be water. XM maintained if they could only attract 10 percent of traditional radio's daily audience of 225 million, they would be successful. But their thresholds of profitability have steadily increased along with the mountains of debt. And now they truly believe they can stretch that number to 45 or even 90 million? Such extravagant projections are going to be reined in by the counter-competitive impact of traditional free radio's fledgling transition to digital with multiple channels and many of the same add-on features like text data and 5.1 surround sound.

Stop sets of multiple commercials on the music channels could be just around the corner.

DOING THE MATH

Jesse Eisinger broke down the market valuation numbers for Sirius in his Wall Street Journal article in December.

At $9.02 a share (the stock price at that time), its market value based on 1.74 billion owned shares was $15.7 billion. If each subscriber generates about $12 a month in revenue, that totals $144 per year. But investors look at "contribution margin," which measures cash flow after most variable costs except marketing and SAC or subscriber acquisition costs. Sirius expects a 70 percent contribution margin so the $144 "net" value of each subscriber is reduced to $100 per year.

The Dog Star suddenly looks like a balloon in search of a pin.

The Sirius "churn rate" is always being debated, but a reasonable estimate is 2 percent per month or 24 percent a year for this exercise, he noted. That means after four years, the equivalent of all subscribers roll over or churn off. Consequently the contribution margin becomes $100 times four years or $400. But we also have to subtract the SAC. That number was $200 for 2004, but Sirius is hoping and analysts are projecting that this number could be reduced to $50 in the future. Allowing for a very generous $50 SAC, we now have a value of $350 for each Sirius subscriber. Divide the market cap of $15.7 billion by 350 and you can then project that the market is valuing Sirius as if it has 45 million subscribers, instead of the real-world 1 million.

The Dog Star suddenly looks like a balloon in search of a pin.

BALANCING THE BALANCE SHEET

Despite highly inflated expectations and stock prices, satellite radio does bring a valuable alternate service to the marketplace that extends and augments what traditional radio has done for 85 years. But at what price to consumers that will sustain the providers as viable businesses?

The economics of both Sirius and XM still seem way out of balance at present subscription rates and income levels. Something will have to change. Higher monthly fees, along with tiered surcharges for "premium" features like Stern and NFL games, are only months away.

When both services first launched, much of the hype and attraction was over their "commercial-free" programming. Few clear-thinking observers thought that would last very long. Over the past year, commercials have started to appear. Most of the Sirius talk channels now include commercials. Karmazin, who started his career as a CBS radio spot salesman, never lost his lust for ad dollars. He recently divulged his fervent desire that 5 percent of all radio ad revenue, about $1 billion, would flow into Sirius. Stop sets of multiple commercials on the music channels could be just around the corner.

Nobody ever mentions the issue of satellite vulnerability involving these services. "Rock" and "Roll" and the Sirius birds are insured, but a technical flaw with Rock and Roll projects their life span to only seven or eight years instead of the planned 15-year target. XM now will have to launch two more replacement satellites over the next two years to cover the shortfall.

With all the things that can go wrong, having multi-million dollar standbys ready to blast into orbit on short notice seems mandatory. All it takes is for one lucky asteroid or some errant piece of space junk to crash into any of these and instant catastrophic damage could result. This technology has proven to be reliable and durable so far, but loss of service for any extended period of time could wreak havoc and huge financial losses for either.

LOOKING AHEAD

In spite of an insatiable appetite for spending and debt, almost everyone seems to agree that satellite radio is here to stay. But very soon, real and significant cost-cutting measures are going to have to be a part of their operations. This may already be happening at Sirius under the frugal hand of Mr. Karmazin. While an XM/Sirius merger is a future possibility, the most likely scenario that will ensure the survival of these services will be increased ownership and control by major investors like the automobile companies and large group traditional broadcasters.

When subscriber growth flattens out and these companies still find themselves bathed in red ink, Wall Street undoubtedly will exact its punishment. At that point, the underlying owners will become more actively involved in restructuring the business models that could then subdivide and transition either XM or Sirius into consolidated sub-units of traditional radio. Failing AMs were bought up by profitable groups and FMs over the past 20 years. The new owners had to subsidize the AM operations until they could be transformed into profitability.

As unthinkable as it may sound, Mel could be the tip of that iceberg in reverse, as many of the present employees of XM and Sirius could once again be working for their old bosses in radio companies they left behind.

RW welcomes other points of view. Write to radioworld@imaspub.com.■

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The Deadly Psychology of Schedules and Deadlines

Success in Project Planning Takes More than Statistics

by Barry Blesser

Everyone has experience with projects that are late. We have all heard excuses and explanations — mostly true — from carpenters, architects, technicians, programmers and engineers. In fact, missed project deadlines occur so frequently that they are rarely a surprise. If we understood the psychology of scheduling, planning and managing, would projects be completed on time?

The answer is definitely yes, but only if we consider three fundamental properties of human psychology: our unwillingness to contemplate the unknown, our inability to envision thousands of details and our improved efficiency when deadlines are imminent. Project leaders study the technical details in schedules, but they seldom study the psychology of project schedules.

Consider a hypothetical project of some complexity. Its duration is proportional to the sum of the time required for completing each of hundreds, if not thousands, of individual tasks.

What does it mean to estimate the duration of every task? For tasks that are repeated frequently, we already have an appreciation for those factors that produce variations in completion time.

Consider just one task in a large project: splicing a cable. Some splices take 200 seconds to complete; some take 180, and some 320. Using the data from hundreds of splices, we can construct a data table showing how frequently a splice is completed in a given amount of time. The extremes might range between 100 and 400 seconds, and the average time might be 284 seconds. The probability of a splice taking between 150 and 250 seconds might be 95 percent.

These are the statistical parameters for the task of one cable splice.

Dr. Barry Blesser, director of engineering for 25-Seven Systems, is a former associate professor at MIT and past president of the AES; he is considered one of the grandfathers of digital audio. His book, "Auditory Spatial Awareness of Aural Architecture," will be published by MIT Press in 2005.

PLAN FOR SURPRISES

There is no single number for the time to splice a cable because there are always surprises and unknowns. Variations in time might arise from an unplanned interruption by a colleague, a defect in the center conductor or a headache after a big lunch. As an exercise, try listing all those surprises that could influence the time for completing a familiar task, even the trivial task of brushing your teeth. Without surprises, the time to splice a cable or to brush your teeth would be constant and predictable.

For a large project like building a new studio, the composite of many individual tasks, the total completion time has dramatically larger variation. By assuming that all tasks have worst-case durations, a project might take as much as 24 months, and by assuming best-case durations, a project might take as little as six months.

During my 40-year career, I have never met a leader who overestimated surprises. Although surprises are an everyday occurrence, they rarely are analyzed. Yet the implications of surprises are unacceptable in projects where there are painful consequences for missing deadlines. Something has to give.

The easiest solution is to separate deadlines from the intrinsic variability in a project's duration. For example, a leader could select the project's goals such that the most likely duration was 30 months, while the actual deadline was 40 months. The extra 10 months becomes the padding margin for surprises.

Project leaders study the technical details in schedules, but they seldom study the psychology of project schedules.

Using statistics, the leader might then compute that there is a 90 percent chance of meeting this deadline. But with a deadline of 30 months there is only a 50 percent chance, and with a deadline of 20 months there is only a 10 percent chance.

Expectations determine lateness. With financial budgeting, we never plan to spend the last penny, and with time budgeting, we should never plan to need the last hour. If you want to meet a deadline reliably, add margin to the schedule; the project will always complete early.

This simple approach fails because it works against the natural psychology of fear as a source of motivation. Publish a schedule with 10 months of margin, and the staff is likely to take long lunch breaks, to arrive late after a relaxed breakfast and to socialize when they should be working. Conversely, if the schedule shows that the project is modestly late before it begins, the staff is motivated by a sense of urgency to work diligently. Because efficiency improves with an immediate deadline, being late becomes a virtue, not an indication of failure. But if the project is hopelessly late, the staff gives up because there is no possibility of meeting the deadline.

Some enlightened leaders use a hybrid approach: a public schedule without padding (to manage psychology), and a private schedule with padding (to manage reality).

There are other ways to compensate for surprises. Separate a project's requirements into an ordered list of priorities.

In the example of the studio project, sort tasks into Phase 1, containing only the minimal requirements to go on air, and Phase 2, containing those extra requirements that make the studio cosmetically complete. A studio is still functional without permanent lighting and dressed cables. Depending on when Phase 1 actually completes, some, all or none of the requirements in Phase 2 can be completed before the deadline. The project may still be late, but the painful consequences of missing the deadline have been avoided. Using this approach, a leader compensates for surprises by adjusting dynamically how many low-priority tasks will be completed.

THE DANGER OF THRASHING

One reaction to surprises, called thrashing, guarantees that a project will be very late. Thrashing occurs when the staff works hard and appears to be making progress, but the task is never actually completed. This occurs if the requirements contain hidden contradictions — there is no possible solution — or if quality expectations are unrealistically high; perfection is the enemy of "good enough."

A thrash-detector should sound an alarm when a task is not approaching closure at a rate consistent with the effort being expended.

As an illustration of thrashing, consider the task of installing a cable between two studios. After installation, engineers discover that it passes through a region with inordinately high power-line fields. The hum is an unacceptable –80 dB while the specification calls for –100 dB. The cable is then re-installed along a second path, but excessive RF coupling produces other problems. Again, the cable is re-installed, but the third path is obstructed by an impenetrable wall. Finally, a specialty cable is ordered, but it has an unknown delivery time.

Repeated attempts to find the perfect cable installation, which does not exist, produces thrashing: doing the same task multiple times.

There are also examples of contradictory requirements, such as searching for a full-featured processor that is also small, simple, reliable and inexpensive. One could look forever. Sometimes the remedy for thrashing involves making the task disappear, either because it is not really needed, or because there is a simple work-around.

Projects will be completed on time if the leader has an appreciation for surprises. While the specifics of surprises can never be known beforehand — else they would not be surprises — an experienced leader anticipates their existence. He also recognizes the psychology of deadlines, pads the schedule with margins, sorts task priorities before beginning and intervenes when tasks are thrashing. Together, these simple approaches work like magic for managing a schedule to meet a deadline. ■

More.

The Wheatstone GENERATION-5 has the POWER and FEATURES Stations Demand the MOST

LOTS More!

Our Generation-5 provides your operators with a straightforward traditional control surface coupled with all the benefits of digital technology. It gives you the flexibility of system-wide source, mix and destination control (any signal any*where*), a powerful mix-minus section and a complete event store, name and recall system. One wire from this surface can control THOUSANDS of wires in your technical operations center.

And while the G-5 *feels* like an analog console, its DSP-based mixing engine keeps your digital sources digital while converting analog sources to switched digital, eliminating crosstalk and noise. It can furnish remote and telcom functionality on any input fader without fear of feedback—a real plus in back-to-back daily operations. Its built-in graphic displays keep operators on top of things with just a glance. And since the entire system is software based, you can accommodate any format with a press of a button.

Like all our Generation Series consoles, the G-5 has complete failsafe options available, such as automatic fail-over DSP and CPU cards and redundant power supplies. We can even provide scheduling software and studio mounted satellite cages that can be configured to mix independently from your main routing system.

*At **WHEATSTONE** we've built and sold over a thousand digital audio consoles.* The G-5 is a culmination of all that experience. *Benefit from our expertise — choose **WHEATSTONE**!*

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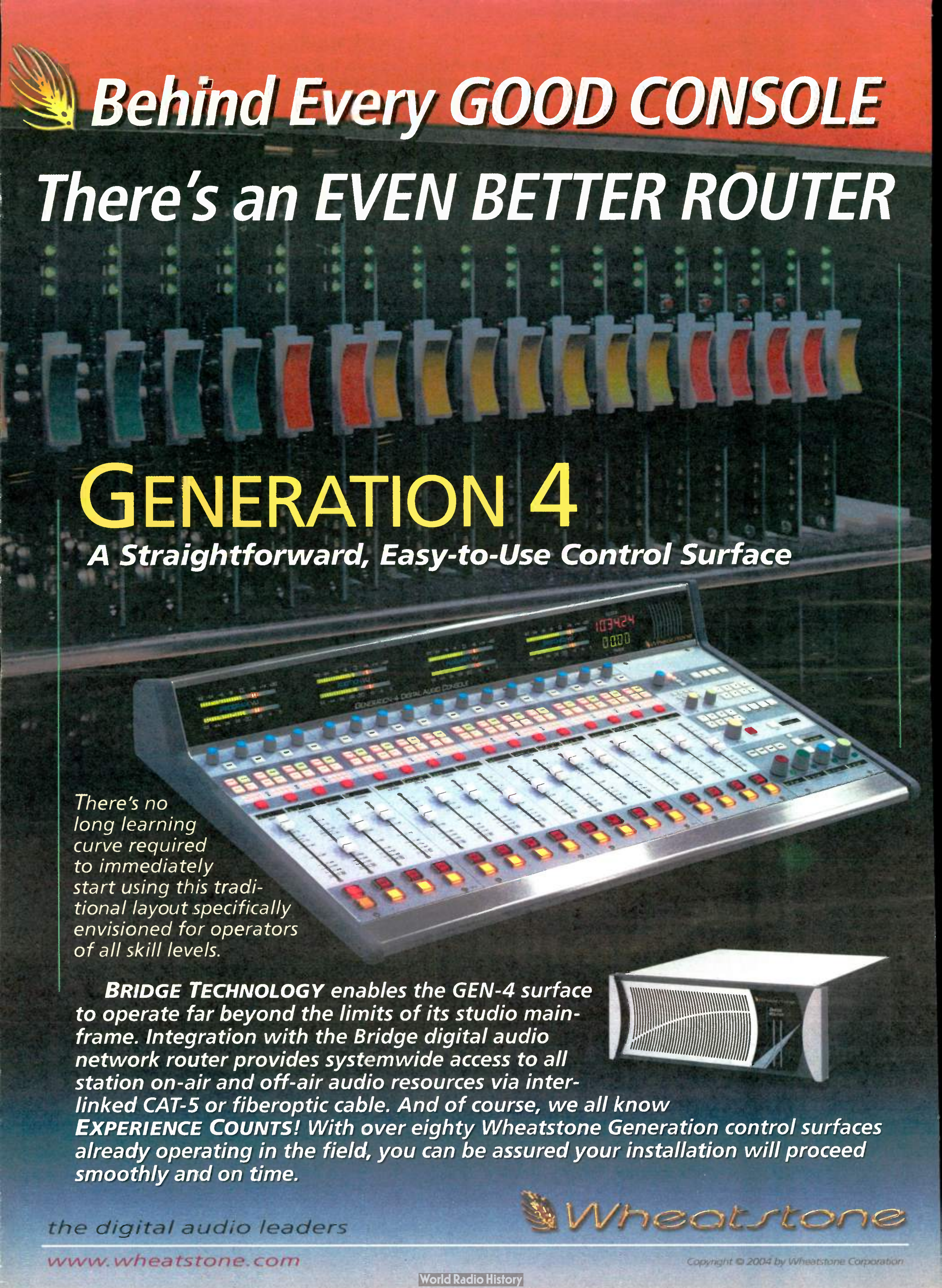
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There's an EVEN BETTER ROUTER
GENERATION 4
A Straightforward, Easy-to-Use Control Surface
There's no long learning curve required to immediately start using this traditional layout specifically envisioned for operators of all skill levels.
BRIDGE TECHNOLOGY enables the GEN-4 surface to operate far beyond the limits of its studio mainframe. Integration with the Bridge digital audio network router provides systemwide access to all station on-air and off-air audio resources via interlinked CAT-5 or fiberoptic cable. And of course, we all know EXPERIENCE COUNTS! With over eighty Wheatstone Generation control surfaces already operating in the field, you can be assured your installation will proceed smoothly and on time.
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