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DESIGNER INTERVIEW

Tom Hartnett

He Takes Radio On the Run

Tom Harnett of Comrex Talks About the Challenges And Benefits of Sending 7 kHz Audio via a GSM-Equipped Cellphone Carrier

By Thomas R. McGinley

Since the advent of cellphones in the 1980s, engineers have wanted to seize their many advantages for remote broadcast applications. Until recently, the problems have been poor quality and the difficulty in interfacing with broadcast equipment.

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Understand Ground Loop Problems

By Bill Whitlock

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

Transferring audio signals from one box to another may seem trivial, but when it comes to noise, every signal interface is truly a danger zone! Signals accumulate noise as they pass through system equipment and cables. Once noise contaminates the signal, it's essentially impossible to remove it without altering or degrading the original signal.

Therefore, noise coupling must be prevented along the entire signal path. Because equipment ground connections have profound effects on noise coupling at signal interfaces, we must appreciate how interfaces actually work, as well as when, why and how equipment is grounded.

Many designers and installers of audio systems think of system grounding and interfacing as a black art. How many times have you heard someone say that a cable is "picking up" noise, presumably from the air like a radio receiver? Even equipment manufacturers often don't have a clue as to what's really going on and simply blame problems on "bad grounding." The most basic rules of physics are routinely overlooked, ignored or forgotten. As a result, myth and misinformation about grounding are epidemic.

In this, the first of a two-part series, we will explore in detail the effects of circuit grounding in audio systems and the problems that can result.

NOISE AND DYNAMIC RANGE

The dynamic range of an electronic system is the ratio, generally measured in dB, of its maximum undistorted signal output to its residual noise output or noise floor. Listeners will generally describe a 10 dB level decrease or increase as half as loud or twice as loud respectively, and a 2 or 3 dB change as just noticeable. Up to 120 dB of dynamic range may be required in high-end "audiophile" sound systems installed in typical homes.[1] (Footnoted references are found at the end of the article.)

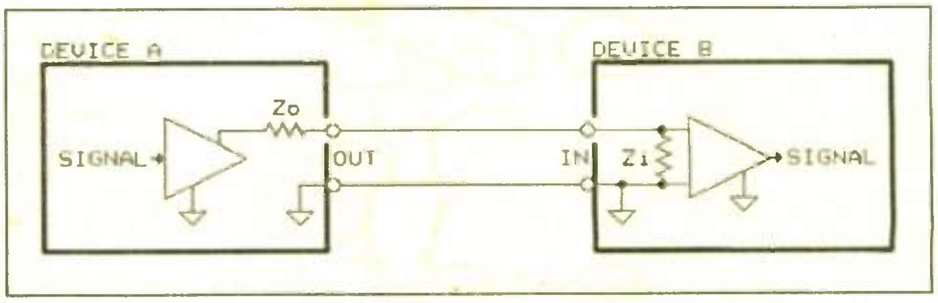

Fig. 1: Basic Unbalanced Interface

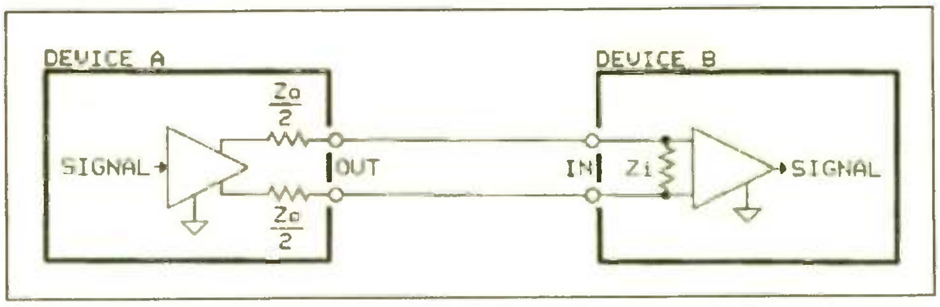

Fig. 2: Basic Balanced Interface

Of course, a predictable amount of random noise is inherent in all electronic devices and must be expected. It manifests itself as hiss in audio systems. While usable dynamic range can be limited by random noise, other noises generally are much more noticeable and irritating. In this paper, the term "noise" will mean non-random noise coupled from the AC power line, or interference from other electrical or electronic systems, heard as hum, buzz, clicks or pops in audio systems.

INTERFACE CIRCUIT BASICS

Electric current will flow only in a complete circuit that consists of a source of voltage (the electromotive force) and a load. Current from a source must always return to that same source. This law of physics is often forgotten as soon as the nebulous term "ground" is mentioned. But electrons obey physical laws ... and do not read schematic diagrams!

Of course, impedance determines how much current will flow for a given applied voltage. Impedance is the total apparent AC resistance of a circuit that contains both resis-

GROUNDING, PAGE 17

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From the Editors

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RW-EE: And Now, Something Extra

By Michael LeClair

Michael LeClair

Welcome to Radio World Engineering Extra. This publication is for everyone interested in the technology of radio broadcasting and a curiosity to know more about it. In our bimonthly issues we will cover topics ranging from the latest on digital radio to the basics of broadcast engineering.

We'll be hearing from the leading engineering voices in our industry with a new series of designer interviews to give you an inside look at current and future broadcast technologies. And we'll be hearing from you, the working engineer, with accounts of what it is really like to be out there facing the daily challenges of radio broadcasting.

Engineering Extra is designed to go deep into the technology and offer detailed understanding of the engineering that makes our business work. We won't be afraid to offer graphs, diagrams and even (deep breath) mathematics to explain the "how" behind the technology.

Some of our pieces will be long in order to explore a complex subject fully, such as Bill Whitlock's paper on studio grounding in this issue. This gives us the chance to dig deep and deliver more of the background information that helps you in your daily work.

> **Radio World Engineering Extra is designed to go deep into the technology and offer detailed understanding of the engineering that makes our business work.**

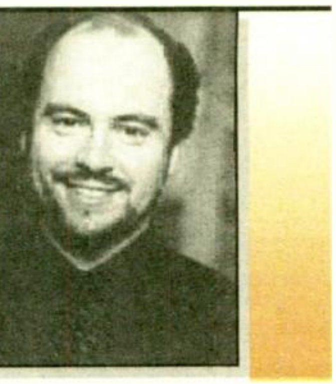
We also want to bring you stories from the field. We'll have white papers, like the article on IBOC implementation from Bill Harland of ERI; and we'll chat with innovators, like Tom Hartnett of Comrex.

As the technical editor of Engineering Extra, I will be reviewing the latest technical papers, talking to leading equipment manufacturers and working with the industry's best writers to bring you each issue.

Getting a new publication off the ground has been an exciting process. As we have planned and assembled this opening issue over the last few months, it has been both a whirlwind of activity and, frankly, a lot of fun. I hope you enjoy each issue as much as I do. ■

New Content From Familiar Friends

By Paul J. McLane



Paul J. McLane

"I think it's a great idea."

That's a phrase I've heard often lately, after explaining to readers and clients why we've launched this new edition of Radio World.

In a way, RW-EE brings us full circle.

At one time, Radio World appealed only to engineers. But as the nature of radio engineering has changed, so has Radio World.

Long-time supporters of our newspaper know that purchasing and technical decisions now are made by both engineers and non-engineers, techies and non-techies. Radio World, 28 years old in 2005, is "the newspaper for radio broadcast managers and engineers"; it does a superb job of covering our industry in a way that is informative to both audiences. It is the oldest — and still the best — in the IMAS Publishing Inc. family of publications.

But as the editor, I often am faced with a conundrum. Perhaps I would like to cover an important topic that cannot be reported in a typical article of 900 words and one graphic. Perhaps a great, detailed technical report — one I know you'd love to read — lands on my desk, but it is too long to publish and too daunting for some RW readers.

Many readers have asked me for in-depth discussions — stuff that an engineer would like to sit down with over a cup of coffee, but that might cause other managers to put the paper aside.

This "deep tech" edition of Radio World allows us to offer a service to our core technical audience without sacrificing the special mix of content that sets Radio World apart. Our goal is to provide stories and analysis in greater technical detail than is now possible in any publication. Further, this edition is available to readers both in and outside of the United States.

Radio World itself will continue to offer content for the key decision-making audiences in the radio broadcast industry including engineering, management, ownership, production, programming, suppliers and regulators.

I'm delighted to introduce Michael LeClair, CPBE, as technical editor of the new edition. Michael is chief engineer of the four-station WBUR Group based in Boston and performs contract engineering for other stations. He has 26 years of experience in broadcast engineering. He is a long-time contributor and adviser to Radio World and has worked closely with me as a Radio World "Cool Stuff" Award judge and colleague.

His background includes work in numerous disciplines including the design and installation of digital and analog studio systems, computer networks; transmitter systems including digital HD Radio, analog FM and high-power AM directional arrays; automation; wide-area networks, regional radio networks and satellite systems; telco; architectural, electrical and HVAC planning; department management; budgeting; training; and legal compliance. He has written for Radio World about equipment, new technology trends and IBOC deployment.

Please tell your friends about Radio World Engineering Extra. It is free to radio engineers and qualified technical personnel; note that you must sign up separately to receive it. You can do so at *www.rwonline.com/eng-extra*.

And let us know what you think, and how we can make this publication better. Write to me at *radioworld@imaspub.com* and to Michael at *mleclair@wbur.bu.edu*.■

White Paper

Planning Factors for FM IBOC Operations

An Overview of the Implementation Techniques for Simulcasting IBOC Digital Radio With FM Analog Signals

By Bill Harland

The author is director of marketing for Electronics Research Inc.

The implementation of simulcast operation of an IBOC digital signal combined with an FM analog signal requires a number of careful considerations. The general problem is to combine the two signals, the analog FM and the digital carriers, which both occupy the same spectral band and need to be transmitted together.

It should be understood that the FM station is obliged to make any modifications that may be required to its existing facilities to maintain its currently allocated effective radiated power (ERP), and that the ERP of the IBOC digital signal is 1 percent (-20 dB digital-to-analog ratio) of the FM station's licensed analog FM ERP. It should be stressed that the implementation of IBOC broadcasting is site-specific and a number of parameters need to be considered. (See Fig. 1.)

There are three primary methods of implementing simulcast analog FM and digital IBOC operations, namely using:

- A common transmitter and a common antenna
- Separate transmitters and a common antenna
- Separate transmitters and separate antennas

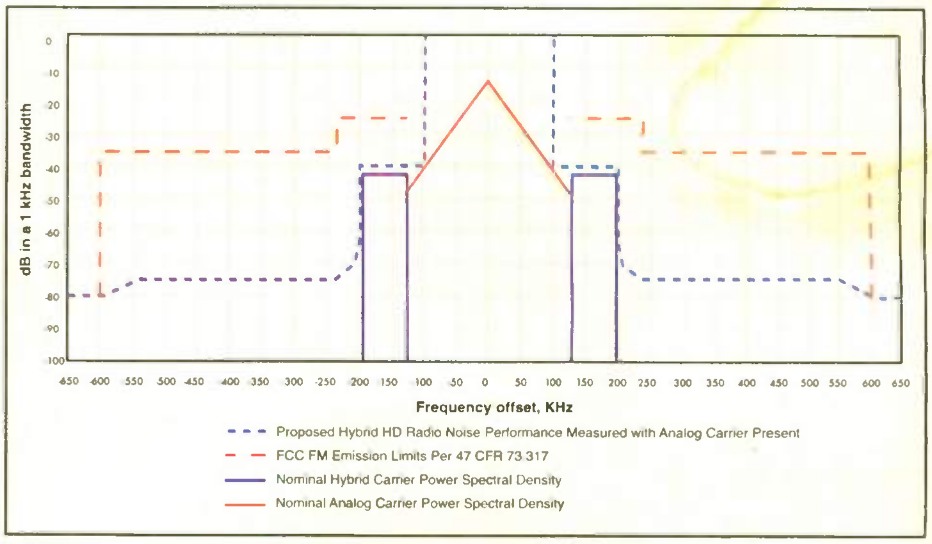


Fig. 1: Composite HD Radio Waveform

The major consideration in selecting a method is the impact on the system design as it relates to:

- Uniformity of signal coverage in the desired area of service;
- Isolation between the analog FM and IBOC transmitters, as this relates directly to the likelihood of spurious and other out-of-band emissions that are undesirable;
- The amplitude and phase performance of any filters and combiners used in the system;
- The insertion loss of any components used in the system;
- The cost of the equipment, the installation of the system and the cost associated with its operation and maintenance.

METHODS OF IMPLEMENTATION

In the most basic terms the choices available to implement simulcast analog and digital FM broadcasts are defined as low-level and high-level combining.

Low-level combining

This involves using separate analog and digital IBOC exciters that are combined and amplified in a single power amplifier. This approach has the advantage of being simple in terms of using a common transmitter to generate both the analog and the IBOC digital signal. This in turn is connected to a single transmission line and antenna. (See Fig. 2 on page 6.)

However there are other considerations:

- The IBOC signal requires a power amplifier with much greater linearity than is required for a conventional analog FM signal. These requirements dictate a transmitter using a solid-state, wideband, low-efficiency power amplifier. Hence, it will be substantially more expensive to purchase and to operate than a traditional FM transmitter. At higher power levels a common amplification transmitter design would be too expensive for most stations.
- Another consideration with common amplification is that the mixing of the analog FM signal and the IBOC signal will generate spurious signals that will require a mask filter to comply with FCC out-of-band emission requirements. This mask filter adds cost that further contributes to the high price for transmitters capable of generating both the analog FM and IBOC digital signals.

High-level combining

All approaches to high-level combining of the IBOC digital signal with the analog FM carrier involve separate transmitters for each signal. The two transmitters are then combined into one or two transmission lines that can be fed to either a common antenna or separate antennas.

It should be noted that using separate antennas to broadcast the analog FM and IBOC digital signals, either from two antennas mounted on the same tower or from antennas mounted at different locations, is not currently authorized by the FCC.

The types of practical high-level combin-

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Simulcasting

CONTINUED FROM PAGE 4

ing techniques can be categorized as:

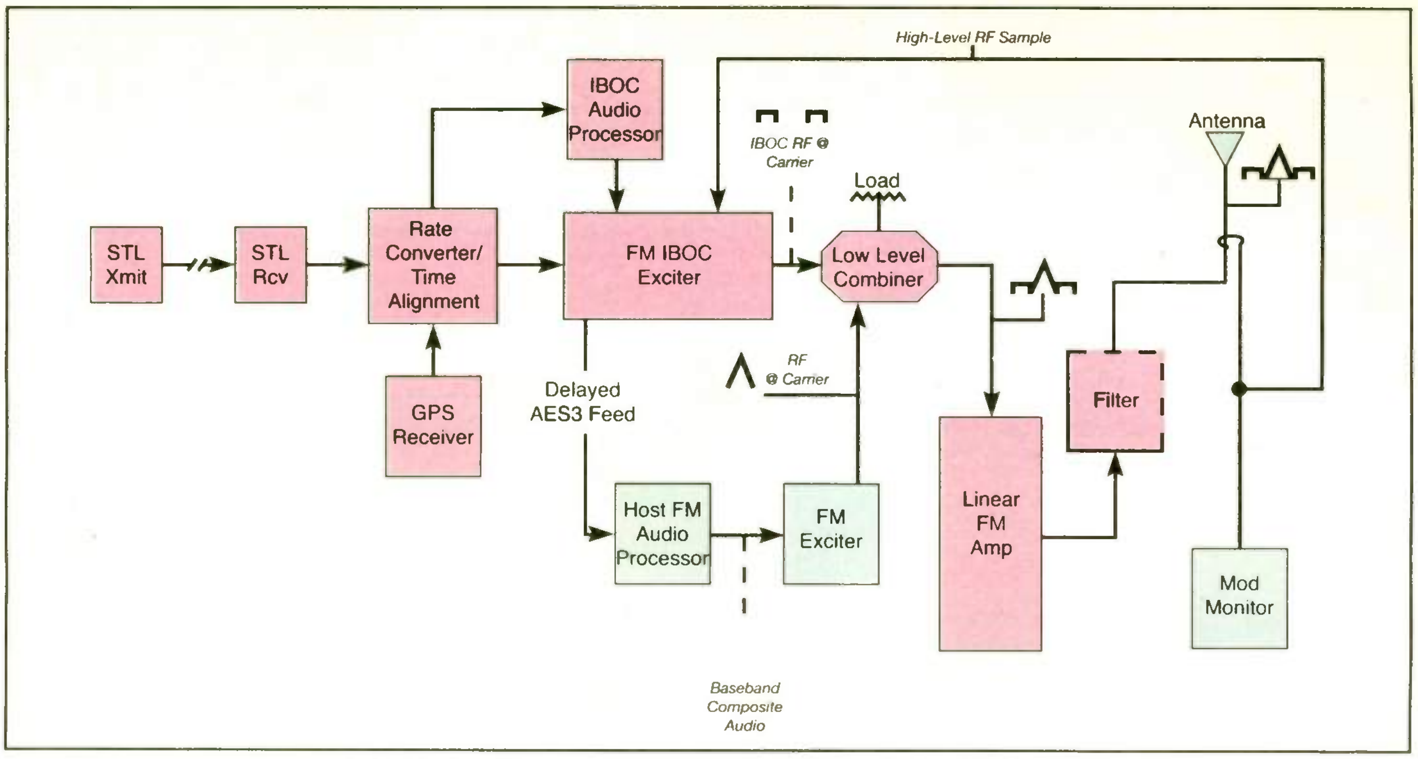


Fig. 2: Block Diagram of a Typical Common Amplification Digital IBOC and Analog FM Transmitter System

- Hybrid combiner, which includes:
    - ◦Injection combining that includes a designed-in loss in both the analog and digital signals. This is also referred to as 10-dB hybrid combining.
    - ◦Low-loss hybrid combining, which makes use of special antenna feed systems that include combining with dual input antennas.
- Free-space combining with separate antennas, which will be discussed briefly even though this method is not approved by the FCC.

Fig. 3: ERI IBOX Medium-Power Hybrid Combiner

INJECTION (10-dB) COMBINING

In this application an injection combiner, such as the ERI IBOX Hybrid Combiner, is fed by the analog FM transmitter and the digital IBOC transmitter. (See Figs. 3 and 4.) The IBOX combines both signals into a single transmission line and FM antenna. This method has a number of advantages:

- There is virtually no distortion to the analog FM or digital IBOC signals.
- There is an excellent input match and very robust isolation between the analog and digital transmitters. This is achievable in a broadband design so units can be manufactured and shipped off-the-shelf.
- This method uses the existing antenna and transmission line.
- This method can also be used with the individual ports of a multistation combining system, without modifications.
- The injection combiner is low cost and its small size makes installation easy in the existing transmission line system.

As always, these advantages do not come without a price. There are some disadvantages to this method of combining the analog FM and digital IBOC signals. Those include:

- The injection combiner throws away 10 percent of analog FM transmitter power and 90 percent of the digital IBOC transmitter power.
- The addition of the 10 percent loss of the analog transmitter output power to the antenna will require either increasing the operating transmitter output power for the existing transmitter by 11 percent, or that an analog transmitter of higher power capability be purchased. The power consumption of the analog transmitter will also increase by 11 percent.
- The digital IBOC transmitter in this configuration must be capable of generating 10 times the output power required to achieve the required ERP for the digital IBOC signal. This will require that the digital transmitter be more expensive to purchase initially and more expensive to operate.
- The power lost in the injection combiner is dissipated into a reject load, and that heat is generally dissipated within the transmitter building. This may also increase the cost associated with operating the transmitter building cooling system. It should be noted that the reject load should be rated to operate well above the absolute required rating to protect against human exposure to hot surfaces, and care should be taken to observe the load manufacturer's periodic maintenance requirements.

LOW-LOSS HYBRID COMBINING WITH DUAL-INPUT ANTENNAS

This method utilizes the hybrid combiner that is part of most standard FM panel antenna systems, and can be implemented for both single-station operation or in systems where several stations are multiplexed into a single-panel antenna array.

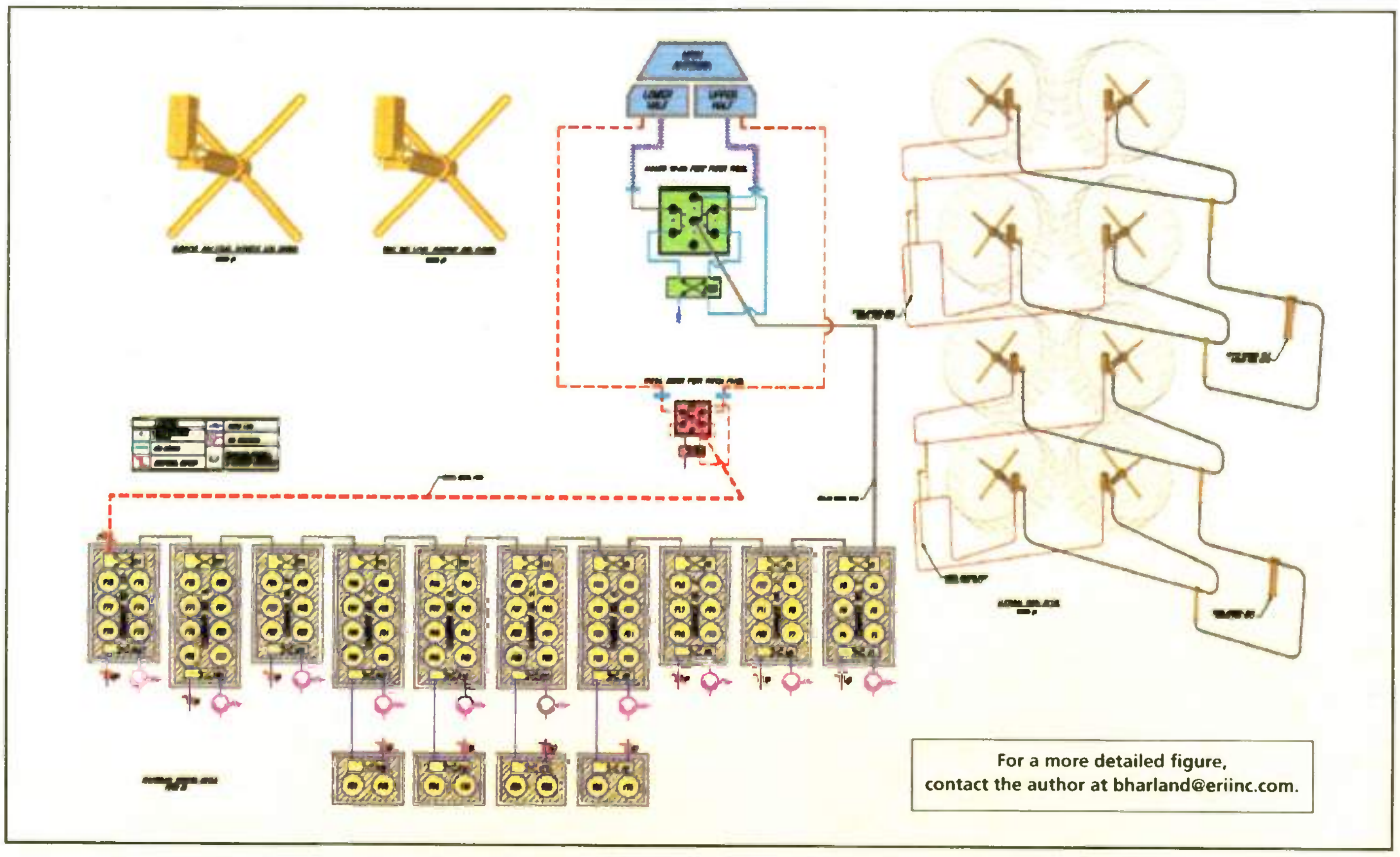
The benefits to this approach:

- This is a low-loss combining method. There is no 10-dB loss in digital path and no additional 0.46-dB loss in analog path.
- The same array elements are being used to radiate both analog and digital signals.
- All existing panel arrays have potential to be retrofitted for dual-input capability.
- There is no additional antenna, meaning no additional use of tower aperture and only a small additional load on tower.
- There is the potential ability to use the existing combining facilities for multiplexed sites with reverse-feed, constant-impedance combining system.
- There is the flexibility to use separate low-power combining systems for digital signals if this is desired.

The drawback to this implementation method is that this is an expensive approach for small markets and/or sites that are not already multiplexed facilities or otherwise have hybrid-fed antenna elements.

The practical implementation of this method of simulcast operations is generally done in one of two ways:

The individual analog FM and digital IBOC signals are fed to the antenna system through a hybrid combiner or power split-



For a more detailed figure, contact the author at bharland@eriinc.com.

Fig. 5: Typical Implementation of Dual-Input FM Panel Antenna With Reverse-Fed Constant-Impedance Combiner

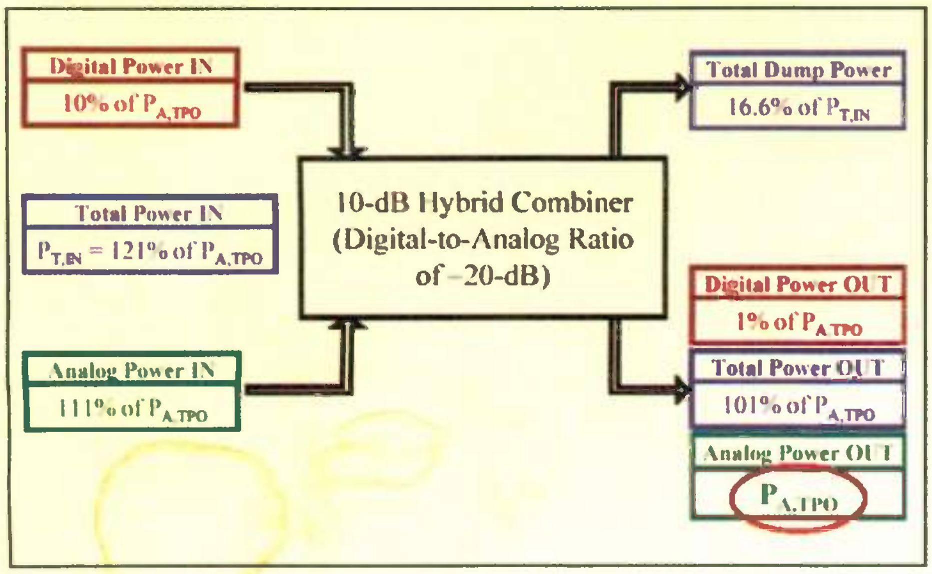


Fig. 4: Relative Power Requirements for IBOC Hybrid Combining

ter feeding separate transmission lines and driving the individual vertical and horizontal radiating elements. In some cases, the panel antenna elements contain individual hybrids at each element that perform the same function as the common hybrid. If there are multiple FM stations multiplexed into the same antenna, those stations may be configured with digital and analog combiners prior to the antenna system inputs. It should be noted that in many cases a circulator or mask filter will be required to provide adequate isolation between the analog FM and digital IBOC transmitters. (See Fig. 5 on the facing page.)

The other technique that may be applicable in multiplexed antenna systems that have constant-impedance combiners is to reverse feed the digital IBOC signal into the dump (reject) load port of the individual station combiner modules. The filter system combines the digital signals for each station in the system together and they appear at the broad port of the combiner. This technique requires a dual-input antenna to combine the analog FM and digital IBOC signals.

This method has the benefit of using the existing combiner system. Most constant-impedance combiners can be retrofitted for this mode of operation. The use of the broad port for emergency operation or expansion is lost in this scheme and there will also be a requirement for a circulator or mask filter to provide adequate isolation between the analog FM and digital IBOC transmitters. (See Fig. 6 on page 10.)

FREE SPACE ANTENNA COMBINING

The FCC recently approved the use of separate antennas for the analog FM and digital IBOC simulcast operations. It merits a brief discussion as there are ongoing experiments into the feasibility of this method.

The basic approach is to use individual antennas with one radiating the analog FM signal and the other radiating the digital IBOC signal. The combining of the two occurs in free space.

The FCC requirements for separate antenna implementation are:

- The digital transmission must use a licensed auxiliary antenna;
- The auxiliary antenna must be within 3 seconds of latitude and longitude of the main antenna; and
- The height above average terrain of the auxiliary antennas must be between 70 percent and 100 percent of the height above average terrain of the main antenna;
- Both antennas must be nondirectional.

There are a number of advantages to this approach:

- As with the dual-input antenna combining method, this is a low-loss combining technique. There is no 10-dB loss in digital path and no additional 0.46-dB loss in analog path.
- It is possible to achieve very high isolation between the analog and digital transmitters with very little separation between arrays (approximately 10 feet between closest array elements).
- There is the potential to use an existing antenna if available (such as an existing auxiliary antenna).
- The flexibility to use separate low-power combining systems for digital IBOC signals if desired.

The negative aspects to this approach are:

- If an existing auxiliary antenna is not used, the additional antenna needed will add to tower loading and take up vertical aperture if vertical separation is used to provide the necessary isolation.
- Consideration needs to be given to possible variations in coverage due to phys-

SIMULCASTING, PAGE 10

Answers to Your Questions About FM Antennas

By Richard J. Fry

The following responses were prompted by questions sent to me online and in other forums. What's your question? Write to the address at the end of the article.

Q. I'm having trouble serving some distant markets. Using my present transmit site, how can I improve my signal strength there? Should I use more height, more bays, less bays … ?

A. This situation is likely to be more about the propagation path than the number of bays. Whether using eight bays or one bay to produce a given nominal ERP, if antenna azimuth patterns are the same in/near the horizontal plane, then the ERP toward distant markets near the radio horizon also will be the same, so field strengths there would not change.

> **No matter how many bays, or what their spacing is, an FM transmit antenna might have a 10-15 dB null in its azimuth pattern when side-mounted on a fairly large cross-section tower.**

Of course, raising the transmit antenna radiation center is always good, if that is an option. In the United States, exceeding the maximum antenna height for the class of station means that the maximum ERP must be reduced, in order to maintain the same 60 dBµV/m contour as achieved at the maximum antenna height. But the higher antenna also means better path clearances for the direct ray and its Fresnel zones, which is the reason that distant field strength is maintained even with that lower ERP. This reality may improve distant field strength in some specific areas, depending on their path profiles.

But you might also consider this:

1. Commission a propagation study for your licensed ERP, site and antenna height, over the specific terrain to your problem area. From whatever field strength is predicted there for a 2 m receive antenna height, subtract at least 10 dB to account for the losses caused by "urban clutter."

2. Measure the field strength at 2 m receive antenna height for random locations in the problem area. Determine their average agreement to the end value from Step 1. If agreement is fairly close, probably your coverage there is all you can expect for that ERP, transmit antenna height and location. If not, go to Step 3.

3. Commission a pattern study from the antenna manufacturer using two bays (at least) of your antenna type mounted exactly as you have it on a model of your tower, including ladders, conduits etc. If the measured values of the h-pol and v-pol azimuth patterns are not at or above the pattern RMS in the directions toward your problem zone, consider remounting the antenna at a different azimuth bearing on the tower, and/or adding parasitics to smooth out the pattern so as to bring ERP up to nominal in that direction.

Every dB of ERP improvement produced by bringing up the pattern gain toward the problem zone will add another dB of field strength there.

Q. What affect do antenna beam tilt and/or null fill have on FM coverage?

A. Beam tilt is sometimes specified to direct the theoretical maximum antenna gain (ERP) directly at the radio horizon, rather than slightly above the horizon — as it would be with no beam tilt.

This practice is most useful for high-gain transmit antennas installed at very high elevations. Commercial TV broadcast transmit antennas often employ beam tilt, as the antenna gain and height details for TV applications make beam tilt more useful — particularly so for UHF.

For the typical transmit antennas and antenna heights used in FM broadcast, though, the practical effect of adding beam tilt is better understood as making slight changes in the aim of a floodlight, rather than a spotlight. Therefore the difference in distant field strength between an antenna with beam tilt and one with no beam tilt is virtually immeasurable for many installations.

This point is illustrated in Fig. 1, which shows no practical difference in field strength at or near the radio horizon for any of the three antennas plotted. Most of the field strength differences due to beam tilt shown in Fig. 1 occur within about six miles of the antenna site — which is not the region of the coverage area where they usually are assumed to take place.

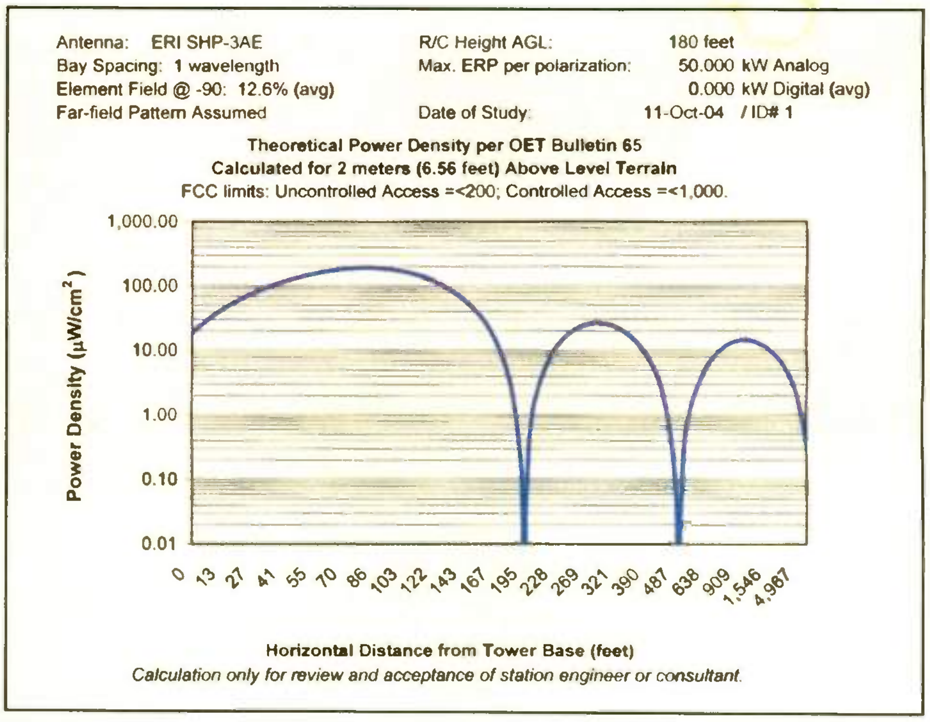


Fig. 2: Comparison of Power Densities for Two Antenna Configurations

Null fill is useful when the natural nulls in the elevation pattern of an FM transmit antenna are directed at locations where a significant number of receivers are located. This situation can occur for certain combinations of transmit site location, antenna elevation pattern and antenna installation height.

If unfilled, such nulls can cause zones of weaker-than-expected signal strength near the antenna, which may be prone to multipath distortion. Fig. 1 shows the effect on nearby field strength when 10 percent first null fill is added to the eight-bay antenna of this example.

Q. Does 1/2-wave vertical spacing of the transmit antenna bays provide improvement in distant coverage, and/or building penetration there?

A. Distant coverage is provided by radiation toward, and for several degrees below the horizontal plane. For bay counts up to

FM ANTENNAS, PAGE 10

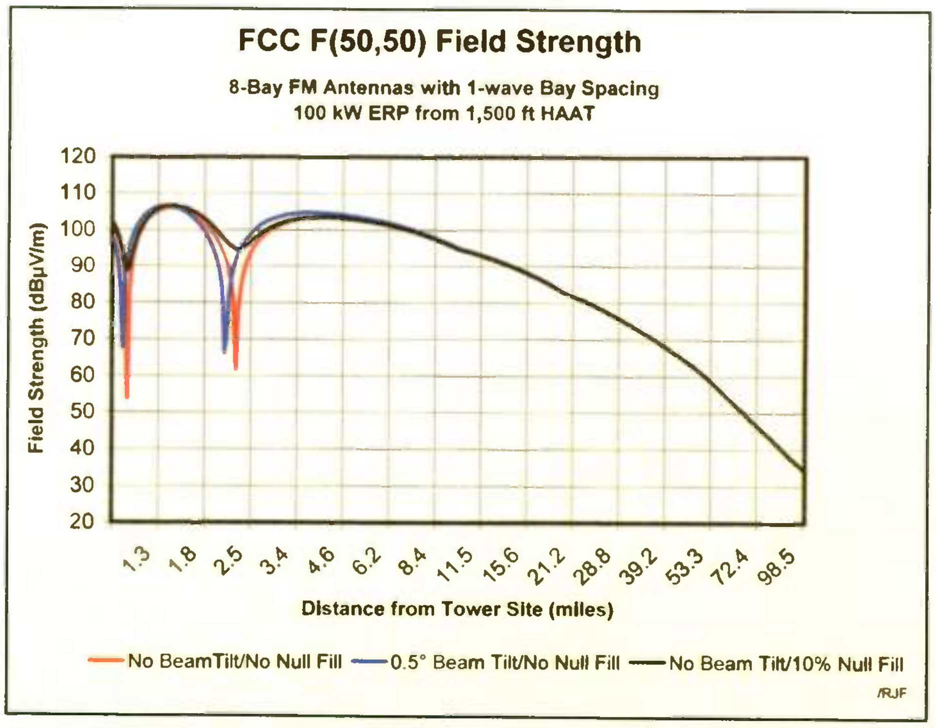


Fig. 1

JUST ENOUGH TEST

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

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

- AES/EBU, SPDIF, ADAT signals
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- Bit statistics
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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.

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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

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americas@nt-instruments.com

Simulcasting

CONTINUED FROM PAGE 7

ically separate radiating elements.

- This method does require filing for a Special Temporary Authorization (STA) for authority to operate with the digital signal.

DUAL-INPUT FM ANTENNA

The Electronics Research Dual-Input Side-Mount FM Antenna is designed specifically for FM IBOC applications. This new antenna is capable of transmitting both the analog and digital FM signals without requiring a high-loss hybrid combiner or the use of a circulator to attain the required isolation between the digital and analog transmitters. Low-power isolators may be specified based on site-specific considerations, for added protection to the digital transmitter or for multiplexed operations. The design meets the current FCC requirement for informal notification of IBOC implementation.

The antenna is a true dual-input antenna that excites all radiating elements with both analog and digital signals. The design allows the use of a single antenna while eliminating the combining loss as is present in the 10-dB hybrid combining method. A feature of the antenna is the ability to achieve in excess of 30-dB isolation between analog and digital inputs without using an isolator/circulator — no analog signal lost to a reject load and no additional insertion loss from a circulator. Since the same elements are used for both analog and digital signals, both formats have the same horizontal and vertical patterns, and therefore, the same gain.

SUMMARY

There are many factors to consider in planning and implementing simulcast analog FM and digital IBOC operations. There is often more than one implementation method available at any one facility.

Combining technology is evolving and the development of efficient combining methods, specifically improvements in dual input antennas, continues. ■

Electronics Research Inc. is a supplier of broadcast and telecommunications equipment in Chandler, Ind.

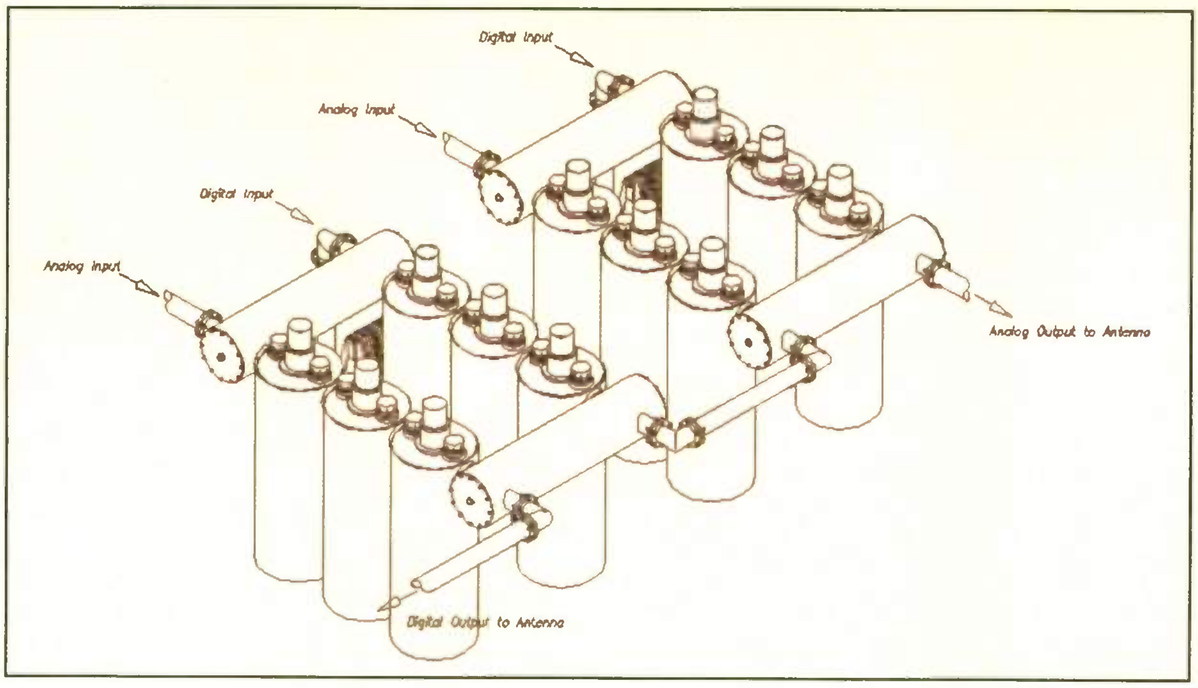


Fig. 6: Reverse-Feed Constant-Impedance Combiner

FM Antennas

CONTINUED FROM PAGE 8

at least eight, the main lobes of the elevation patterns of FM transmit antennas typically are within a dB or so of each other in this region, whether the antenna is full- or 1/2-wave spaced. So if the azimuth patterns and licensed ERP are the same, any antenna configuration would radiate very nearly the same ERP toward distant sites.

"Building penetration" is purely a function of the field strength outside the building. More field strength outside = more field strength inside.

The biggest contributors beside path loss to distant field strength are the h-pol and v-pol azimuth patterns of the transmit antenna. No matter how many bays, or what their spacing is, an FM transmit antenna might have a 10-15 dB null in its azimuth pattern when side-mounted on a fairly large cross-section tower. Making sure that such a null isn't aimed at an important coverage sector is very important. That's the reason for the pattern study suggested earlier.

Q. Doesn't 1/2-wave vertical bay spacing greatly reduce the chance for blanket interference around the transmit site?

A. It can be useful, but probably the real-world blanket interference performance won't be as different as intuition might suppose.

Consider a three-bay, 1-wave spaced antenna compared to an eight-bay, 1/2-wave spaced antenna. If both systems radiate 50 kW maximum ERP from a radiation center 180 feet above level terrain, and if the antenna patterns really are those assumed for them (a big "IF"), then the fields from these two antennas will be identical beyond ~3/4 mile from the site.

In the "donut" between ~200 and ~2,500 feet from the tower, average power density peaks at 2 m above level ground from the eight-bay, 1/2-wave antenna are ~7 μW/cm², and ~20 μW/cm² for the three-bay, 1-wave antenna.

From zero to ~200 foot radius from the tower, the eight-bay antenna peaks at ~4.5 μW/cm², and the three-bay at ~200 μW/cm². That's the large difference that intuition expects, but probably also it's a lot closer to the tower than expected. Chances are that not too many listeners are located there.

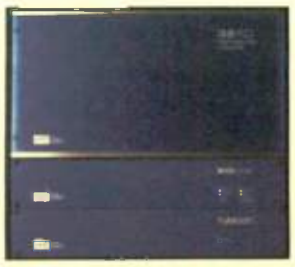
Fig. 2 on page 8 shows the two plots from which the above values were taken. Power densities were calculated for the OEM's published elevation patterns, and a perfectly omni azimuth pattern for a popular c-pol transmit antenna design. ■

Richard J. Fry, CPBE, is an RF systems analyst and retired FM applications engineer. Send him your questions to rfry@adams.net. Also visit http://rfry.org.

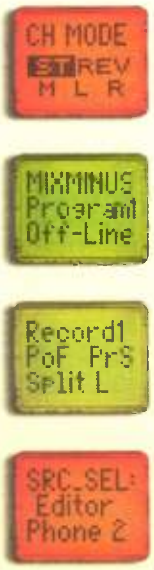
AT LAST, A BOARD GEEKS AND JOCKS CAN BOTH DROOL OVER.

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.

rubicon™

Come see for yourself Rubicon's brains and beauty, power and performance.

It'll be love at first sight.

Engineering great radio.™

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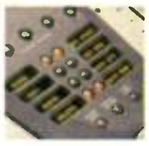
The world's best* IP-Aud

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

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.

If you need a routing switcher but aren't made of money, consider Axia, the Ethernet-based audio network. Yes, Ethernet. Axia is a *true network*. Place our audio adapter nodes next to your sources and destinations, then connect using standard Ethernet switches and Cat-6. Imagine the simplicity and power of Ethernet connecting any studio device to any other, any room to any other, any building to any other... you get the idea.

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.

AxiaAudio.com

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.

ENCO
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We're already working with some great companies. Check AxiaAudio.com to find out who's next.

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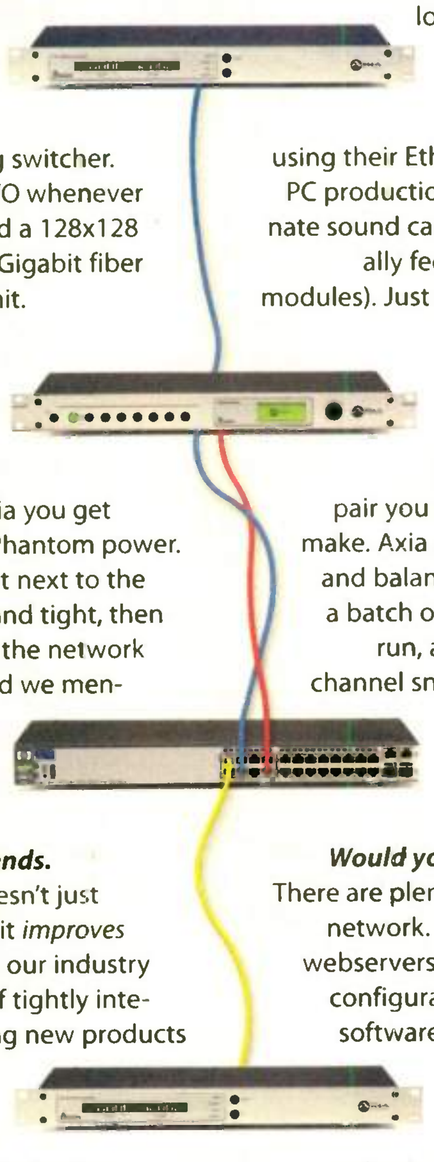
broadcast studio system.

udio broadcast studio system. Damned marketers.

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.

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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.

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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 multichannel snake. Use media converters and a fiber link for extra-long runs between studios — or between buildings.

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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.

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Guy Breaks Down the Anti-HD Radio Arguments

The Digital Bus, With HD Radio On Board, Has Left the Terminal. Be On It or Be Under It.

By Guy Wire

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

It continues to amaze me how many knowledgeable broadcast engineers and industry observers persist in calling HD Radio a flawed system not worthy of replacing analog and are forecasting its failure and demise. Online list servers seem to attract many of these unconverted.

Changing minds and opinions on this topic might be a little like trying to convince those opposed to the war in Iraq that U.S. presence there will further our objectives in defeating world terrorism. In the HD Radio proceeding, there also appear to be quite a few undecideds. Will Ibiquity's technology ultimately succeed? Somebody should run a poll to see how this stacks up. In both cases, the playing field is constantly changing.

STATUS QUO

Let's give the anti-HD Radio core positions a reality check one by one:

1. Analog still serves radio's needs very adequately. With new design techniques it can perform even better. Why obsolete hundreds of millions of existing receivers?

Lots of HD Radio naysayers tend to favor the status quo. Just because it's digital doesn't mean it's better, they say. Analog AM is still receivable on radios made over 80 years ago. Over 500 million of them still serve their owners well in this country alone. Making them unusable would be unthinkable. Yet the huge improvement of noise-free full-fidelity stereo that HD Radio brings to AM is blissfully ignored. This reminds us of all the horse-and-carriage fans of the 1890s. Proponents of the newly invented automobile back then had to prove their case.

The argument for FM HD Radio is a little less convincing. Analog FM with RBDS scrolling text on the plains of Kansas works fairly well. Think of it as a fast horse-drawn carriage on a really smooth road with a narrated travel guide.

But the ability to add features and enjoyment pretty much ends there. The riding experience, just like the listening experience for those in congested cities full of tall buildings or on very bumpy roads in hilly terrain, is not so enjoyable.

A century ago, citizens worried about all the new problems and "interference" that introducing the automobile would cause existing travelers. Vehicles with gasoline engines were noisy and caused more pollution. As more of them took to the roads, pedestrians and horse-drawn carriages were crowded off to the side. But the advantages of the automobile over horses and walking far outweighed the disadvantages. Mufflers were invented, roads were paved and widened and sidewalks were built.

HD Radio right now is like the first automobiles to travel unpaved roads. Think about what it will offer in another five or 10 years. History is squarely on the side of better technology as it pushes aside older, less efficient methods.

2. HD Radio causes new interference to adjacent-channel stations and reduces their coverage, especially AM at night. Why degrade useful reception of weaker signals?

Increased interference to adjacent channels is one of the compromises HD Radio does inflict on existing operations. The impact on the FM dial has been demonstrated as minimal. FM already tolerates similar degrees of interference in large markets and short-spaced situations.

The AM case is a bit more problematic. As more HD stations come online, some stations will probably lose some fringe area and distant nighttime service. But with less than 5 percent of all radio listening occurring on AM at night, this may prove to be more of a non-problem than many inside the industry think.

Whatever methods are used to resolve interference complaints, one aspect of this issue seems clear. For radio to thrive and remain successful, it will have to focus on serving its local and primary coverage area the best it can. The impact of the satellite services is pushing terrestrial radio in this direction anyway. Super-serving the local audience will have to become every station's mantra.

> It amazes me how many knowledgeable broadcast engineers and observers are forecasting HD Radio's failure.

3. HD Radio lacks the killer app to attract public interest in buying the necessary new receivers. Any new consumer product needs this to be successful.

Most HD Radio detractors seem convinced that consumers will have little interest in forking over several hundred dollars more to buy an HD Radio instead of the inexpensive analog models they know and understand. Without a killer application, consumers won't buy them and HD Radio will fail just like the AM stereo experience.

I suggest that this paradigm is changing.

Every new technology-based consumer product that establishes itself as a long-term success has one or more killer apps. The automobile offered more speed, versatility and comfort. The first ones were very expensive and not that reliable. It took a while before assembly-line manufacturing techniques and improved quality control made them reliable and affordable for most consumers.

Thankfully, the advanced capabilities of today's consumer electronics industry move the development of new products like digital radios along rather quickly. As the penetration of XM and Sirius satellite radios accelerates, radio manufacturers are now looking at designing multi-mode digital radios that will include all digital formats.

Integrating separate RF sections and programming the DSP engine to include HD and XM functions in the same radio is relatively easy since the codecs are virtually identical. This is a natural evolution for car radio offerings and best serves consumer needs and preferences. What proves successful as a new feature in car radios usually finds its way into home radios and portable models.

MORE TO COME

The first-generation HD Radios only offer some of the advanced features that make HD Radio a superior performer over its analog parent. Most consumers will probably not be asking for an HD model or recognize a killer app when shopping for a new radio as more gen-one HD Radios start appearing in the marketplace. Awareness of what HD Radio offers will definitely take some time. Perceptions of it will be blurred with satellite services.

Noise-free, multipath-free reception is not by itself a killer app. Neither is the scrolling radio-text feature in its present form. But as more consumers become aware that these improvements are part of the satellite radio offerings already, they will naturally come to expect the same from terrestrial radio.

Data services will not flourish in HD Radio until protocols are more fully standardized and developed to exploit that resource. TIVO for radio, the "Buy Button" and other PAD services are the tip of that iceberg. We can only dream about what shape and form future data-delivered services might take in succeeding generations of HD Radios. Certainly streaming video and fully interactive features will be among them, especially in the all-digital HD Radio format. You can bet satellite will have them.

Offering a second program service within the same RF channel will be an important new capability for radio that only digital can deliver. Public radio stations everywhere are now planning their next rounds of CPB grant funding to include HD Radio conversion mainly to be able to take advantage of this powerful addition. Commercial stations have been slow to recognize the value of a second channel, but as soon as the rules for such service are established and one large group owner moves forward with it, you can bet many more will quickly follow.

Surround sound is shaping up to be the bona fide "wow-inducing" killer app for HD Radio. Consumers are flocking to home theatre 5.1 sound as a very hot seller in video and television. Cars automatically are suited for accommodating 5.1 speaker positions. Most manufacturers are ramping up to include this feature in many new luxury car models starting next year. Selling new surround sound versions of all the hit songs of the past could be a big shot in the arm for the troubled recording industry.

4. Ibiquity's technology is proprietary and not an open architecture that can accommodate future improvements from outside developers.

While this does make it a bit more difficult for independent companies not affiliated or invested in Ibiquity to bring improvements to it, it does not preclude them. Witness Ibiquity's change of codecs from PAC to HDC. They say it's unique, but HDC is merely a multi-streaming version of HEAAC (MPEG AAC+ with SBR), the same codec used by XM satellite. Ibiquity will most certainly be looking to incorporate worthy and compelling improvements to HD Radio in the future. It is, after all, only software.

Digital Radio Mondiale proponents see that system as technically superior and more flexible than the AM HD Radio solution. The two systems are really more similar than they are different, both using the same digital modulation building blocks. Software driven platforms are continuously adaptable and tweakable.

Even though the first generation of HD technology is set, there is really nothing to stop Ibiquity from co-opting and integrating some of the DRM features or other techniques yet undiscovered into AM HD Radio as the rollout moves forward with succeeding generations. Let your imagination contemplate what might be used or discovered for improving performance and mitigating interference.

5. Unlike HDTV, HD Radio does not have a mandatory, FCC-imposed conversion schedule. As with AM stereo, "Let the marketplace decide" is no decision at all and not the directive the industry needs to make this work long-term.

This objection has suddenly become less relevant in the face of the recent FCC decision to suspend the HDTV conversion timetable. Grand scale technology changes take time, money and effort to implement. All sorts of unknowns and variables change the course of man's best intentions.

Ibiquity learned from the mistakes of previous efforts to change or improve the basic engine of radio. Unlike FM Quad and AM stereo, HD Radio formed a solid alliance of investors and partners from the beginning from across the industry including broadcasters, transmission equipment companies and the all-important receiver companies. All the players are in this game together, with a carefully devised rollout strategy and timetable, making the chance of failure a more remote possibility.

The fact that a hard FCC mandate for conversion is not in place for HD Radio may not make much difference in the end as the rollout gains traction. As more stations broadcast with the new technology and more receivers penetrate the marketplace, the public will acquire and use it. Momentum towards the necessary critical mass that will make this happen is well underway. Analog-only hardware will slowly be displaced and fall by the wayside as antiquated relics of the past.

Even though the first generation of technology is set, there is really nothing to stop Ibiquity from co-opting and integrating some of the DRM features or other techniques into AM HD Radio.

THE SELLING OF RECEIVERS

What will propel HD Radio sales more than anything else will be the simple fact that most automobile manufacturers will offer HD Radio models as an option or as the stock radio in higher-end 2006 models. That rollout accelerates and expands to most models in 2007. Radio manufacturers are focusing their design efforts for virtually all future models on DSP-based programmable platforms. They want to sell new digital models and want to supply radios based on common chipsets and see little future in supporting analog.

A similar mechanism was at work with RBDS. When it first rolled out in the United States in the 1990s, not many stations deployed it. Some that did stopped using it soon after they perceived very few RBDS receivers were being purchased or used. But thanks to its use elsewhere in the world, more and more RBDS-equipped car radios were being delivered in many European and Japanese car models here in the U.S.

In recent years, U.S. made cars started adding RBDS. The NAB now estimates about 30 percent of all car radios have RBDS and over 80 percent of all new cars are sold with RBDS. It's suddenly a hot ticket. Without even asking for it, U.S. consumers and broadcasters alike have discovered an impressive new feature that magically appeared in their car radios.

It's ironic that the satellite services are helping to accelerate the burgeoning public awareness of new digital receivers in the marketplace. Whether terrestrial broadcasters prefer to depend on analog for the long term may not matter. The rest of the world is moving forward with digital services and features in every consumer electronics appliance and device out there. Like it or not, terrestrial radio is being sucked into that vortex.

What further suggests increasing penetration of multi-function digital radios in cars that will include HD Radio is the recent move by many automobile companies to integrate some of the car's monitoring, signaling and navigation operations within the radio's DSP design itself. No longer can you easily replace a stock radio in many new cars with an after-market model without adversely affecting other electronic functions in the car.

This change seems to have crept into new car designs almost insidiously. In the near future, aftermarket car radios could fall dramatically in popularity. One of the reasons many radio manufacturers have delayed introduction of gen-one after-market and home models is the realization that OEM car radios will drive awareness and acceptance of HD Radio more than anything else. Many consumers will get an HD-equipped OEM radio when they buy a new car in the next few years without even knowing it or asking for it.

The stage is set. Radio is entering a new era propelled by new technology. It's really very simple. The digital bus with HD Radio onboard has left the terminal. Be on it or be under it. ■

Guy Wire welcomes your comments and questions. E-mail to gwire@imaspub.com

Hartnett

CONTINUED FROM PAGE 1

Digital telephony was developed and deployed in Europe first. Groupe Speciale Mobile established a wireless digital standard in 1983 that became known as Global System for Mobile Communications. GSM has since become the most widely used cellphone technology outside North America.

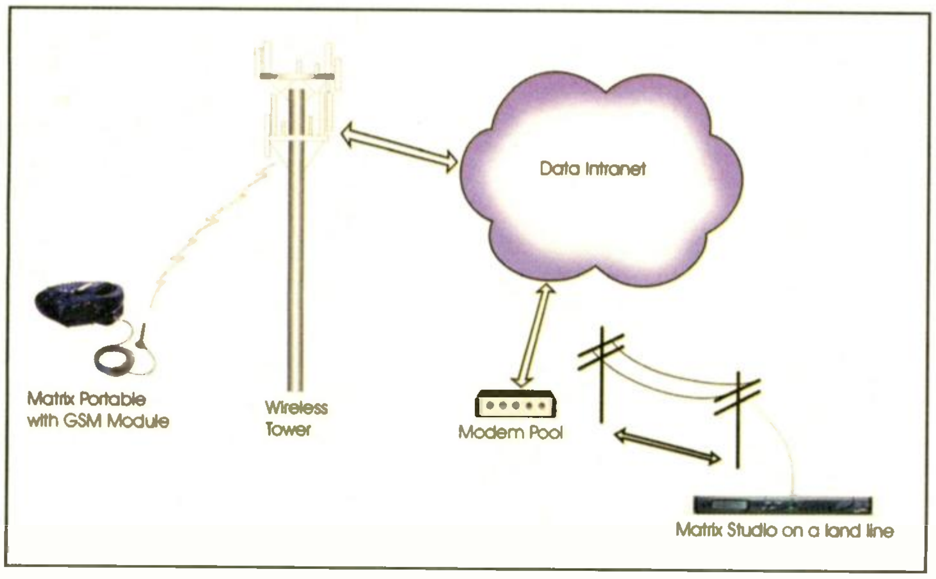

Fig. 1: GSM Remote Broadcast System

Comrex Corp., a developer of POTS codec technology, introduced a breakthrough product this spring that delivers clean 7 kHz audio via a GSM-equipped cellphone carrier. The Comrex Matrix codec, equipped with a special GSM module, provides broadcast-quality remote feeds via the T-Mobile or Cingular networks.

Achieving such performance has been a challenge on which Comrex has been working for a long time. To understand how this came about, we talked with Comrex Director of Engineering Tom Hartnett, who describes some of the background and history of the process.

"We started working on making cellphones sound better in the '80s by coupling our single-line analog extenders with analog cellphones," he said.

Analog cellphones had quickly found widespread use in broadcasting for IFB channels. A few brave souls even tried using the two-line extender on two analog cellphones to enhance frequency response to 5 kHz beyond just single-line extension for on-air broadcasts.

The challenge with digital phones is all about the data.

— Tom Hartnett

NEW CHALLENGES

Hartnett said the two-line extension idea didn't work very well.

"Analog cellphone channels had very limited bandwidth of only 2.5 kHz, plus levels could vary during the call and from call to call. Good two-line extender performance relied on 3 kHz of response with matched and stable levels on each line. Analog cellphone channels also used AGC, which wreaked havoc on our noise reduction circuitry resulting in a very unpredictable result."

That's all behind us now; analog cellphone technology is ancient history and rapidly disappearing.

To be able to use any of the predominant digital cellphone networks effectively for broadcast presents a different set of variables.

"The challenge with digital phones is all about the data. Cellphones allow for a miniscule amount of user data throughput — around 7 kilobits per second — and very powerful compression algorithms are required to put any audio at all over this small a channel. Making it sound better than a normal phone call is even more challenging."

"GSM in North America operates in either the 1900 MHz band or 850 MHz band; it works at 1800 MHz and 900 MHz in many other countries," Hartnett said.

"The Comrex GSM module is available in North American and international versions that support these bands, respectively. It is supplied with an external mag-mount antenna, which can add to the stability of a call significantly. Not only does the antenna provide more gain than a typical handheld cell antenna, it can be placed away from people and other RF obstructions for better signal strength. Since the low GSM bands are essentially half the frequency of the high bands, a single antenna is cut to cover both bands."

GSM uses a standardized Subscriber Identification Module, a smart card that resides in the phone. Once the SIM is activated it can be moved to any GSM-capable phone, such as the Comrex GSM module, in order to activate the device.

When the Personal Communications Service bands were opened up in the United States in 1993, three incompatible designs were used. Time Division Multiple Access, Code Division Multiple Access and GSM were employed, depending on the carrier.

GSM is better suited for using codecs to convey high-quality audio than TDMA because it uses Circuit-Switched Data as part of its standard, and provides a fairly steady throughput of data. CSD establishes a dedicated channel that links the segments of the signal path to create a single, unbroken line for each telephone call, thus avoiding the long delays encountered with the other protocols.

While CDMA also uses CSD, its networks transmit data in bursts rather than a continuous stream, which causes breakup in real-time audio transfers. CDMA, used by Verizon and Nextel, was optimized this way to be more efficient in the constantly varying traffic conditions encountered in large, IP-based data networks.

Hartnett said, "CDMA and TDMA are perfectly capable of moving the kind of data we need, but simply aren't provisioned by the carriers the way we need it. Broadcasters aren't power users who can dictate to a cellphone company how to provision its data channel. Usually that requires a bigger player, like a government contract."

Many of the former TDMA networks like Cingular and AT&T Wireless have migrated to GSM. But AT&T does not deliver it in a format friendly to an audio codec.

"GSM companies are more interested in delivering Internet access than Point-to-Point data, so we're using a data service they consider to be a 'legacy' product."

Luckily, many other data services also depend on this legacy provisioning. For now, T-Mobile and Cingular have emerged as the U.S. cellphone carriers that can be used for the Comrex GSM solution.

TOM'S ALGORITHM

Even though GSM allows an unbroken channel path for real-time data transfers, the data rate is still limited to 9.6 kb/s. That presents another serious challenge to being able to get higher frequency response through the channel. Comrex has had to craft the coding and error correction in its codec to be able to squeeze 7 kHz of low-distortion response through such a limited channel.

Hartnett said the big breakthrough is the use of a new algorithm, "the best in the world at these low data rates." He jokingly refers to it as "Tom's Algorithm" but will only say that it's a special mix of several standard algorithms with custom Comrex modes perfected for the application.

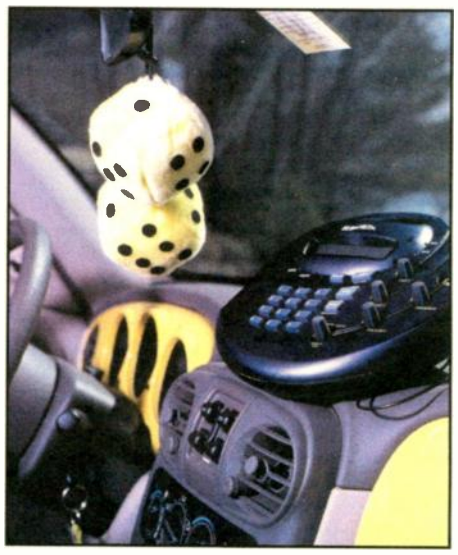

Fig. 2: The Matrix Codec With GSM Module

The other critical improvement is that the new GSM module does not use an external cellphone to establish the call. The Matrix design includes an integrated GSM phone that "has the ability to alter the error correction used in the GSM data channel resulting in a more reliable connection for audio."

Even though voice performance is good, Hartnett does not advise using it for music or as an emergency STL for AM.

"Music is pretty nasty over GSM," Hartnett said. "The algorithm actually uses a voice synthesizer like in a digital handset. I think POTS codecs make a great emergency STL, but I wouldn't recommend GSM for the job."

Using a cellphone for improved quality, remote broadcast applications involving talent requires an easy-to-use hardware package. With the optional battery pack, the Matrix/GSM Module combo qualifies as a simple "grab and run" field tool for anyone who can use a POTS codec.

"The beauty of the Matrix/Module solution," Hartnett said, "is that there's very little configuration needed."

Once the module is attached, the Matrix "wakes up" in GSM mode and acts very much like a cellphone. A call is placed via the keypad like a normal POTS codec call. The studio needs a Matrix rack or a Bluebox attached to an analog POTS line, which will answer the incoming call automatically. Talent plugs in a broadcast mic, sets the level and begins talking. Return IFB is recovered easily in headphones.

One of the common issues confronting the use of digital gear is latency or delay. Hartnett says the system itself only adds about 120 milliseconds of delay but the cellphone network can add an additional 1/4-to-1/2-second delay to that.

He adds, "It can be difficult to conduct a conversation, much like a satellite delay, but it can be overcome with a little practice."

In Europe, the use of High Speed Circuit Switched Data (HSCSD) speeds up GSM data throughput beyond 9.6 kb/s to allow frequency response as high as 10 to 12 kHz. But Hartnett says this enhancement is not likely to gain traction in the United States.

"The main constraint holding down data rates is that cellphone companies have moved to other technologies to better support Internet access and use. These are not nearly as friendly to real-time audio. It's not likely we'll see HSCSD in North America for the same reason."

Wireless telephony is now moving into the third generation of improvements beyond PCS using new protocols that can achieve much higher data rates approaching 2 Mb/s, primarily aimed at better supporting Internet performance. But Hartnett is cautious about how these might improve his new product.

"Data services like GPRS, 1XRTT and EDGE are all much friendlier to Web-browsers than to audio codecs. Whenever you utilize a network like this for other than its intended purpose, you have to make compromises. The compromise we'll likely need — to realize improvements — is a lot of added delay."

Hartnett observes this reality will impact all new data service offerings, both wired and wireless.

"The trick will be leveraging their benefits and minimizing the compromises to build a product that has value. I can't tell you much more than that, other than 'we're working on it.'"

For the near term, the most likely improvement Comrex will offer to the mobile GSM product line will be a smaller, more portable package. ■

For information on Comrex GSM products, visit www.comrex.com.

Thomas R. McGinley is a technical advisor to Radio World and director of engineering for Infinity Broadcasting in Seattle.

Grounding

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tance and reactance (capacitive and/or inductive), as is the case with virtually all real world circuits. For AC circuits, impedance is the functional equivalent of resistance.

An interface is a signal transport subsystem consisting of a line driver (an output), the line or cable itself and a line receiver (an input). When interconnected, they form a series circuit called a voltage divider. Every output has internal impedance, measured in ohms, called its output impedance or source impedance.

For practical reasons, it cannot be zero for real devices. Likewise, every input has internal impedance, measured in ohms, called its input impedance. For practical reasons, this cannot be infinite for real devices. See Figs. 1 and 2, where Zo is output impedance (split equally for the balanced output) and Zi is input impedance.

The main purpose of an interface is to deliver maximum signal voltage to the input. Since the current is the same in all elements of a series circuit, voltage drops are proportional to impedances. This means Zi should be much higher than Zo. In typical equipment, Zo ranges from 50 ohms to 1 k-ohms and Zi ranges from 10 k-ohms to 100 k-ohms. This transfers 90 percent to 99.9 percent of the available signal voltage.

Whether an interface is unbalanced or balanced depends only on the impedances of the two signal conductors with respect to ground. In unbalanced interfaces, one signal conductor is grounded while the other has a higher impedance. In balanced interfaces, both signal conductors have equal non zero impedances. More details on balanced interfaces will be discussed later.

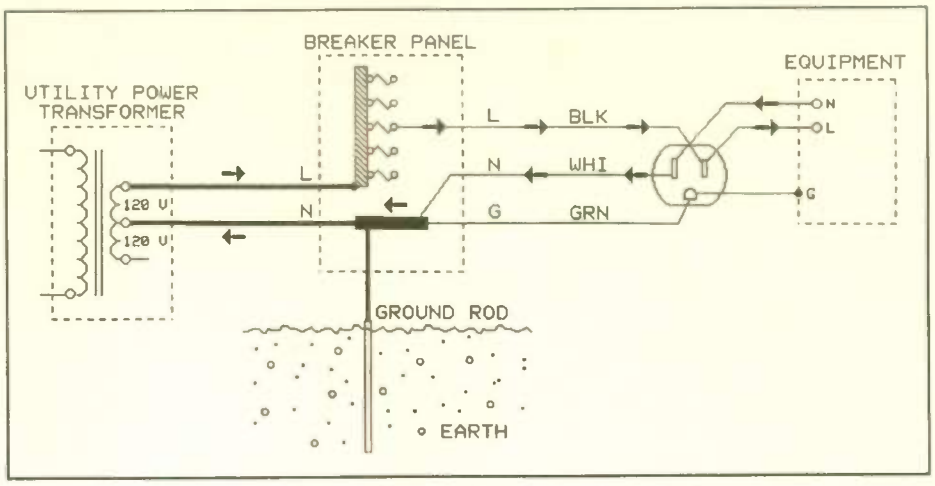

Fig. 3: Current Flow Under Normal Conditions

AC POWER SYSTEMS AND GROUNDING

Grounding has several important purposes and most often, a single ground circuit serves, intentionally or accidentally, more than one purpose.

For example, the equipment chassis may be grounded for safety reasons, but also serves as a signal reference point for internal electronic circuits. High-performance audio systems are unique in two ways: (1) the signals cover a broad, nearly five-decade, frequency range, and (2) the

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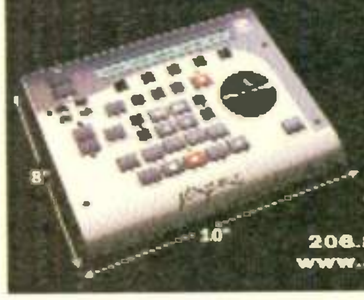

Grounding

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signals can require a very wide dynamic range. Adding to the difficulty is the fact that AC power frequencies and their harmonics also fall within the signal frequency range. At power and audio frequencies, a so-called ground loop can allow noise and signal currents to mix in a common wire. Therefore, we must understand how ground circuits work if we expect to control or eliminate noise in audio systems.

Earth grounds protect from lightning

To an electrical power engineer or electrician, a "ground" involves safety, fault currents and earth connections. To an electronics engineer, a "ground" is simply a zero-volt reference point used when referring to other voltages in a circuit, and it may not be connected to earth at all. The crucial difference in these definitions is the requirement for a real earth connection. In general, an actual earth ground is necessary only to protect people from lightning.

Before modern standards, such as the National Electrical Code (a.k.a. NEC or Code) were developed, lightning that struck power lines often was effectively routed directly into buildings, starting a fire or killing someone. Lightning strikes involve millions of volts and tens of thousands of amperes, producing brief bursts of incredible power in the form of heat, light and electromagnetic fields.

Electrically, lightning is a high-frequency event, with most of the energy centered around 300 kHz. That's why wiring to ground rods should be as short and straight as possible (bends increase inductance). The most destructive effects can be avoided by simply giving the strike current an easy, low-impedance path to earth ground before it enters a building. Virtually all modern, electric-power distribution uses lines that connect the "neutral" wire to earth ground.

Fig. 3 shows how AC power is supplied through a "three-wire service" to outlets on a typical "branch circuit" in a building. One of the service wires, which is often not insulated, is the grounded or neutral conductor. The neutral is bonded to earth ground at the service entrance as required by Code. This earth ground, along with those at neighboring homes and at the utility poles, provides the easy paths for lightning to reach earth.

Telephone, CATV and digital satellite system cables must also divert lightning energy to earth ground before it enters a building. The phone company's gray box or network interface unit, which should be mounted outside the building, provides this protection for phone lines. NEC Article 820 describes the requirements for grounding the shield of CATV lines before they enter a building. Satellite TV dishes must also be earth grounded, generally via a coax grounding block supplied with the installation kit.

Myth and misinformation about grounding are epidemic.

Whenever possible, all these other earth ground connections should be made to the same ground rod used for the utility power. Because of soil resistance, thousands of volts could develop between separate ground rods if a lightning strike should occur. This could seriously damage a computer modem, for example, because it straddles power and telephone line grounds. Remember that any wire leaving or entering a building can bring lightning inside!

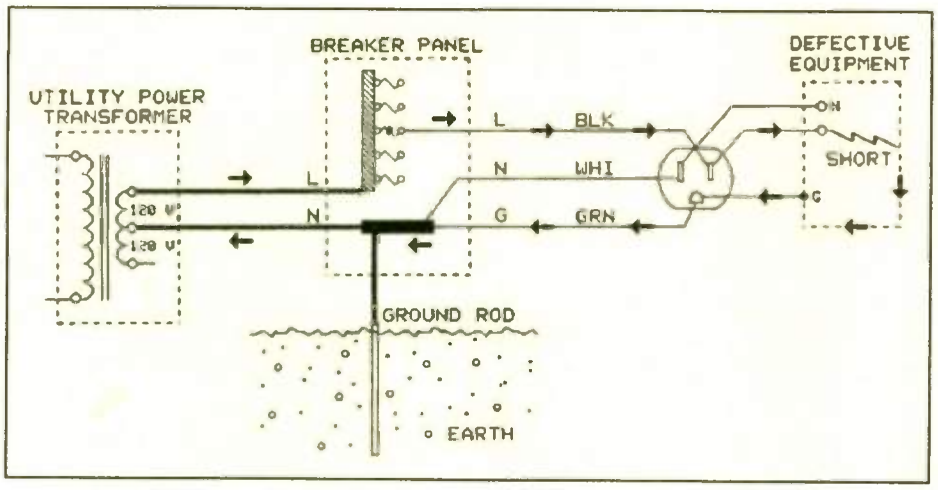

Fig. 4: Current Flow With Faulty Condition

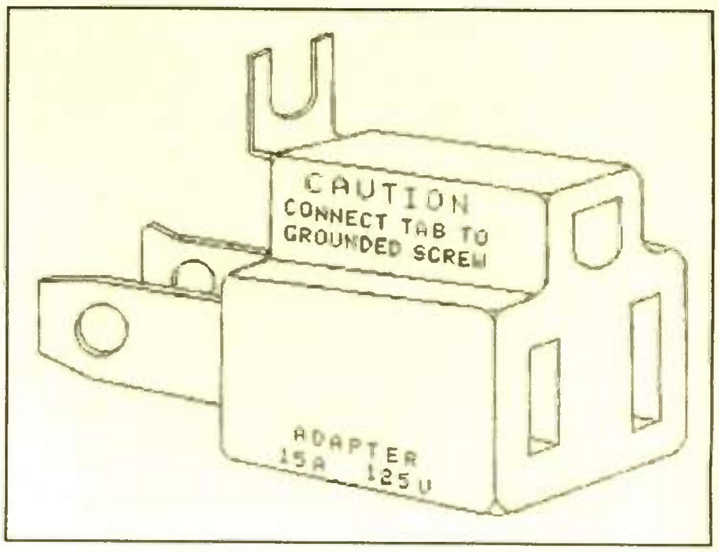

Fig. 4A: 3 to 2 Adapter

Safety grounding for equipment faults

Any AC-line powered device having conductive exposed parts (including signal connectors) can become a shock or electrocution hazard if it develops certain internal defects. Insulation is used in transformers, switches, motors and other internal parts to keep the electricity where it belongs. But the insulation may fail for various reasons, and often in ways that electrically connect the "live" power line to exposed metal — a fault. A washing machine, for example, could electrocute someone who happened to touch the machine and a grounded water faucet at the same time.

To prevent electrocution, most devices have a third wire connecting exposed metal to the "safety ground" terminal of modern AC outlets. This terminal is connected, through either the green wire or metallic conduit, to the main "breaker box" or "electrical panel" where it's tied to neutral. Neutral (white) and line (black) wires are part of the normal load circuit. The green wire or conduit is intended to carry fault currents only. See Fig. 4.

Fig. 4B: Ungrounded equipment can kill.

Since the safety ground is also tied to earth ground, the voltage that might exist between equipment and other earth-

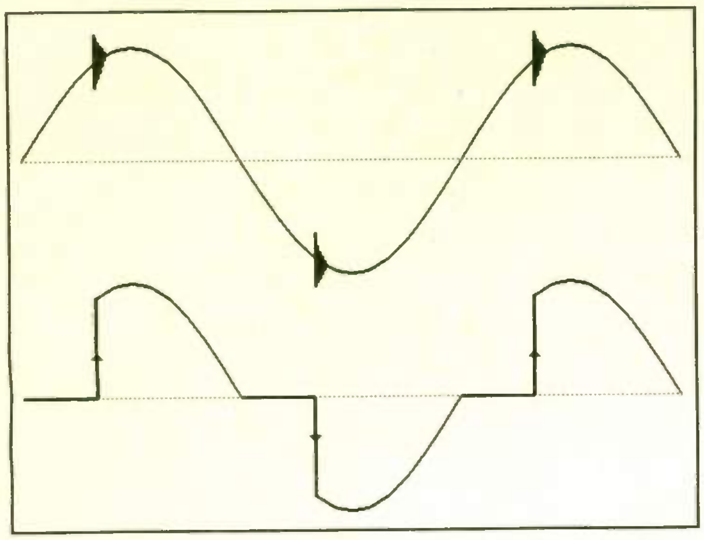

Fig. 5: Lower waveform shows current drawn by dimmed bulbs. Upper waveform shows resulting noise on powerline.

grounded objects (water pipes, etc.) during fault conditions is also minimized. Electrical power engineers refer to these voltages as "step" or "touch" potentials.

With safety grounds in place, potentially deadly equipment faults simply cause high currents from power line "live" to "safety ground," quickly tripping circuit breakers and removing power from those branch circuits. Safety grounding in most residential and commercial buildings is provided through metal conduit, metallic "J-boxes," and saddle-grounded or "SG" outlets.

NEC also requires safety grounding of wiring raceways and equipment cabinets, including rack cabinets. Per Article 250-95, safety grounding wires, which may be bare or insulated, must have a minimum size of #14 copper for a 15 A or #12 copper for a 20 A branch circuit to assure rapid circuit breaker action. This grounding path must be bonded to the safety grounding system, not to building steel or a separate earth ground system. Separate earth ground rods cannot provide safety grounding! Soil resistance is far too high to guarantee tripping of a circuit breaker under fault conditions.

It's always a good idea to verify the integrity of any safety-grounding scheme by measuring its resistance with an approved commercial "ground resistance" tester. General-purpose ohmmeters are not suitable. Outlets mis-wired with safety ground and neutral swapped can cause serious system noise problems. Check the wiring (colors) of suspect outlets or troubleshoot using a clamp-on ammeter.[2,3]

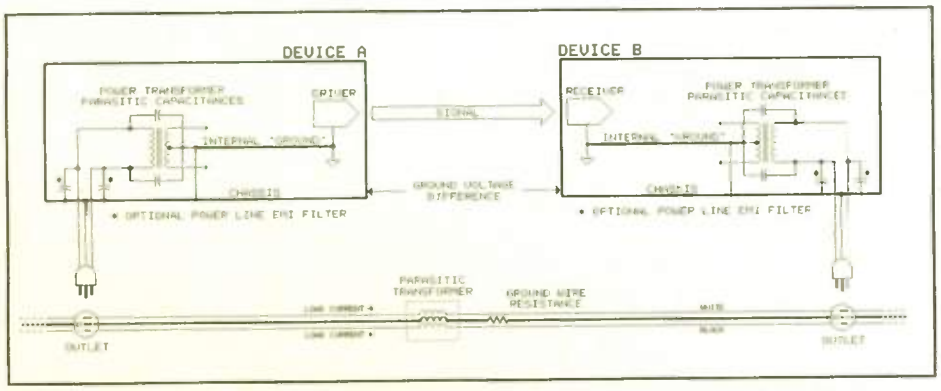

Fig. 6: Voltage difference is created by parasitic transformer in premises wiring.

Important: Never, ever use devices such as three-prong-to-two-prong AC-plug adapters, also known as "ground lifters," to solve a noise problem! See Fig. 4A. These adapters are intended to provide safety grounds (via the screw holding the cover plate to a grounded J-box) in cases where three-prong plugs must be connected to two-prong outlets. Remember that audio, video and other cables that connect equipment can also carry lethal voltages throughout the system if just one "ground lifted" device fails. Consider two pieces of equipment with three-prong AC plugs that are connected by a signal cable. One piece has a ground lifter on its AC plug and the other doesn't. If a fault occurs in the lifted piece of equipment, the fault current (many tens of amperes) will attempt to flow in the signal cable. It's very likely that it will either melt or go up in flames.

Consumer audio and video equipment was responsible for nine electrocutions in the United States in 1997. In the same year, this equipment caused 1,900 residential fires that resulted in 110 civilian injuries, 20 deaths and over $30 million in property losses.[4,5] See Fig. 4B.

Some small appliances, power tools, consumer electronics and outlet-mounted power supplies are equipped with two-prong AC plugs. Sometimes called "double insulated," these devices are specially designed to meet strict UL and other requirements to remain safe even if one of their two insulation systems fails.

Often there is a "one-shot" thermal cutoff switch inside the power transformer or motor windings to prevent overheating and subsequent insulation breakdown. Only equipment originally supplied with a two-prong plug should ever be operated without safety grounding. Equipment originally supplied with a three-prong grounding plug must always be operated with the safety ground properly connected.

GROUNDING PROBLEMS IN THE REAL WORLD

Real-world systems usually consist of at least two pieces of equipment that are connected to, and powered by, the AC line and are also interconnected by signal cables. The power line connections unavoidably cause a significant voltage to exist between the chassis or local ground of any two pieces of equipment, whether they're safety grounded or not. We must accept this fact as reality.

Grounding myths

Myth #1: "Earth grounds are all at zero volts" — presumably with respect to each other and to some "mystical absolute" reference point. This leads to lots of fanciful ideas about ground rods causing system noises to disappear! In fact, the soil resistance between ground rods is much higher (often tens of ohms) than a wire between them.

Myth #2: "Ground wires have zero resistance" — therefore, they can carry a reference (zero volts) ground point to many locations in a system, eliminating voltage differences. In fact, wires are very limited in their ability to "extend" ground points. The DC resistance of a wire applies only at very low frequencies and is directly proportional to its length. For example, the resistance of 8 feet of #10 gauge copper wire is about 0.008 ohms.

Additionally, current in any conductor produces a magnetic field in the surrounding space and an alternating current will produce an alternating magnetic field. This alternating field reacts with the wire in a way that opposes current flow, raising the apparent resistance (or reactance) of the wire in direct proportion to frequency. This property, called inductance, is nearly independent of wire diameter (gauge), but is directly proportional to length and increases significantly at bends or loops. Thus, the same 8-foot wire has an impedance of 22 ohms at 1 MHz (AM broadcast band). Substituting a 1/2-inch-diameter copper rod lowers the impedance only slightly to 18 ohms Further, a wire resonates (becoming an antenna) when its physical length becomes a quarter wavelength, in this case at about 31 MHz.

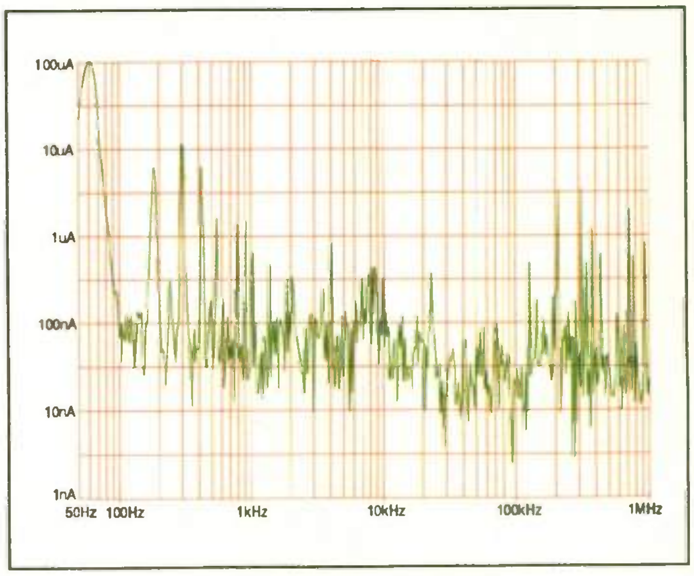

Fig. 7: Typical AC Line Spectrum

Power-line noise spectrum

Power-line voltage normally consists of a broad spectrum of harmonics and noise in addition to a pure 60 Hz sine wave. Power supplies create this noise in electronic equipment, fluorescent lights, light dimmers and intermittent or sparking loads such as switches, relays, or brush type motors (blenders, vacuum cleaners, etc.). Fig. 5 shows how abrupt changes in load current, caused by an ordinary phase-control light dimmer in this case, generate

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Grounding

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high-frequency noise on the power line.

At high frequencies, a building's power wiring behaves like a system of mis-terminated transmission lines gone berserk, reflecting this high-frequency energy back and forth throughout the building's wiring until it is eventually absorbed or radiated.

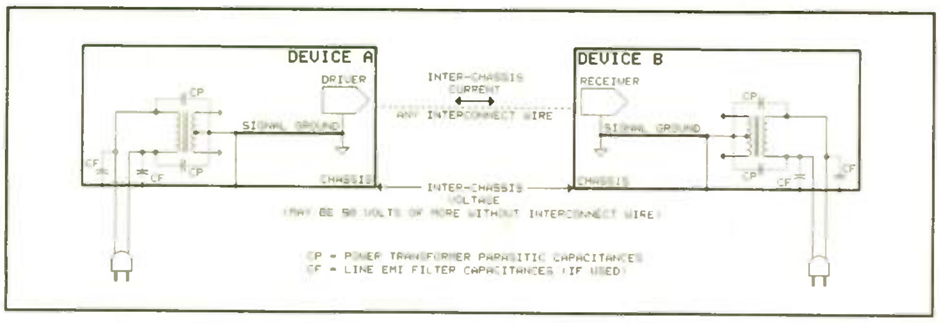

Fig. 8: Leakage current flows between equipment with two-prong plugs.

Capacitive coupling into the ground system

A voltage difference between two conductors produces an electrostatic field in the space between them, and current must flow to charge or discharge the field to a different voltage. In an AC circuit, this current is responsible for an apparent resistance called capacitive reactance, which is inversely proportional to frequency. Therefore, for a given applied voltage, current flow through a capacitor increases in direct proportion to frequency.

In all real equipment, parasitic capacitances — never shown in schematics — exist between the power line and the equipment ground and/or chassis as shown in Fig. 6. They are the unavoidable interwinding capacitances of its power transformer. If the equipment contains power-line RFI/EMI filters, they further add to this capacitance. These capacitances cause relatively small "leakage" current flow between power line and ground in every AC-operated device.

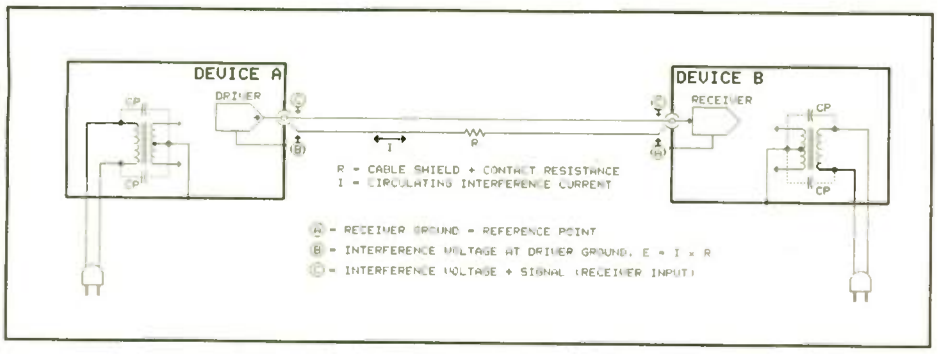

Fig. 9: Common Impedance Coupling

> Never, ever use devices such as three-prong-to-two-prong adapters to solve a noise problem.

Ungrounded equipment

Many electronic devices, generally low power, use a two-wire power cord. If they're not grounded some other way, such as via rack-mounting or connection to grounded equipment via signal interconnections, the chassis of these devices can assume an open-circuit voltage up to 120 volts AC with respect to safety ground. However, the available current is very low. For UL-approved consumer equipment, leakage current is limited to 0.75 mA (0.5 mA for office equipment), which will cause a noticeable, but harmless, tingling sensation as it flows through a person. Fig. 7 shows current vs. frequency for a typical 3 nF capacitance and 120-volt AC from a typical outlet.

Therefore, any wire connecting an ungrounded device to safety ground, or two such devices to each other as shown in Fig. 8, will form a "capacitive ground loop" for the leakage current. We must accept the existence of these leakage currents as a fact of life.

Grounded equipment

Safety-grounded devices use three-wire power cords. The leakage currents in these devices are allowed to be much higher (up to 5 mA) because they normally flow back to the neutral connection through the building safety ground wiring. The cumulative leakage current of many devices can create small but significant voltage drops between outlets on a branch circuit, as shown in Fig. 6 on page 19.

In addition, the line, neutral and ground conductors of branch circuit wiring form an unintentional transformer. Under normal conditions, instantaneous load currents in the line (black) and neutral (white) conductors are equal and opposite. Therefore, the magnetic fields they generate are also equal and opposite. If the grounding (green) conductor were perfectly equidistant from them, there would be no voltage induced in it. In reality, physical asymmetry of this wiring causes significant voltage to be induced over the length of the grounding conductor.

Thus, inter-outlet ground voltage differences generally increase with physical distance and can reach 1 volt or more. Higher voltages are often seen between grounds on two different branch circuits if they are fed from opposite "phases" of the incoming utility power. Therefore, any wire connecting two grounded devices to each other will create a "hard-wired ground loop." Loop current can easily reach 100 mA or more.

GROUNDING EFFECTS ON AUDIO SYSTEMS

Not only are real-world systems usually connected to AC power, they are also interconnected by signal cables. If leakage currents or inter-outlet ground voltages are allowed to couple into audio signal circuits, the result is hum, buzz, pops, clicks or other symptoms. Exactly how the coupling occurs can depend on whether interfaces are balanced or unbalanced, and whether devices are grounded or ungrounded.

Coupling in unbalanced interfaces

Consumer and so-called "semi-pro" audio electronics most often use unbalanced signal interfaces. These interfaces are extremely susceptible to noise coupling, making it very difficult to achieve professional levels of dynamic range in all but the simplest systems.

By far, the most serious problem in unbalanced interfaces is illustrated in Fig. 9. The grounded conductor is an essential part of the signal circuit, but it also is a path for power-line leakage currents that flow between the devices. Because the interface is a series circuit, voltage generated over the length of the grounded conductor is directly added to the signal seen by the receiver. Since the impedance of the grounded conductor (usually but not always the cable shield) is common to both the audio and leakage current circuits, this mechanism is called common-impedance coupling.[6] Since the output impedance of device A and the input impedance of device B are in series with the cable's signal conductor, its impedance has an insignificant effect and is not represented here.

Consider a 25-foot interconnect cable with foil shield and a #26 AWG drain wire. From standard wire tables or actual measurement its shield resistance is found to be 1.0 ohms. If the interchassis leakage current is 300 µA, the noise voltage will be 300 µV. Since the -10 dBV reference level for consumer audio is about 300 mV, the noise will be about 20 log (300 µV / 300 mV) or about 60 dB below a reference signal. For most systems, this is indeed a poor signal-to-noise ratio.

Common-impedance coupling can become very severe between two grounded devices, since the ground voltage difference effectively is forced across the ends of the grounded signal conductor. In some situations, noise voltage may actually be larger than the reference signal.

Less common is noise coupling into cables from nearby electric or magnetic fields. Strong AC electric fields radiate from any conductor operating at a high AC voltage (a neon sign, for example). AC power cords are a more common but much less potent source. An AM radio broadcast antenna or other nearby RFI source, including arcing contacts, can also generate strong electrostatic fields. A conductive cable shield that completely surrounds the inner signal conductor (100 percent "coverage") prevents capacitive coupling to the signal circuit. Foil shields usually have 100-percent coverage. Braided shields, because they have small openings, generally vary from 85 to 95 percent, which is entirely adequate in most cases.

Strong AC magnetic fields radiate from any conductor operating at a high AC current such as building wiring, power transformers, electric motors and CRT displays. Ordinary cable shielding, whether copper braid or aluminum foil, has no significant effect on audio frequency magnetic fields. In general, both electric and magnetic field strengths decrease rapidly with distance from the source, making physical separation an effective treatment. Unbalanced interfaces, unlike balanced ones, cannot nullify noise induced by electric or magnetic fields.

SUMMARY

Hum and buzz are often serious system problems that limit dynamic range. Although it may be the source of the noise, the power line and its wiring is rarely the problem. All properly wired, code-compliant AC wiring normally produces small voltage differences within the grounding system and all equipment normally produces small leakage currents. Earth ground rods do not drain away these artifacts and they cannot be eliminated with massive copper conductors. But signal interfaces, especially the unbalanced kind, can allow these tiny voltages and currents to couple into signal paths.

In part two, we'll talk about balanced interfaces, how to make quick work of troubleshooting, how to fix the problem interfaces without compromising safety or audio quality and why some so-called "cures" are a waste of money. ■

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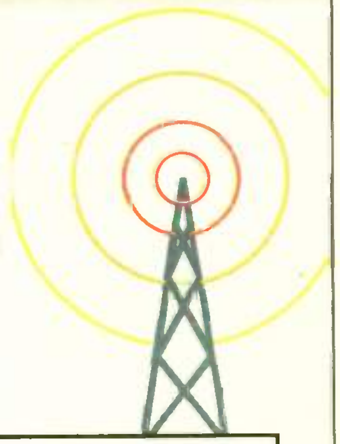

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The Psychology of Technical Quality

A Digital Innovator Argues That Managers Should Frame Their Questions More Carefully

By Barry Blesser

Modern technology, be it hardware, software or any complex system, is often riddled with bugs and defects. Why is it so difficult to install, design and manufacture equipment without such problems?

The answer is simply that, when creating technology, human beings have intrinsic limitations. While the principles of engineering often are articulated as a rational set of rules, professional engineers, managers and operators are still people. People have inadequate memory, incomplete rules of inference, emotional baggage and private agendas. Evolution optimized our species for survival as a social animal operating in a hostile world, not as a logical creature operating in a rational world.

Understanding human psychology as applied to engineering can greatly improve quality and reliability. This is a fundamental premise of industrial psychology, which I constantly use at 25-Seven. While my experience has been that few people in the radio and audio industries have been exposed to such academic disciplines, some of the basic ideas are so simple and self-evident that they can be put into practice without special knowledge or training.

Managers should not confuse speculating about failure with negativism or professional incompetence.

Barry Blesser

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.

FRAME THE QUESTION

I begin with two premises: (a) everyone takes pride in and benefits from a well-engineered result, and (b) achieving such a result follows from asking the right questions during the engineering process.

Herein lies the challenge. How do we spot flawed questions that undermine our ability to achieve desired results?

I discovered the principle of carefully framing the question while directing large engineering projects for such companies as AKG, EMT, Orban, Studer and Lexicon; but the ideas have broad applicability.

Consider: it is easy to ask obvious questions of a development team: "Have you checked the design carefully? Have you tested the system? Have you followed accepted methods for writing software? Are we ready to ship?"

Everyone on the team will answer "yes" because they want to demonstrate that they are dedicated, serious professionals who have made every effort to produce a good product. The manager can ask focused questions about particular issues, and he will get logical, coherent answers that are believable, and usually true. At a psychological level, the team wants to please the manager and be rewarded for providing answers consistent with a positive outcome.

But in this example, our questions failed to probe the unknown risks where bugs and defects remain hidden and ready to bite.

REINFORCE, DON'T UNDERMINE

Now consider an alternative line of questions with a different set of assumptions.

The manager asks the group to speculate on "hypothetical" points of failure, regardless of how likely or unlikely they are to occur: "If the design were to fail, imagine ways in which that might happen."

Someone might suggest that dirty contacts could produce a sequence of very rapid closures, or that somebody might lean on the front panel and simultaneously press an illegal and unexpected key combination, or that a configuration file might have a character that was not visible, and so on.

The team still wants to please the manager; but the manager has refocused the goal on creating failure scenarios, rather than on listing what already works.

Once a list of hypothetical failure mechanisms has been compiled, the manager and the team can consider the likelihood and the effort to test for such mechanisms. In my experience, from a list of 100 such mechanisms, 60 are not worth considering and 20 have already been handled carefully; but the final 20 definitely need to be explored. When those cases are analyzed carefully, the quality of the product is very much improved.

Managers should not confuse speculating about failure with negativism or professional incompetence. One cannot simultaneously ask that failure scenarios be explored, and then castigate a colleague for discovering a flaw through this process. "Why didn't you think of that before" reactions are guaranteed to reinforce "tell me what I want to hear" responses, thereby undermining the entire process.

The fact that a certain flaw may not be discovered until a proper stimulus is provided is a reflection of human limitation. An effective manager provides such stimulus by framing the questions properly, and rewarding responses that expose flaws and risks.

Consider an application of this method in a radio station where the question is to articulate all the mechanisms that might result in being knocked off the air.

While everyone has considered backup power generation, nobody might have considered testing the fuel tank, which gets modestly drained on each monthly test. Somebody might come up with the observation that a critical telephone hybrid was not part of the auxiliary power system. Somebody else might observe that a cable with a vacuum leak would still work under a short test, but would fail after 20 minutes of use. Scenarios can be very complex and not the least obvious until creative people focus on inventing them.

REWARDS

I have used this approach with great success in a variety of situations, often unrelated to technology. A marketing plan, a proposed new studio or a change in the structure of an organization can all benefit from this approach.

Frame the right questions, and the probability of success will increase dramatically. In simple terms, allow the staff to be rewarded for exploring human limitations and frailties without fear of being demoralized or degraded. The idea is common sense, but it is also counter-intuitive because so many people mistakenly equate a focus on what could go wrong with "glass half empty" negativism.

In reality, those managers who are eager to discover risks are, in fact, the greatest optimists because they really want to achieve perfection. ■

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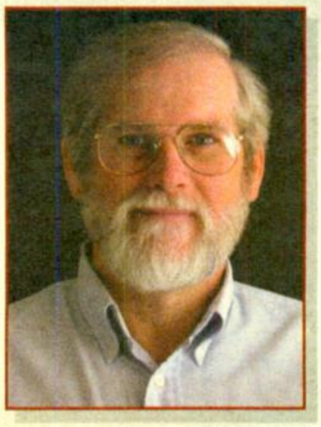
DESIGNER INTERVIEW



Andrew Calvanese

Building Today's Console Systems

Wheatstone's Andrew Calvanese Talks About Consoles and the Latest in Manufacturing Technology

By Michael LeClair

Audio consoles remain at the heart of any radio facility. Few design choices in a new studio will have as great an impact on overall performance as the choice of console and its design.

In this installment of our Designer Interview series, we took an inside look at a company that designs audio consoles for the broadcast and audio industries.

CALVANESE, PAGE 15

The Math of Split-Level Combining

By George Cabrera

Harris Corp.,
Broadcast Communications Division

High-level combining is a practical way to upgrade existing FM transmitters to IBOC. Very well understood, with readily available 10 dB couplers, it is a very attractive method with which to upgrade even the highest power levels in the field. Unfortunately, it offers only a combining efficiency of 83 percent, as 90 percent of the IBOC power is wasted into a dummy load, along with 10 percent of the generated FM power. Additionally, the power headroom demanded from the FM transmitter is proving to be an unsurpassable requirement for some existing sites ready to upgrade.

This paper presents an innovative combining technique that minimizes the FM and IBOC power loss in high-level combining. Instead of combining two totally incoherent signals with its known resulting losses, this method first achieves partial coherence of the signals, and only then combines them at an optimum-coupling ratio. A system implemented this way offers two main benefits: a higher overall system efficiency, and the elimination of the need for FM headroom. Though the paper focuses on the particular case of FM HD Radio upgrade, this technique can also be used whenever low loss is needed when combining dissimilar signals.

INTRODUCTION

High-level combining has been widely used since the beginning of IBOC conversions. It requires a new digital transmitter, around one-third the size of the main transmitter, whose IBOC output is high level combined with the FM signal, using a 10 dB coupler. The power lost in the reject load and the power headroom demanded off the main transmitter rate are the two main complaints against this method.

Split-Level Combining (SLC), an improved IBOC high-level combining scheme, is presented in Fig. 1.

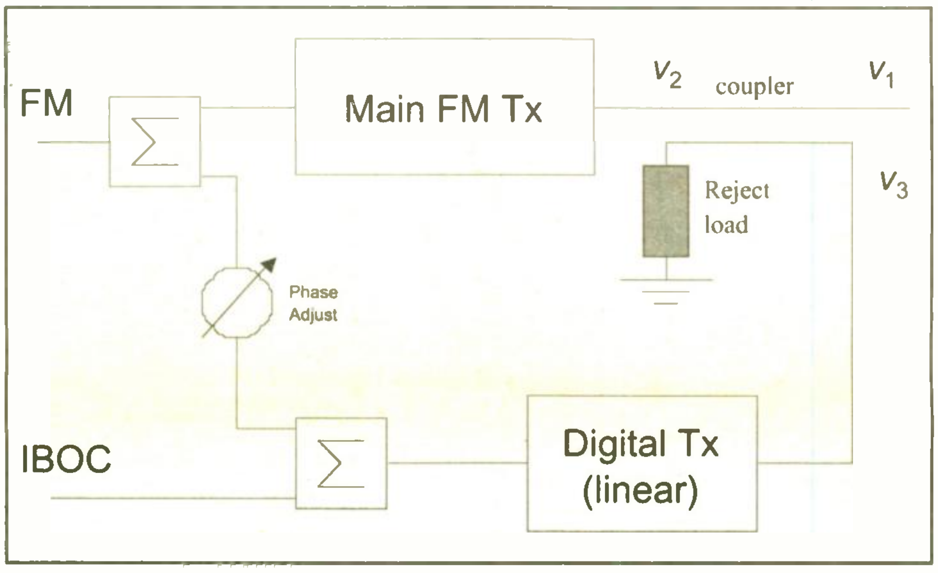


Fig. 1: Split-Level Combining

The digital transmitter carries not only the IBOC signal, but also a small portion of the FM signal that has been phased to add at the output combiner with the signal generated by the main FM transmitter.

The new Split-Level scheme is one solution to the problem of upgrading to IBOC without impacting the output requirement from the main FM transmitter. It also reduces the power lost in the reject load by at least 50 percent. These features, along with the increase in the overall system efficiency, will be studied in this paper.

COMBINING ANALYSIS

When two signals are combined, the instantaneous output voltage $v_1(t)$ is given by the sum of each coupled input voltage:

$$v_1(t) = S_{12} \cdot v_2(t) + S_{13} \cdot v_3(t)$$

The average output power, P_1, is obtained by averaging, over time, the square of the resultant output voltage:

$$P_1 = \frac{1}{T}\int_0^T \frac{(S_{12} \cdot v_2(t) + S_{13} \cdot v_3(t))^2}{Z_o} \cdot dt$$

In the case of coherent signals, the numerator can be simplified as the square of the sum of their *rms voltages*:

SPLIT-LEVEL, PAGE 8

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From the Tech Editor

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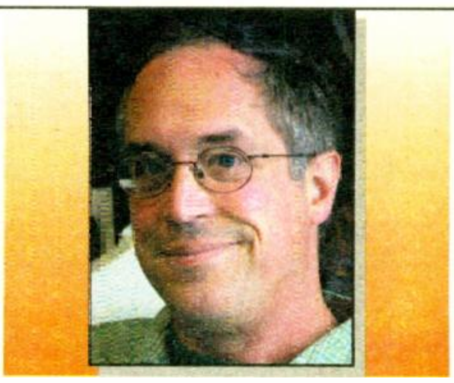
To Further the Discussion

By Michael LeClair



Michael LeClair

As a working engineer I like to take every opportunity I can to meet with my fellow engineers and trade stories. In our market we have an active SBE chapter with regular monthly meetings and an additional Radio Engineers Luncheon that meets quarterly. And occasionally, a few of us will run into one another at a shared tower site and grab a few minutes to compare notes.

These meetings, whether by chance or schedule, are far from being wasted time. I learn from other engineers what does and doesn't work. From time to time I contribute something from my own hard-won experience to help someone else solve a problem.

In recent years, this sharing of knowledge and stories has exploded on the Internet with the wide use of e-mail listservers. Wise equipment manufacturers have joined in the online discussion to provide comments that reflect the designer's perspective and with offers to help where needed. Keeping up on these discussions has become a normal part of the working day for most engineers. As fewer engineers are now maintaining a larger number of stations, it is the rare individual who has the personal experience and education to resolve the wide range of technical problems that may arise, let alone plan for future technologies.

I like to think that *Radio World Engineering Extra* can also play a similar role in helping to disseminate technical knowledge. In many years of attending trade shows and conventions I have seen a wealth of deep technical papers concerning our industry Unfortunately, the majority of working engineers don't have the opportunity to attend these shows on a regular basis and they miss out on an important source of information. This is why we will be publishing a regular series of "white papers" that delve into the latest broadcast engineering topics in a detailed way. It is my goal to pick out the best of these papers and bring them to you.

In recent gatherings of engineers, a topic that always comes up is the current development and deployment of HD Radio, or the IBOC digital system. In this month's issue we have a paper from George Cabrera of Harris Corp. that explains the theory behind a new system for IBOC transmission, which is known as Split-Level Combining. This new technique offers another method of generating the digital IBOC signal that I expect will be appealing to a number of engineers. As a theory paper it includes a lot of mathematics, but I think you will find the paper to be quite elegant in its reasoning and analysis.

Also in this issue we have an extended discussion with Andrew Calvanese of Wheatstone Corp as a second installment in our Designer Interview series. The function and design of broadcast consoles have undergone nothing less than a revolution in the last eight years and this candid interview talks about that change and where we are headed.

But that's not all. We've got a fine paper on the measurement of AM noise in broadcast transmission systems, a second installment from Bill Whitlock on audio grounding and a piece from Stephen Poole of Crawford Broadcasting about hurricane survival.

This paper is for you, the working engineer, and we want to hear from you about your interests and concerns. I would love to hear feedback about what you read here and what you might like to see in the future Please feel free to e-mail your comments and suggestions to *mlrwee@verizon.net*. ■

Correction

The Oct. 27 issue of *Radio World Engineering Extra* included an image on page 8 accompanying the article "Answers to Your Questions About FM Antennas" by Richard J. Fry.

Only part of the image was printed. Here is the correct version of the figure.

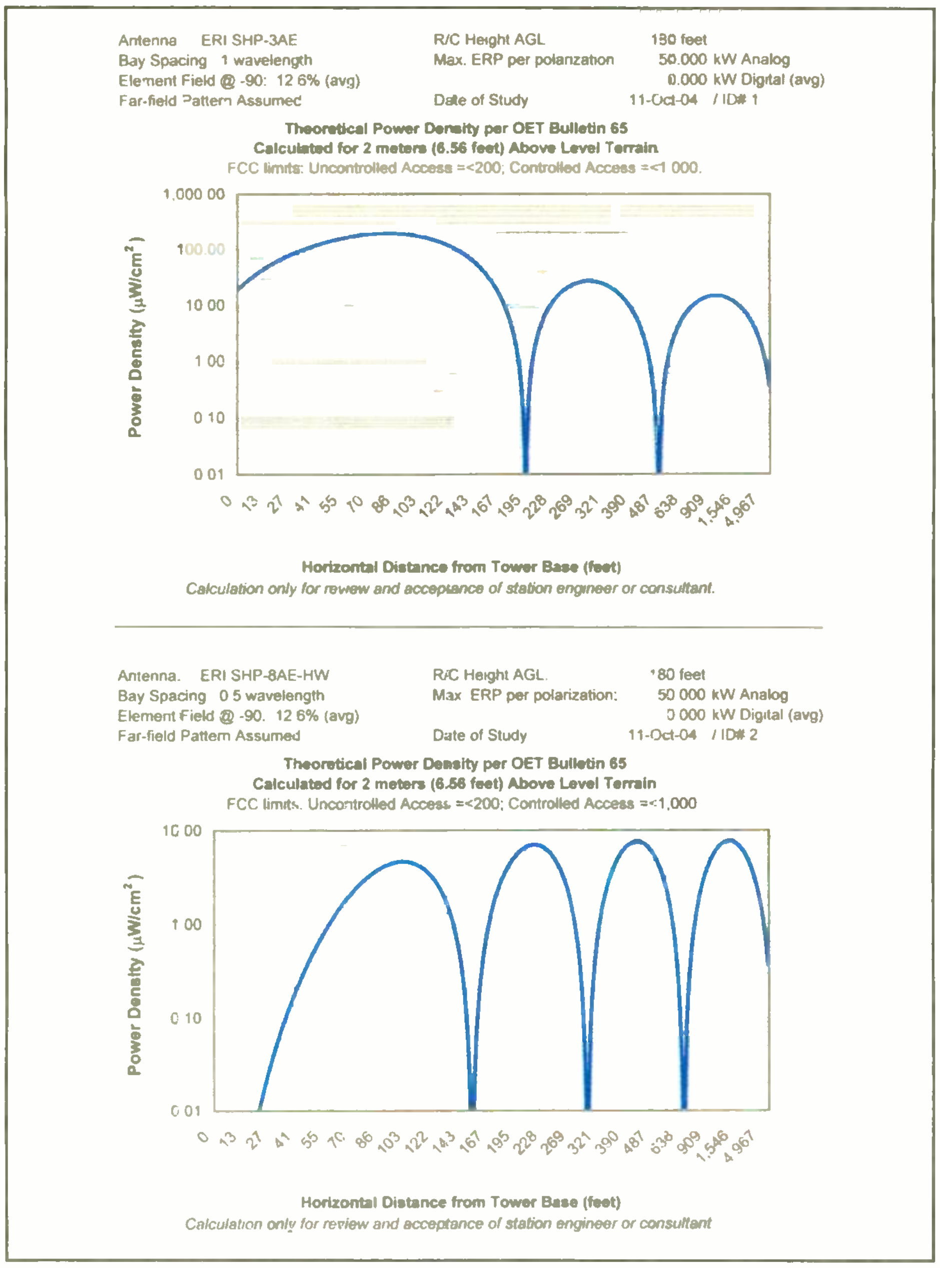


Fig. 2: Comparison of Power Densities for Two Antenna Configurations

Understand Ground Loop Problems: Part II

By Bill Whitlock

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

In part one of this article, in the Oct. 27 *Radio World Engineering Extra*, 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 small voltage differences and/or allows small power-line currents to flow between pieces of equipment in a system. Problems arise when these voltages or currents couple into audio signal paths, and we showed that unbalanced audio interfaces are particularly vulnerable to such coupling.

In this, part two of the article, we will explore balanced interfaces.

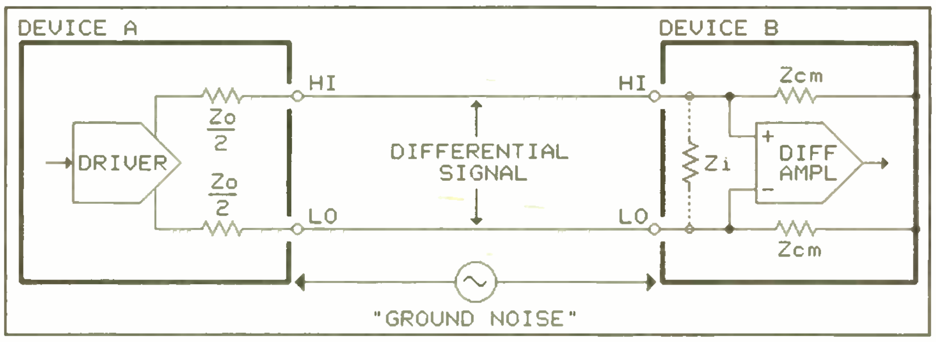
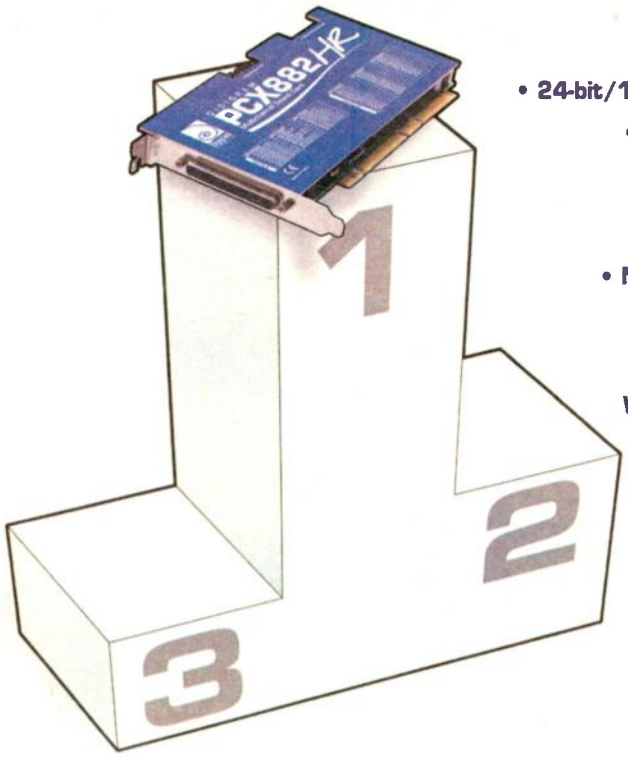


Fig. 1: Common-Mode Impedances in Balanced Interface



COUPLING IN BALANCED INTERFACES

Professional audio systems have traditionally used only equipment having balanced interfaces. In my humble opinion, only equipment with balanced inputs and outputs should ever be called professional.

The use of balanced interfaces is a potent technique to prevent noise coupling into signal circuits. It is so powerful that many systems, such as telephone systems, use it instead of shielding as the main noise reduction technique.

Balance vs. Symmetry

The true nature of balanced interfaces is widely misunderstood.

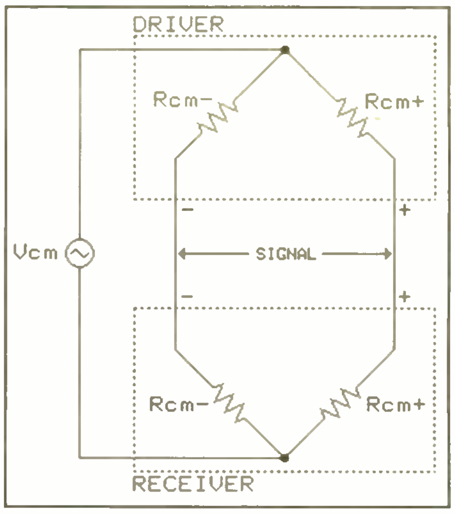


Fig. 2: Bridge imbalances convert common-mode noise to signal.

For example, one common definition is: "*Each conductor is always equal in voltage but opposite in polarity to the other. The circuit that receives this signal in the mixer is called a differential amplifier and this opposing polarity of the conductors is essential for its operation.*"[1] This, like many such explanations in print, describes signal symmetry (equal in voltage, but opposite in polarity), but misses the single most important property of a balanced interface.

A concise and accurate definition is: "*A balanced circuit is a two-conductor circuit in which both conductors and all circuits connected to them have the same impedance with respect to ground and to all other conductors. The purpose of balancing is to make the noise pickup equal in both conductors, in which case it will be a common-mode signal which can be made to cancel out in the load.*"[2]

This concept is further refined in the IEC Standard 60268-3: "*Therefore, only the common-mode impedance balance of the driver, line and receiver play a role in noise or interference rejection. This noise or interference rejection property is independent of the presence of a desired differential signal. Therefore, it can make no difference whether the desired signal exists entirely on one line, as a greater voltage on one line than the other, or as equal voltages on both of them. Symmetry of the desired signal has advantages, but they concern headroom and crosstalk, not noise or interference rejection.*"[3]

Common-Mode Rejection

A simplified balanced interface is shown in Fig. 1. Theoretically, a balanced interface can reject any interference due to ground voltage differences, magnetic fields or electric fields, as long as it produces identical voltages on each of the signal lines and the resulting peak voltages don't exceed the capabilities of the receiver. Any voltage that appears on both inputs, because it is common to both inputs, is called a common-mode voltage.

A balanced receiver uses a differential device, either a specialized amplifier or a transformer, which inherently responds only to the voltage difference between its inputs. An ideal receiver would have no response to common-mode voltages. In reality, the response is not zero and the ratio of differential gain to common-mode gain of this device is its common-mode rejection ratio, or CMRR. It's usually expressed in decibels, where higher numbers mean better rejection.

Note that the common-mode (with respect to ground) output impedances of the driver and input impedances of the receiver effectively form a Wheatstone bridge as shown in Fig. 2. If the bridge is not balanced or nulled, a portion of the ground voltage difference, V_{cm}, will be converted to a differential signal on the line. The nulling of the common-mode voltage is critically dependent on the ratio matching of these pairs of driver/receiver common-mode impedances. The nulling is relatively unaffected by impedance across the lines — only the common-mode impedances matter!

This bridge is most sensitive to small fractional impedance changes in one of its arms when all arms have the same impedance.[4] It is least sensitive when upper and lower arms have widely differing impedances. Therefore, we can minimize the CMRR degradation in a balanced interface caused by normal component tolerances by making common-mode impedances very low at one end of the line and very high at the other.

The output impedances of virtually all line drivers are determined by series resistors (and often, coupling capacitors, too) that typically have ±5 percent tolerances. Because of this, typical drivers can have output impedance imbalances in the vicinity of 10 ohms. The common-mode input impedances of typical balanced input circuits is in the 10 kohms to 50 kohms range, making the CMRR of the interface exquisitely sensitive to normal imbalances in driver output impedance.

For example, the CMRR of the widely

GROUNDING, PAGE 19

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AM Noise: The QC Standard for FM Broadcast

By Joel Bump

The author is president and director of engineering at Radio Design Labs and a former radio-engineering consultant in southern California. He is responsible for the design of the RDL ACM-1, ACM-2 and ACM-3 AM noise monitors.

It has been 16 years since I first published a series of detailed technical articles in Radio World on the subject of AM noise in FM transmission systems.

Over the years, a number of engineers I knew in the southern California area have retired or moved on to other markets and positions. More solid-state transmitters are in service, and AM noise has been routinely monitored and controlled in hundreds of stations improving service areas and reception quality. Yet demands on FM transmission system bandwidth persist; and questions often are asked about proper coupling methods for AM noise monitoring in both new installations and existing transmitter plant upgrades.

AM NOISE DEFINED

AM noise is the unwanted amplitude modulation of an FM carrier. There are two types of AM noise: *synchronous* and *asynchronous*.

Asynchronous noise consists of amplitude modulation unrelated to the FM modulation of the carrier, typically caused by power supply hum or vibration. Unless there is a serious problem with the transmitter, asynchronous AM is far less significant than synchronous AM. Unwanted AM modulation produced by modulation from baseband audio and all subcarriers is synchronous AM, sometimes referred to as incidental AM.

Consistent control of synchronous AM noise can result in improved audio clarity, better stereo separation, lower crosstalk into subcarriers and extended service area.

EFFECTS OF UNCONTROLLED AM NOISE

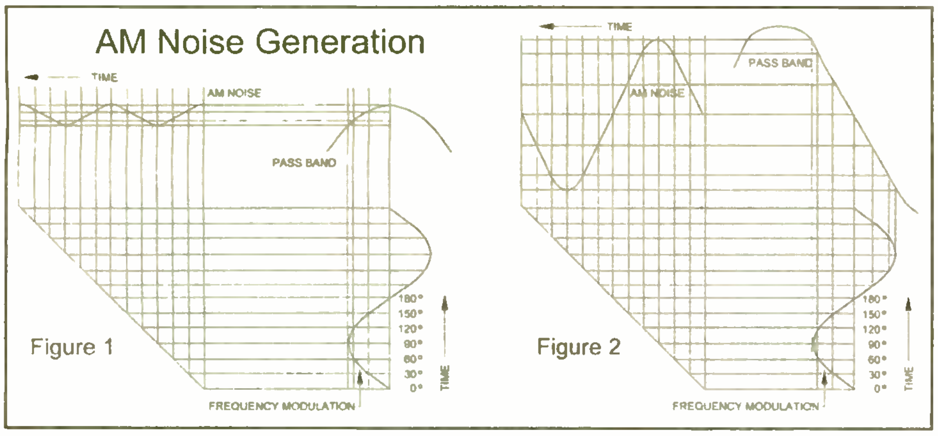
FM receivers ultimately are required to produce analog output signals. This is accomplished by converting the frequency swing of the incoming carrier to amplitude values that feed the receiver output or are further decoded to produce stereo audio outputs. Receiver designs seek to minimize the effect of carrier amplitude variations during this process, but received RF levels in weak signal areas or in mobile receivers vary drastically over short distances. When the received signal instantaneously drops below a certain threshold in the radio, the received amplitude variations are directly detected and combined with the baseband audio.

The result is audible noise in the stereo audio, reduced separation, poor subcarrier performance and a reduction in the station's effective "solid" service area.

SOURCES OF AMPLITUDE VARIATIONS AT THE RECEIVER

Amplitude variations that can be detected by the receiver may result from multipath distortion of the carrier. At the relatively high frequencies in the FM band, transmitted signals reflect off many surfaces, from hills to buildings to power lines. When a direct and a reflected signal reach the receiving antenna at the same time, they will add or subtract, resulting in significant variations in level. I have measured multipath variations as great as 30 dB over distances of 50 feet.

Amplitude variations from AM noise are also inherent in the actual transmitted carrier. AM noise produces an effect similar to multipath at the receiver in weaker signal areas. Large and often important portions of a station's coverage area may be located in these regions. AM noise in the transmitted signal tends to multiply the effects of multipath significantly. A moderate level of AM noise together with moderate, otherwise unobjectionable, multipath can produce highly objectionable noise in many receivers. This effective reduction in coverage area can be controlled by ensuring that low AM noise is transmitted at all times.



Figs. 1 and 2: These graphs show the AM noise waveform and relative amplitude produced as a result of modulation. The frequency modulation waveform drives the carrier above and below the center frequency. The passband of the transmission system at each frequency produces variations in the carrier amplitude producing AM noise. Time is indicated in 30-degree increments relative to the modulation frequency. The vertical lines drawn from the modulation waveform represent instantaneous carrier frequencies above and below the station's assigned carrier frequency. The intersection of the instantaneous frequency with the passband slope is carried to the left side of the graph, where the resulting AM noise is plotted against the same time increments.

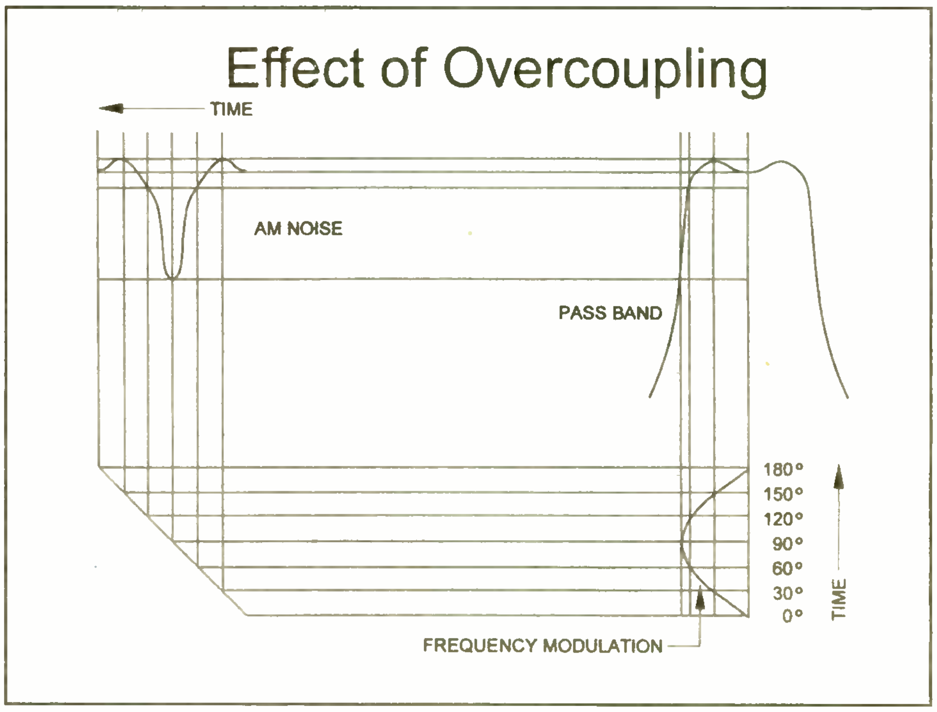


Fig. 3: Effect of Overcoupling

HOW SYNCHRONOUS AM NOISE IS PRODUCED

Synchronous AM noise results from tuned circuits. Coupling between RF amplification stages in a transmitter, tuned output circuits, low-pass filters, antenna tuning and even transmission line bullets contribute to AM noise. Ideally, all power is transferred equally across the frequency deviations of the transmission system. In practice, however, sideband attenuation is never perfectly equal and many system elements vary with time and temperature.

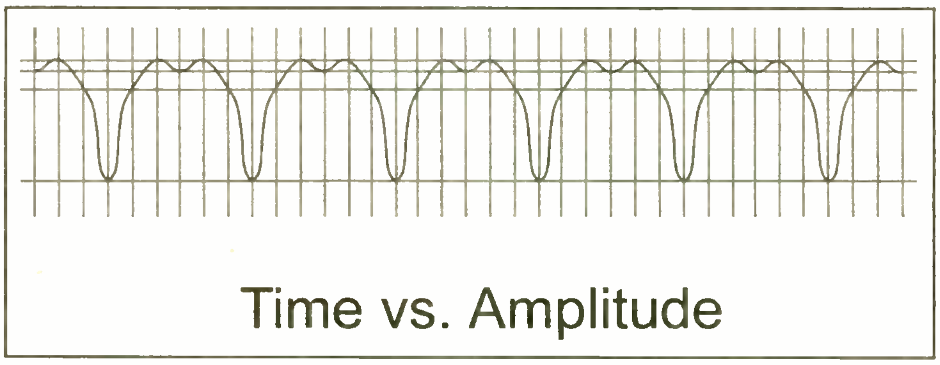


Fig. 4: AM Noise With Overcoupled Passband

The fundamental production of synchronous AM noise is shown in Fig 1. As the FM carrier frequency shifts with modulation, shown as a sine wave, the passband slopes produce a direct variation in the carrier amplitude. These variations are defined as AM noise. Because the passband is symmetrical, amplitude variations result from both the higher and lower frequency slopes. The resulting AM is twice the frequency of the FM modulation.

If the center of the passband is shifted below the center frequency, the resulting AM will be the same frequency as the FM modulation. As seen in Fig. 2, the amplitude of the AM noise also increases as the tuning shifts off center.

These diagrams visually indicate how AM noise is produced. Multiple amplification and tuning stages are designed to maintain the flattest passband and widest bandwidth. Interstage coupling, however, results in actual passbands that are less uniform than in these basic examples. Coupling may be increased to produce flatter response over a broader range of frequencies while simultaneously producing steeper skirts.

The passband in Fig. 3 results from some interstage overcoupling and is more nearly representative of a multistage transmission system. This example shows only one-half cycle of modulation applied to the carrier, yet the resulting AM noise is now four times the modulation frequency. The waveform of the AM noise does not exhibit uniform positive and negative amplitudes, but it does produce four AM cycles for each FM cycle applied to the carrier. This graph clearly shows the effect of the skirts on the amplitude of the AM noise. If the passband in Fig. 3 were slightly wider, the same frequency deviation would produce significantly lower AM peak amplitude.

Fig. 4 shows the AM noise produced by

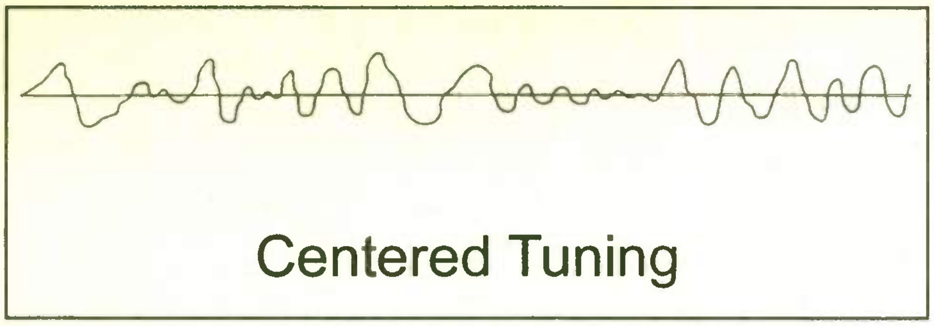


Fig. 5: AM Noise With Centered Passband

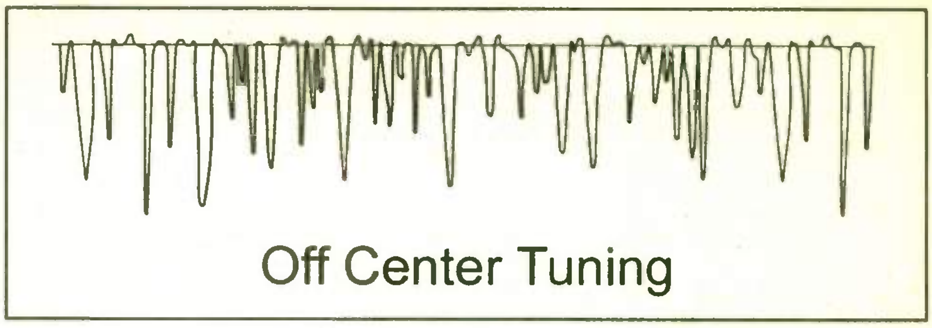


Fig. 6: AM Noise With Offset Passband

the passband from Fig. 3. This waveform exhibits a high-peak excursion relative to the corresponding RMS voltage. It is particularly important to note the ratio of RMS to peak energy in the AM component. It is the peak amplitude modulation that is subject to detection in the receiver. If the slope of the passband skirts in Figs. 3 and 4 were tightened, the peak AM excursion would become narrower while still producing the same objectionable amplitude at the receiver.

The sine wave modulation used in these examples clarifies the generation of AM noise and its relationship to the system passband. The nature of program audio and subcarrier modulation consists of narrower waveforms containing even less RMS energy while still producing the same peak amplitudes.

WHERE AM NOISE SHOULD BE MEASURED

The best measurement would include the effects of every bandwidth-limiting factor in the transmission system. Ideally, AM noise should be measured at the output of the transmitting antenna without the effects of any external reflections. Since that is not practical, the optimum sample is immediately prior to the transmission line feeding the antenna.

It is imperative that the sample is taken from a directional sample of the forward carrier wave. The sample should be as close to the antenna in the signal path as possible. It should be after the harmonic filter and after any other notch filters or coaxial switches.

The location of the monitoring sample is very important and can be useful in verifying the mechanical integrity of the plumbing prior to the sample. In addition to verifying transmitter performance, I have found AM noise readings to be instrumental in identifying burned bullets in rigid line or RF switches before they became off-air critical.

Many older transmitters and some line sections provide monitor outputs that are "capacitively coupled" to the carrier. Such samples must be avoided for AM noise measurements because they contain harmonic and reflected components that produce erroneous AM. When unwanted signals combine with the forward signal, spurious AM is produced that would cause the engineer to mistune the transmitter, degrading rather than improving performance.

Certain sampling slugs that fit line section ports are capacitive as evidenced by a coupling adjustment screw adjacent to the output jack. These samplers must not be used for AM noise.

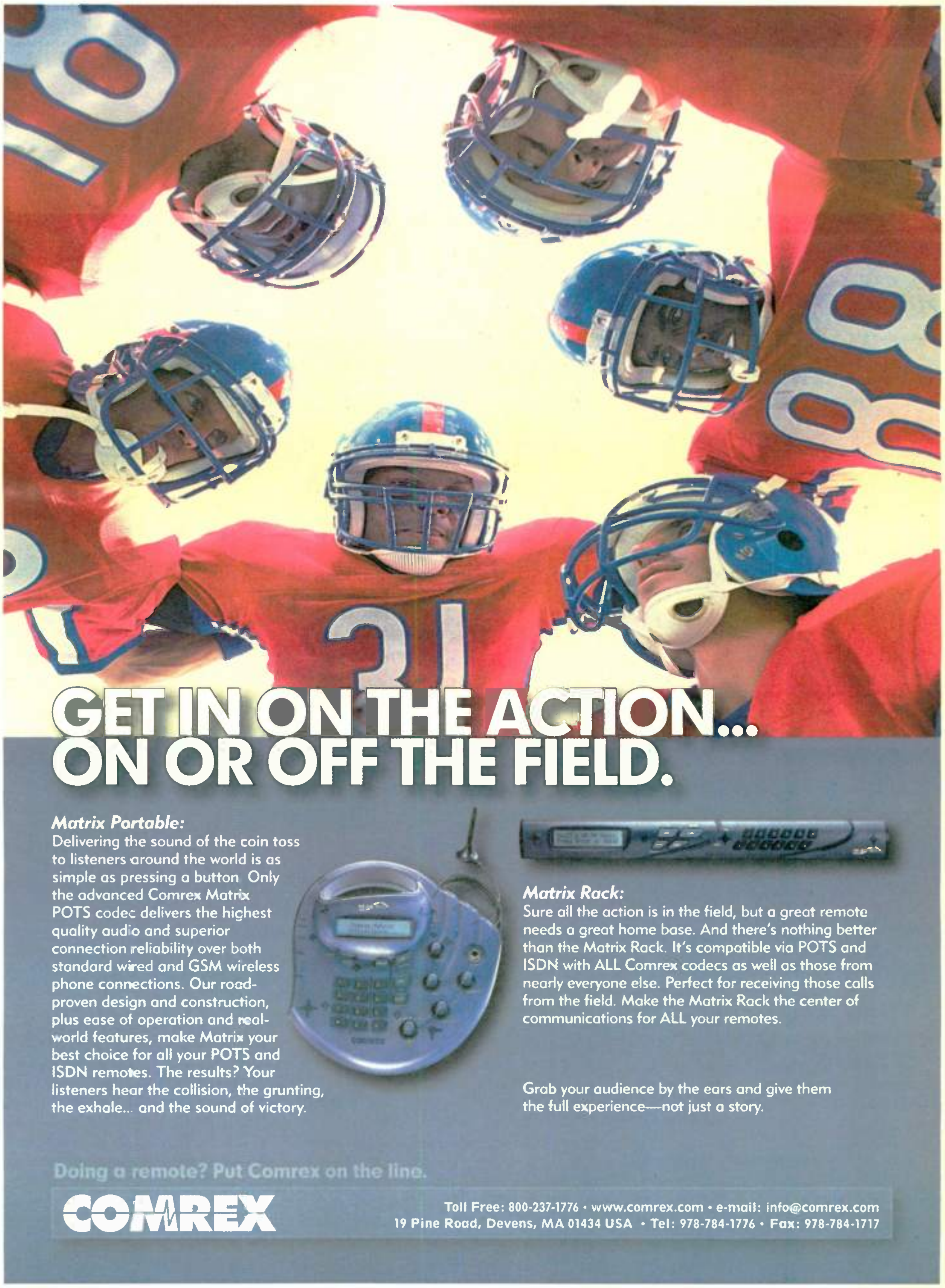
SAMPLING RF CARRIER FOR AM NOISE MEASUREMENT

A transmission line section with an available sampling port is normally installed just prior to the transmission line. A directional sampling slug is used for proper AM noise measurement. The slug should be oriented toward the forward carrier wave and the AM measurement detector must be connected directly to the output of the slug.

Two characteristics of the slug are most important. First, it must have sufficient RF output level to produce linear detection over the wide dynamic range to be measured. Second, it must have sufficient 50-ohm internal load dissipation to deliver its output continuously into a 50-ohm detector load. Some common sampling slugs do not meet these requirements. Available samplers meeting the requirements are listed at *rdl-net.com/pdf/Data_Sheets/acm-3.pdf.*

The AM noise levels being measured in a properly operating facility can be a low as 70 dB below the carrier level. Clearly, if the sample is to be accurate, it cannot contain any spurious material. For that reason, the detector must be directly connected to the sample. If the detector is connected to the sampler using a coaxial cable, even minor reflections in that cable will produce serious errors in the detected AM. The output from an AM detector contains DC plus detected

AM NOISE, PAGE 20



Split-Level

CONTINUED FROM PAGE 1

$$P_1 = \frac{{v_1}^2}{Z_o} = \frac{(S_{12} \cdot v_2 + S_{13} \cdot v_3)^2}{Z_o}$$

where v_1, v_2 and v_3 are the corresponding rms values (vectors) for the waveforms $v_1(t)$, $v_2(t)$ and $v_3(t)$.

However, for incoherent signals, such as those at different frequencies (FM and IBOC), the total power is obtained by the sum of each coupled *power*. This can be proven by solving the square of the resultant instantaneous voltage.

$${S_{12}}^2 \cdot v_2(t)^2 + 2 \cdot S_{12} \cdot S_{13} \cdot \overbrace{v_2(t) \cdot v_3(t)}^{average\,=\,0} + {S_{13}}^2 \cdot v_3(t)^2$$

Due to orthogonality between $v_2(t)$ and $v_3(t)$, their multiplication averages over time to zero, amounting to no contribution to the average power at the output. The resulting term, to be averaged over time, has been reduced to:

$${S_{12}}^2 \cdot v_2(t)^2 + {S_{13}}^2 \cdot v_3(t)^2$$

This means that the average power at the output, for noncoherent signals, is obtained by summing each power coupled from the inputs.

$$P_1 = {S_{12}}^2 \cdot P_2 + {S_{13}}^2 \cdot P_3$$

It is important to note that this value is smaller than the value obtained by the square of the sum of their rms voltages (properly phased), and accordingly, some power is being lost.

In the following sections we will see the advantage of adding some coherence between the signals v_2 and v_3, such that the middle term does not totally cancel.

HOW MUCH ADDITIONAL FM INJECTION

We will achieve partial coherence by adding some FM signal at port 3, along with IBOC. We will now determine the minimum amount of additional FM injection needed so that the main FM power can remain unchanged.

Let's assume that v_1 is the output voltage, v_2 is the FM voltage applied at the through port (where the main FM transmitter connects), and $v_{3\text{-}fm}$ is the FM voltage applied at the coupled port of the output combiner (where the digital transmitter connects).

To further simplify the analysis, let's make $Z_o = 1$, and the total coupled FM power into port 1 (output) = 1. Then at 100 percent Transmitter Output Power (TPO), P_1=1 and v_1=1.

$$P_1 = (S_{12} \cdot v_2 + S_{13} \cdot v_{3\_fm})^2 = 1$$

Solving for $v_{3\text{-}fm}$, the FM voltage needed at the coupled port 3 to achieve output power P_1=1:

$$v_{3\_fm} = \frac{1 - S_{12} \cdot v_2}{S_{13}}$$

Let's use this expression for the condition of no impact on the main FM power. This condition forces v_2=1, and then the

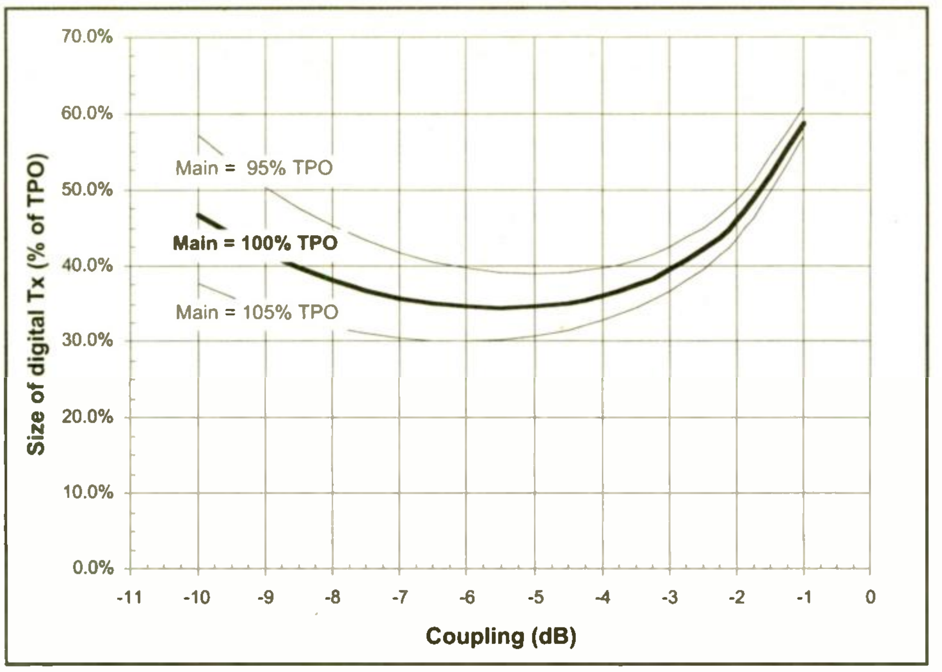


Fig. 2: Relative Size of Digital Transmitter That Guarantees the Main FM Transmitter to Run at 95-, 100- or 105-percent TPO

value $v_{3\text{-}fm}$ obtained is the FM voltage needed at port 3 that guarantees the main FM transmitter power is not affected. In other words, viewed from the main transmitter's port (port 2), the combiner has no loss.

If we now sweep a range of coupling factors, we obtain the additional FM injection that is needed, as a function of the coupler used, in order to make the main FM path lossless.

Table 1: Additional FM injection needed not to impact the main FM transmitter output.

Coupling (dB)	***Additional FM injection (percent of TPO)***
-10	2.6 percent
-9	3.4 percent
-8	4.3 percent
-7	5.6 percent
-6	7.2 percent
-5	9.5 percent
-4	12.6 percent
-3	17.2 percent

HOW MUCH IBOC INJECTION

The peak IBOC voltage needed at the coupled port 3 (v_{3_iboc}) to achieve the proper injection at the output is:

$$v_{3\_iboc\_peak} = \frac{0.1 \cdot K_v}{S_{13}}$$

where K_v is the peak to average for the IBOC signal, in linear terms, needed to comply with the RF mask. A value of 1.778 (5 dB) will be considered throughout the analysis. The ratio between the IBOC and FM through the digital transmitter is:

$$IBOC : FM = 20 \cdot \log\left(\frac{0.1}{1 - S_{12} \cdot v_2}\right)\ dBc$$

Using the case in which the main transmitter remains at 100 percent TPO (v_2=1), and considering a 6 dB coupler, the IBOC/FM ratio needed is –2.5 dBc.

In Table 1 we saw that the required amount of secondary FM injection at port 3 increases with the coupling factor, but in general it is a small fraction of the main service or TPO.

On the other hand, the required generation of IBOC at port 3 is inversely proportional with the coupling coefficient. The smaller the coupling coefficient (the looser the coupling), the more IBOC power will need to be generated.

Considering both effects at once, tighter coupling coefficients require more additional FM injection but less IBOC generation. Let's find the optimum coupling value that minimizes the size of the digital transmitter.

OPTIMUM COUPLING COEFFICIENT FOR MINIMUM-SIZE DIGITAL TRANSMITTER

The total voltage at port 3 is:

$$v_{3\_total} = v_{3\_fm} + v_{3\_iboc\_peak}$$

This generates a peak power of:

$$P_{3\_total\_peak} = \left(\frac{1 - S_{12} \cdot v_2}{S_{13}} + \frac{0.1 \cdot K_v}{S_{13}}\right)^2$$

This value includes fast, highly compressed envelope peaks. It is the short duration of these peaks, much shorter than the amplifier thermal constant (its temperature will hardly change with modulation) that allows the device to reach higher levels than in a CW mode. Accordingly, as we want to compare the size of a digital transmitter, whose envelope is being pulsed, to the corresponding 100 percent TPO (that is provided by a CW-operated FM transmitter), we will have to adjust these peak levels to its equivalent maximum CW operation.

Based on peak power measurements, we rate the maximum CW operation of a digital transmitter 0.5 dB below its maximum pulsed level. As an example: an 11.22 kW peak-capable digital transmitter will be considered to be a 10 kW CW unit (10 kW = 11.22 kW-0.5dB). This is the value that will be used to compare sizes between the FM and Digital transmitters.

The relative size of the digital transmitter, with respect to 100 percent TPO, can then be estimated by:

$$Relative\_size = \left(\frac{1 - S_{12} \cdot v_2}{S_{13}} + \frac{0.1 \cdot K_v}{S_{13}}\right)^2 \cdot 10^{\frac{-0.5\,dB}{10}}$$

If we sweep a range of coupling coefficients (S13), and graph the relative size for three different values of v_2 (main FM output), we obtain the curves shown in Fig. 2.

Several important conclusions can be extracted from the graph above. The optimum-coupling coefficient (for size) is in the region of 5 to 6 dB, instead of the 10 dB coupling used in separate amplification.

The digital transmitter needs to be 35 percent the size of the main FM transmitter (or TPO), in order to provide the secondary FM injection level that is required to make the main FM-path lossless. This value (35 percent) is comparable to the size demanded for separate amplification (33 percent). In other words, with about the same hardware we are now providing a "lossless" solution for the FM path.

The graph also shows that a bigger digital transmitter can afford to over-inject FM, resulting in gain at the main FM port. On the other side, a smaller transmitter, with less injection than the one needed not to impact the FM path, will require the main FM transmitter to overcome some loss.

Finally, by using this tighter coupling coefficient (6 dB), the amount of IBOC generated is reduced by 4 dB, of which 25 percent will get coupled to the output. As a result, the IBOC power dissipated into the reject load is about 30 percent of the amount dissipated in separate amplification. The next section studies the reject power in more detail.

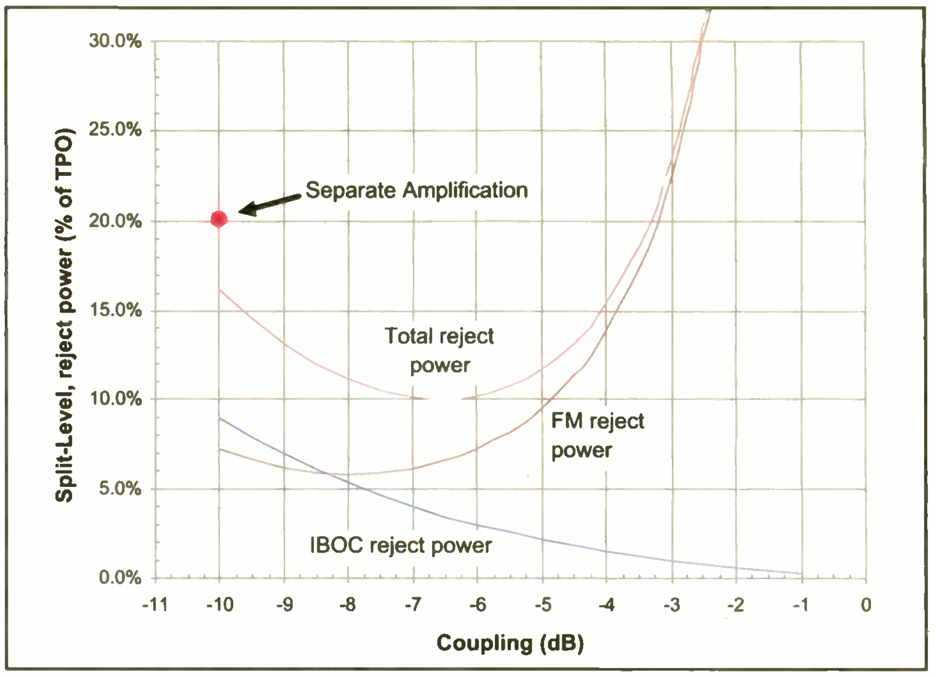


Fig. 3: Reject Power vs. Coupling Coefficient

OPTIMUM COUPLING COEFFICIENT FOR MINIMUM REJECT POWER

The total reject power is composed of:

$$P_{reject} = P_{reject\_fm} + P_{reject\_iboc}$$

where,

SPLIT-LEVEL, PAGE 10

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- USB interface fits any ML1 or DL1
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- Enables data storage in analyzer for later upload to PC
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- Control the analyzer from the PC
- Firmware updates via PC
- MiniLINK USB interface is standard

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World Radio History

Split-Level

CONTINUED FROM PAGE 8

$$P_{reject\_fm} = (S_{13} \cdot v_2 - S_{12} \cdot v_{3\_fm})^2$$

and,

$$P_{reject\_iboc} = \left(\frac{0.1 \cdot S_{12}}{S_{13}} \right)^2$$

To keep our analysis practical, let's select the same size transmitter as the one required for separate amplification (minimum size).

With this restriction the total FM voltage supplied at port 3 will be limited to:

$$v_{3\_fm} = 0.623 - \frac{0.1778}{S_{13}}$$

As the coupling coefficient is changed (or swept), and the digital transmitter size is not allowed to change, we will have to adjust the output of the main FM transmitter (v_2) in order to hold constant 100 percent TPO at the output of the combiner ($v_1 = 1$).

The value of the voltage at port 2 then needs to be:

$$v_2 = \left(\frac{1 - S_{13} \cdot V_{3\_fm}}{S_{12}} \right)$$

The graph in Fig. 3 shows the total reject power for the Split-Level system, and its FM and IBOC contributions. As a comparison, the red dot represents the reject power for a 10 dB coupler separate amplification system.

The minimum in total reject power is found at 6.5 dB coupling, with a value equal to half of the power dissipated in separate amplification.

DIGITAL TRANSMITTER OPERATING IN SPLIT-LEVEL MODE

Calculating Maximum Power in SLC Mode

Considering signal clipping to be our limit (independent of its probability of occurrence), we can calculate the maximum average power at which a digital transmitter can be operated, for different IBOC injection levels.

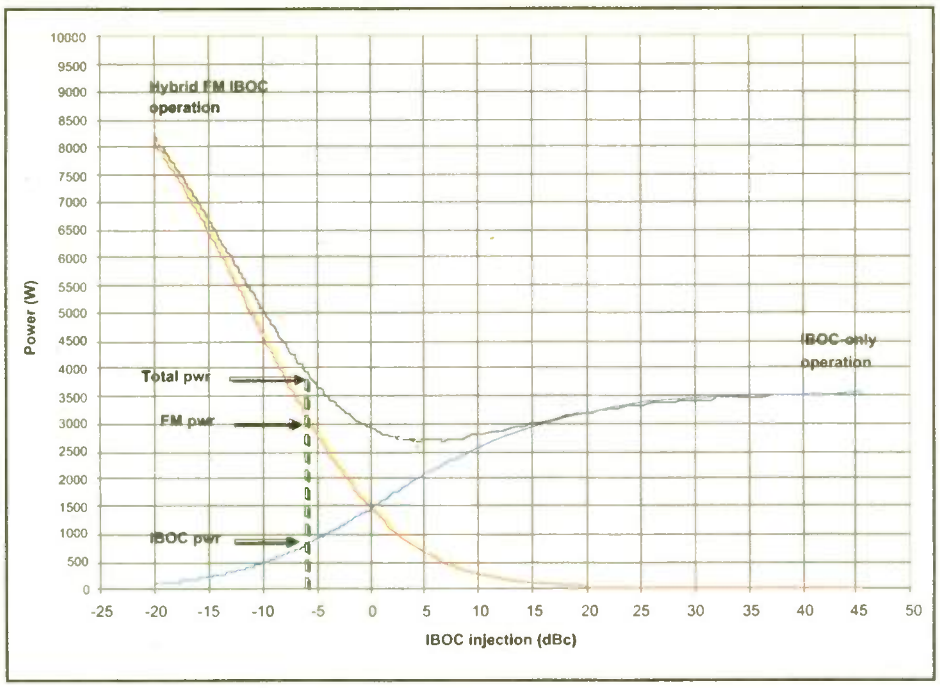


Fig. 4: Digital Tx, Maximum Average Power for Different IBOC/FM Ratios

$$P_3 = P_{3\_fm} + P_{3\_iboc}$$

Recalling that its pulsed saturation rating is 0.5 dB above its CW rating, we can then substitute and obtain:

$$P_3 = {v_{3\_fm}}^2 + \left(\frac{1.06 - v_{3\_fm}}{K_v} \right)^2$$

The total output power of the digital transmitter, relative to its CW equivalent rating, has been plotted in Fig. 4, as a function of the IBOC/FM ratio.

For injection levels below –5 dBc, the peak-to-average ratio (PAR) will be even greater than those found in IBOC-only operation. This is a worst-case prediction, as it considers all IBOC amplitude peaks to always be phase-aligned with the CW FM carrier. To get a more accurate prediction, the probability of this event (phase alignment at peaks) has to be taken into consideration.

Envelope Probability Distribution

The noise-like envelope in IBOC-only operation could be closely approximated by a Rayleigh distribution, given by:

$$E_{IBOC}(v) = v \cdot \frac{e^{\frac{-v^2}{2 \cdot \sigma^2}}}{\sigma^2}$$

where v is the envelope level, s is its standard deviation (σ = √[illegible]), and $E_{IBOC}(v)$ is the probability density function (PDF) of the envelope.

When used in Split-Level mode, the additional FM signal (CW) modifies this distribution. The new PDF is better described using a Rice distribution,

$$E_{SLC}(v) = v \cdot \frac{e^{\frac{-(v^2 + vfm^2)}{2 \cdot \sigma^2}}}{\sigma^2} \cdot I0\left(\frac{v \cdot vfm}{\sigma^2} \right)$$

where v_{fm} is the FM voltage injected, and s is again the standard deviation of the IBOC signal, and *I0* is a modified Bessel function.

Let's compute the average power that is above the minimum saturation point of 5 dB. This power can be seen as the maximum power that can be wasted in IMD regrowth, and still be within the RF mask. When scaled to the total, unclipped average power it becomes:

$$IMD\max_{IBOC} = \frac{\int_{vsat}^{\infty} (v^2 - vsat^2) \cdot E_{IBOC}(v).dv}{\int_0^{\infty} v^2 \cdot E_{IBOC}(v).dv}$$

here vsat is the saturation voltage at 5 dB above rms voltage, and $IMD\max_{IBOC}$ is the integrated IMD power, scaled to the total IBOC average power.

When the FM carrier is added, then we use the Rice distribution function $E_{slc}(v)$ and the IMD is calculated in the same manner, as a ratio of the total integrated power:

$$IMD\max_{SLC} = \frac{\int_{vsat}^{\infty} (v^2 - vsat^2) \cdot E_{SLC}(v).dv}{\int_0^{\infty} v^2 \cdot E_{SLC}(v).dv}$$

We find the maximum SLC operating power at any given level of IBOC injection, by increasing the FM and IBOC content (v_{fm} and s) until we reach the maximum level of IMD. This maximum allowable level of IMD is only a fraction of that previously calculated for the IBOC-only case.

In other words, in order to keep the same ratio of desired IBOC signal to IMD level, the maximum IMD for Split-Level mode has to be reduced as we reduce the IBOC injection level:

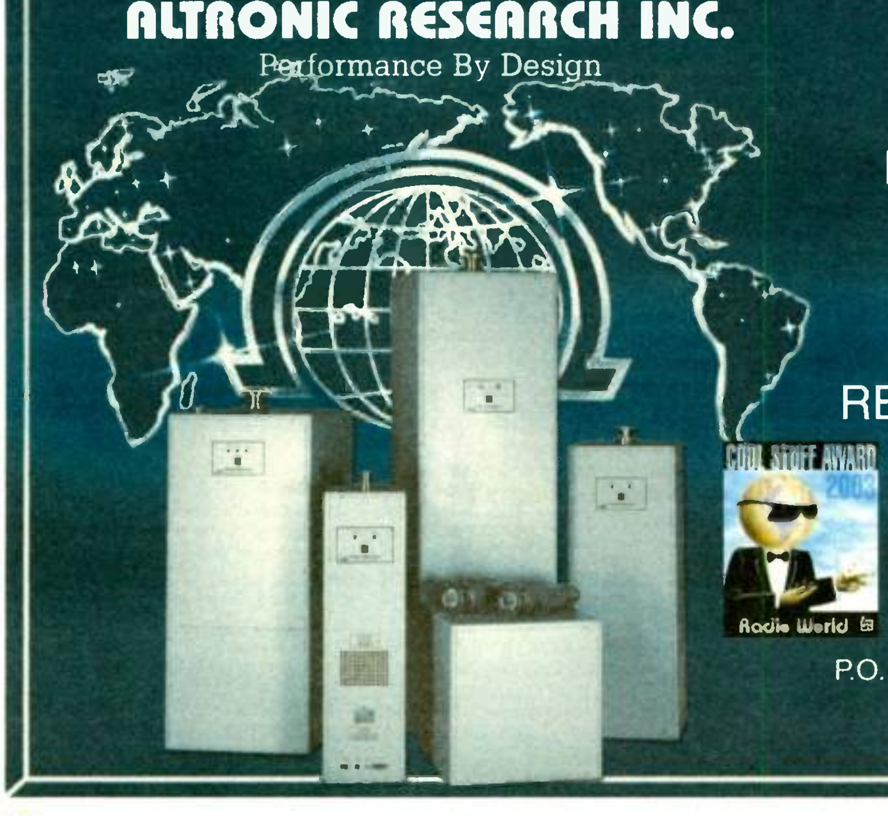
$$IMD\max_{SLC} = \frac{10^{\frac{IBOCinj}{10}}}{1 + 10^{\frac{IBOCinj}{10}}} \cdot IDM\max_{IBOC}$$

This criterion will guarantee the same ratio of desired signal (IBOC average power) to undesired signal (IMD average power), independent of injection level. It does not, however, take into account changes in the IMD energy distribution across the spectrum. For our range of interest, the curve represented in Fig. 5 tracked satisfactorily well experimental data.

The maximum average power, when the envelope probability distribution is considered, is always above the curve obtained by the simpler vector voltage summation.

This extra capability has proven to be more useful in the region of -3 dBc to -5 dBc IBOC injection, which is the region operated by smaller transmitters that are to be combined with 6 dB couplers. In this

SPLIT-LEVEL, PAGE 14

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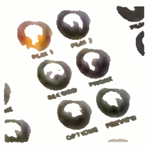

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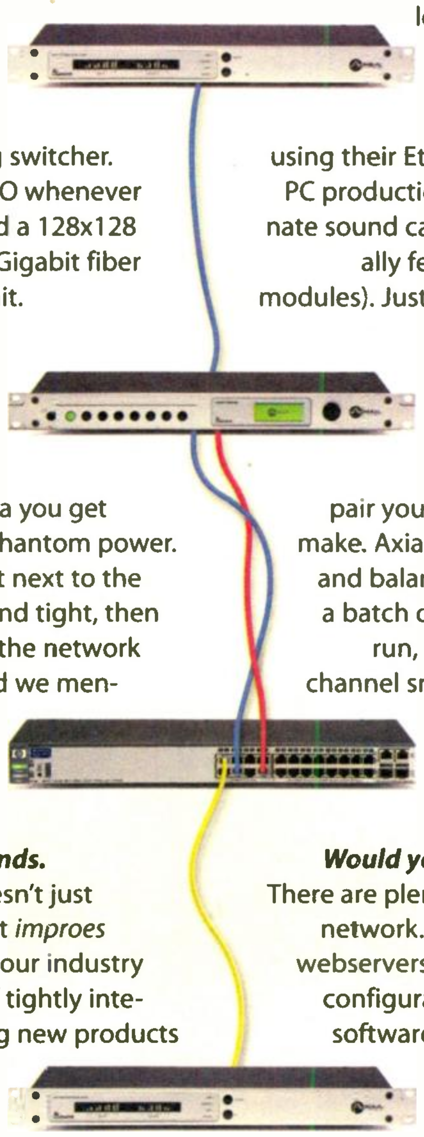

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Split-Level

CONTINUED FROM PAGE 10

region, an additional 0.7 dB can easily be extracted above the limit otherwise imposed by voltage summation.

Referring again to Fig. 5, the maximum operating average power, with an IBOC injection of -2.5 dBc, is about the same of the IBOC-only rating or Separate amplification. We can already predict an improvement in the overall system efficiency, i.e., with the same generated average power from the digital transmitter (with respect to separate amplification), we have upgraded our system to FM IBOC without demanding 11 percent more power from the main FM transmitter.

EFFICIENCY OF A DIGITAL TRANSMITTER OPERATING IN SPLIT-LEVEL MODE

The measured efficiency of a digital transmitter, operating at different levels of IBOC injection is represented in Fig. 6.

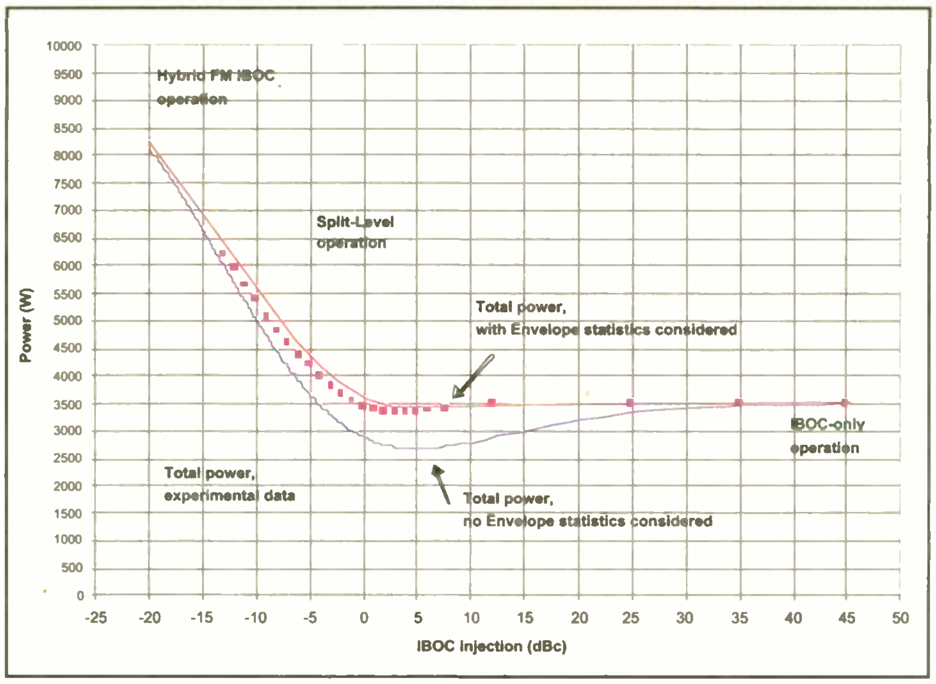

Fig. 5: Maximum Output Power, When Envelope Statistics are Considered

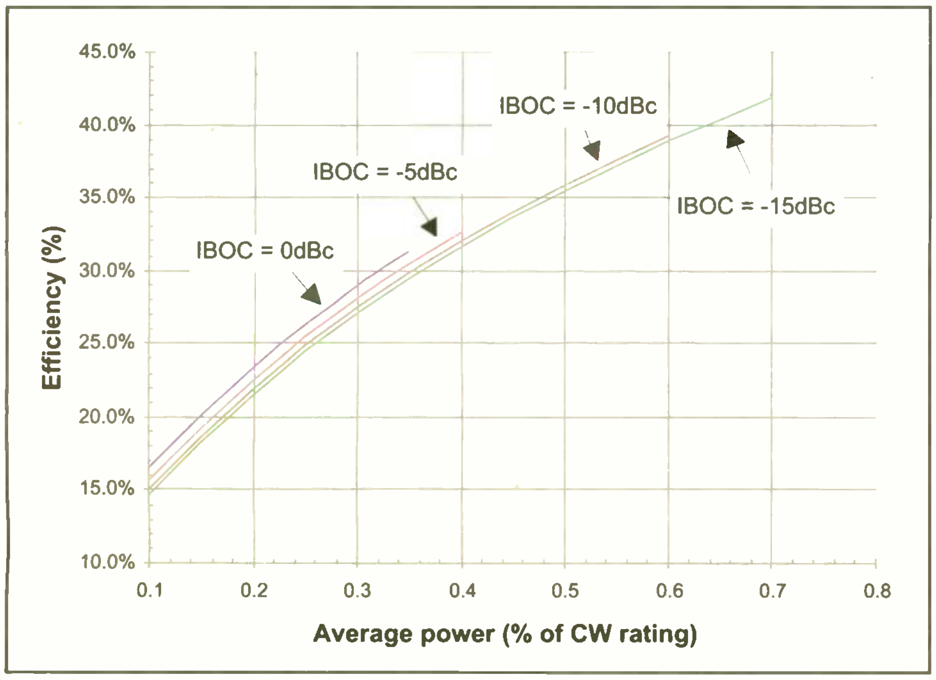

Fig. 6: Digital Tx Efficiency, for Different IBOC/FM Ratios

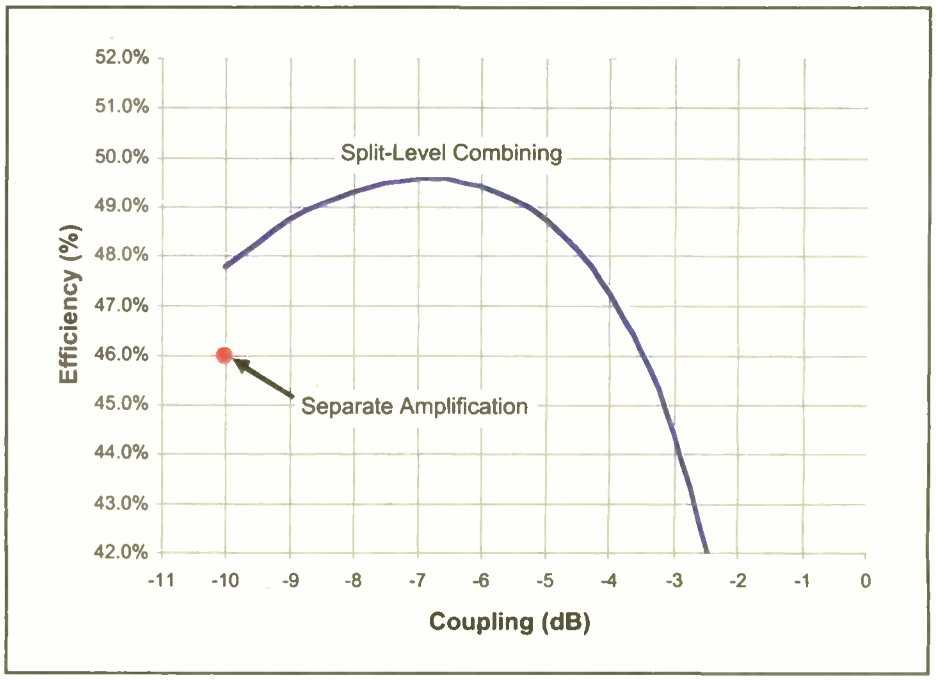

Fig. 7: System Efficiency for Different Coupling Coefficients

For IBOC/FM ratios less than -10 dBc, the efficiency of a digital transmitter is very similar to the efficiency of common amplification. For higher injection levels, the efficiency quickly improves (for the same average power) and approaches the efficiency of IBOC-only operation.

OVERALL SYSTEM EFFICIENCY

Knowing that our output power (including IBOC) has to be 1.01, the overall system efficiency is:

$$\eta_{sys} = \frac{1.01}{1.01 + P_2 \cdot \left(\frac{1-\eta_2}{\eta_2}\right) + P_3 \cdot \left(\frac{1-\eta_3}{\eta_3}\right) + P_{reject}}$$

We will assume a 60-percent efficiency for the FM transmitter (η_2). The efficiency curvefit function used to generate the graph in Fig. 6 will be used to compute the value of η_3. Equations for the other terms have already been presented.

A graph of the overall efficiency, for a Split-Level system implemented using the minimum-size digital transmitter, is represented in Fig. 7.

With the new Split-Level implementation, the efficiency approaches 50 percent, as compared to separate amplification, whose efficiency would be 46 percent. The coupling coefficient for peak efficiency is found at 6.75 dB.

CONCLUSION

We have described Split-Level, an improved high-level combining method of upgrading to IBOC. This method uses a tighter coupling coefficient at the output combiner, which decreases the IBOC losses. Simultaneously, by injecting a small fraction of the FM signal, properly phased, at the coupled port, the total combining efficiency is increased, as both the main and the additional FM injection are partially summed at the output.

A coupling factor of 6 dB is the optimum value to minimize the size required for the digital transmitter and to reduce the dissipation in the reject load, and as such, boosting the overall system efficiency by 3 to 4 percent. Under these conditions, the total reject power is expected to be reduced by 50 percent.

All these benefits are in addition to the fact that with Split-Level Combining, the main FM transmitter does not have to increase its output level to overcome the coupler losses, an inconvenience unsurpassable for many installations in the field.

Finally, the system performance is dependant on the size of the digital transmitter in the following manner: the bigger the digital transmitter to be used, the stronger the FM injection and the tighter the optimum coupling coefficient can be, this way obtaining a lower output power demanded from the main FM transmitter, a lower reject power and a higher overall system efficiency.

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Wheatstone Corp. manufactures a line of consoles at a modern manufacturing facility in New Bern, N.C. Andrew Calvanese, vice president of Wheatstone, talked with us about his knowledge and views on the evolution of console design.

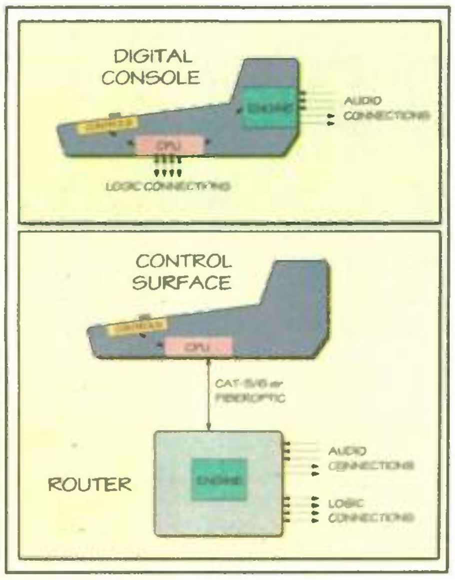

Fig. 1: Digital Console Vs. Control Surface/Router

Radio broadcasting has been in a transition from analog to digital technology for some time. How has this affected the broadcast console market? Are most customers buying digital or analog consoles at this point, or hybrids of these two technologies?

The transition to digital console technology has been going on now for roughly 10 years. As with any transition, this has caused uncertainty in the marketplace and created a new learning curve for everybody. I assume what you mean by "hybrids" is a console that has both analog and digital capability, which is mainly what is shipped today.

In and of itself, a pure analog console can offer higher levels of audio performance—especially bandwidth, noise and distortion—and will continue to do so in the future. However, in most real-world radio stations, the subsequent analog-to-digital conversion that happens when you connect this console to a modern signal chain negates this performance advantage.

What are the main reasons to pick one technology over the other?

Digital consoles typically have better transmission-related specifications, like crosstalk and off isolation (*crosstalk to mix bus when individual channel is off*). In the simplest of consoles an all-analog design will be less expensive, while the reverse is true for complex designs with preset and recall capability.

If you are redoing a single studio, have no digital input sources, have an analog input to your STL chain, and don't see that scenario changing soon, then a good analog console may be the best choice for you. But if any of those factors change you could be better served by a digital system. A hybrid stand-alone console — one that handles both analog and digital I/O — will give you more flexibility in the future and make it easier to interface new equipment.

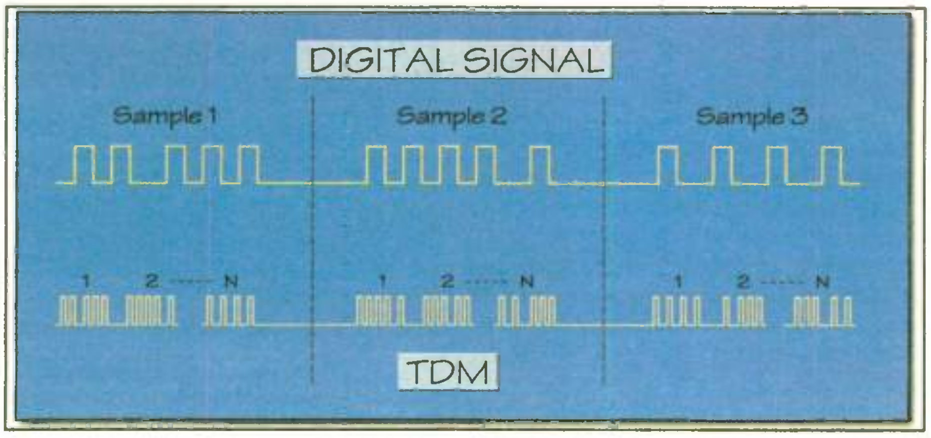

Fig. 2: TDM Transport Mechanism

A router-based digital system will do all that, and additionally provide an open-ended system that is easier to expand. It will also make it easier to repurpose rooms in your facility, via configuration presets, should future requirements dictate substantial change. More efficient use of sources, through source distribution and sharing, is also possible with the router-based designs.

How does the installation differ between these two technologies?

The installation for a digital console typically involves two parts.

First the physical installation must be done and cables connected. Digital signals need to be wired with a compatible cable type, either 110-ohm digital wire or CAT-5. With a router-based system, which uses control surfaces in the studio, the Time Division Multiplex bussing between router cages replaces discrete interstudio wires for each source. The number of point-to-point connections, especially between the individual studios and the rack room, is substantially reduced. What used to be 32 pairs of analog cable now is a single CAT-5 wire.

The second part of the install involves configuring the system through software. Here you configure various options such muting, tallies, timer restarts, etc. All of these options on a traditional analog console were set via DIP switches, which involved pulling individual modules to set the switches. This time-consuming procedure has been replaced by a screen with checkboxes. An entire multi-surface system can be configured from your PC in the rack room.

The router-based platform does involve more configuration work — naming of sources, setting the various permissions, creating presets, determining the source availability on each surface fader; those are things that didn't apply to stand-alone console models. We try to get customers to think along those lines and do preplanning before their system arrives.

Are all digital consoles the same? What specifications should a customer look for in a digital console?

Digital consoles have differences in performance, features and quality just as cell phones or sound cards do.

First, digital consoles can have basic architectural differences that affect performance and sound quality. Digital audio can be characterized from as low as 16-bit to as high as 24-bit and still be within the AES specifications.

We all know that basic audio CDs have 16 bits of resolution. The quality level of 16-bit audio is low enough to have encouraged the development of HD/DVD audio, and the problem gets much worse when you are in a live/production situation where audio is not as tightly controlled as it is in music production and mastering.

There are digital consoles out there that are operating on 16-bit data paths. Likewise the A/D (analog-to-digital), SRC (sample rate converter) and D/A (digital-to-analog) converters that are a necessary part of any practical console have varying levels of resolution and dynamic range from 14 bits to 24 bits. Wheatstone digital consoles use low noise 24-bit converters on analog and digital inputs and outputs for the highest possible audio quality; many other consoles use lower-quality converters.

Some digital consoles are based on, or require, personal computers to operate some or all functions, while others use a single-purpose embedded central processing unit. Some require hard disks while others employ solid-state memory. Generally speaking, a console with an embedded controller and solid-state memory is more reliable and easier to install and get running.

Second, digital consoles have varying levels of features and functions, just as analog consoles do. Are the inputs fixed or routable? Are they reconfigurable to accept different types of input signals, or is it necessary to modify the console to make a change? Can any input accept either analog or digital signals? How many outputs are available? How many mix busses are available? What logic functions are

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Split-Level

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ACKNOWLEDGMENTS

The author would like to thank his colleague Dr. Anders Mattsson, for helpful discussions and support during the statistical analysis of the envelope.

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available, and are they fixed or programmable?

As an example, Wheatstone manufactures more than 10 digital console models, just for radio with different combinations of these features, and I'm not including size variations.

Third, digital consoles are made with different levels of quality and support just as analog consoles have been. The heart of a digital console is still its user interface: the switches, faders, displays and meters that the operator uses. The quality of these items is an important consideration affecting the reliability of the console.

It is especially important with this digital technology, especially if it is new to you, to have a support team to contact that can help manage the technical details, like "Why can't I hear audio from the digital output on my CD player?" — the wrong type of wire or too many connections, for example — or "Why does my production recorded on my automation system sound weird when I play it back?" — wrong sample rate.

What is the importance of "32-bit floating point computation," which is sometimes specified on digital mixers?

To mix signals together essentially means adding them. In analog consoles the "summing bus," or Active Combining Network, was the mechanism for this.

In the digital domain, the same thing is accomplished by taking the individual digital words representing the required signals and adding the words together every sample period in binary math. One of the characteristics of binary math is that the resultant of this addition process has more digits, or bits, than the individual components do. These carry bits mean that your addition processor, the DSP, needs to have more bits available in its resultant — the "accumulator" — than the data words have themselves.

For different kinds of processing, like gain, level or EQ changes, which are all done in digital with binary math functions, this process occurs as well. The bottom line is that the DSP processor needs to be able to process a larger number of bits than the sample words themselves to avoid truncation or overflow errors. Some early digital consoles used 16-bit fixed-point processors — because they were the only ones available that would work with 16-bit/44.1k digital audio — with insufficient bit depth in their accumulators to prevent this type of error. Second-generation consoles came out with newer floating point DSP chips to get around it, while modern DSP chips are available either fixed or floating point with enough bit depth to avoid this problem.

The situation evolved much as did the first transistor amplifiers or the first op-amp consoles; they didn't sound very good until the technology advanced enough to get past the "just-barely-working" stage.

What type of test equipment would a typical station want to acquire in order to maintain a digital console system?

To install and maintain a digital system it is very helpful to have a portable device that can generate and monitor AES audio. There are a number of these available at reasonable prices.

As with most things digital, problems tend to be of the "signal not there" type rather than a qualification issue. Being able to move around the facility injecting and or monitoring a digital signal is important in tracing these problems. To the uninitiated, using a scope to look at an AES digital audio signal is not very helpful because it looks similar whether there is an audio content or not.

What is a router-based console and how does it work?

All digital consoles share a common structure, which consists of an audio mix engine of some sort — usually based on digital signal processing — and a control unit or CPU. The control unit directs the operation of the mix engine and a user interface consisting of the familiar switches, faders, displays and meters, which communicate the operator commands to the CPU. These structures can all be built into the same chassis in a fixed configuration, to create a stand-alone digital console, or they can be cleverly separated into different pieces, in which case you have a control surface-router solution. Each approach has different advantages, which is why Wheatstone makes both types. (See Fig.1 on page 15.)

Fig. 3: Laser Cutter in Operation

A control surface-router solution allows the individual components to be located where it is most practical. If you have a tech center where all of your automation servers, ISDN, processing equipment, etc., are located in common, locating the audio input part of your system in the same space minimizes room-to-room wiring that represents a sizable and inflexible expense. By separating the components of the system, the control surface part can be located in your studios and control rooms and connected with simple wiring — in the case of Wheatstone, standard CAT-5 cable. And if a significant number of audio channels are needed to connect between rooms, again a simple CAT-5 cable can transport many channels simultaneously.

A further advantage is that by separating the control surface from the actual audio, the audio can become part of the whole system, shared everywhere.

Consider the case of a satellite receiver that needs to be able to feed multiple rooms. With stand-alone consoles, you would need to wire the receiver to a distribution amp and then run individual wires from the DA to each room. In the control surface-router approach, you would wire the satellite receiver to the router once and then call up the signal in each room whenever you needed it. This signal — or audio resource, as we call it — becomes integrated into a multistudio, multistation system, where all audio resources and logic functions are available and controllable system-wide.

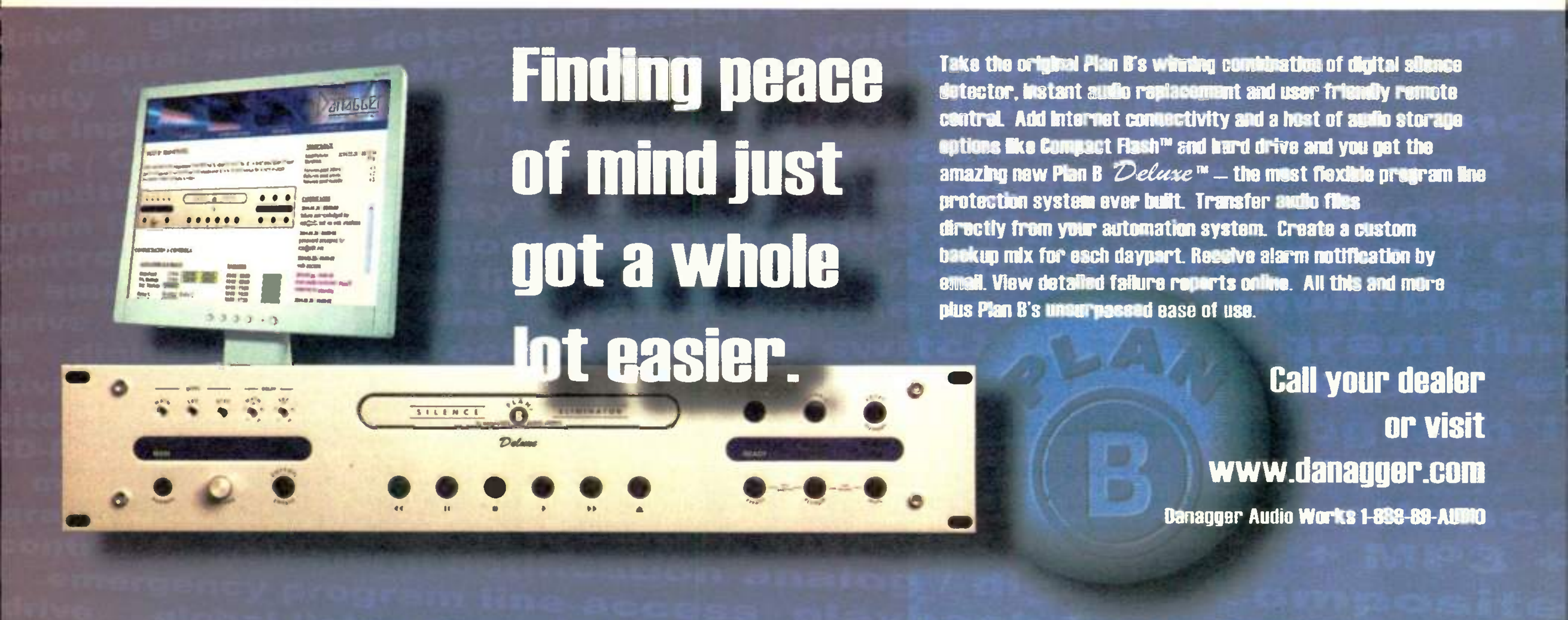
Furthermore, when requirements change in the future, as they do in radio, in a surface-router installation, studios can be added, moved or rearranged with a minimum of rewiring. A Wheatstone control surface connects to a system by means of a CAT-5 cable, as opposed to the dozens of audio connections and hundreds of GPI logic connections required for a stand-alone digital or analog console.

What are the key specifications to consider in this type of purchase and/or key performance metrics?

When considering a system of this type it is wise to look at it as an infrastructure project, like telephone, electric or IT. Does it have the capacity for present and projected use? Does it offer suitable functions, features and performance levels to be useful both today and years from now?

For example, Wheatstone Bridge router systems can route and process multichannel audio should surround sound be in your future.

Is the sound quality exemplary? Look for 24-bit A/D and SRC. How exactly will future changes to the system be made, and will the system need to be disabled to make changes? What cabling is required? Wheatstone recommends running a CAT-5 cable to every potential studio or control room to allow easy installation of a



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control surface or satellite router rack. Some other manufacturers may require more extensive wiring.

As a sidebar, most digital audio systems use some variant of TDM to transport digitized audio.

What is a TDM transport and how does it work?

TDM works by sending the individual data bits representing an audio sample serially at a much faster rate in a prescribed channel order over a single path or connection. When the transmitting circuits and receiving circuits use the same prescribed channel order for the data bits, the individual bits can be reassembled into the proper words representing the audio samples of multiple channels.

It works because the transport link is running at a much higher rate, or frequency, and hence has room within a given unit of time — a sample, or 1/44,100 of a second, for instance — for many more data bits. In effect, each sample period is divided into multiple segments or time windows, one for each of the channels.

A 100baseT — CAT-5 running at 100 megabits — cable has enough data bits to represent 64 discrete 24-bit audio input channels and 64 24-bit audio output channels plus control information. (See Fig. 2 on page 15.)

Who should consider a surface-router console system?

Anyone who is considering a consolidation or build-out project involving multiple rooms and/or new wiring should look seriously at a control surface-router solution for the increased functionality, flexibility and potentially lower total installed cost.

A single room upgrade with existing, usable wiring can be done less expensively with a stand-alone console. The multisurface, multistation resource, sharing hardware and software built into every control surface-router combination is an added expense that doesn't provide any compensating benefit or total installed cost reduction for a single room.

Radio engineers are concerned about reliability in their system designs. As a console system begins to function more like a computer network, what strategies can be used to avoid the possibility of a single-point failure taking an entire technical facility down at the same time?

Controlling single points of failure is a very real issue for today's multistation consolidation projects. When a router-based audio network is the backbone for the whole plant, it is especially important to choose and install a system that has the redundancy and recovery capability that you need.

Again it is wise to think of the system as infrastructure like your telephone system. You probably have your phones on an uninterruptible power supply and/or backup power. It makes sense to do the same with the router system.

Also, as a minimum, make sure that the central core of your system has redundant fail-safe power supplies, as these are still the least reliable part of a system. Add fail-safe power supplies to control surfaces and remote racks as your budget allows.

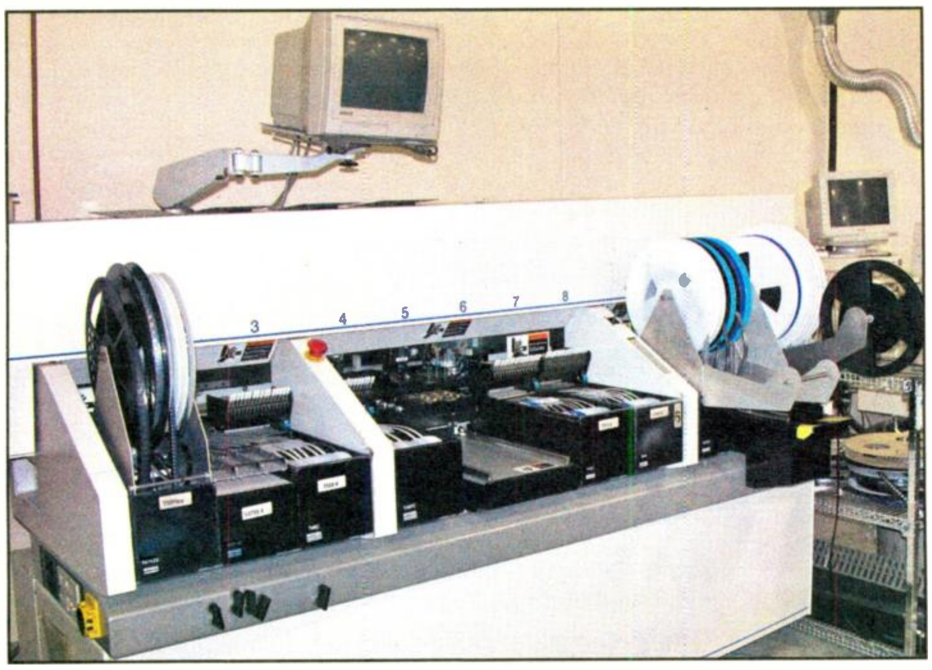

Fig. 4: Surface-mount Machine

Use defensive strategies in deciding your wiring and card configurations. We had a client whose tech room was struck by lightning, causing significant damage to a lot of his equipment. Unfortunately one of the damaged items was an output card in his Wheatstone Bridge system. Because he had wired all of the main program outputs from his multiple studios to this single card, all of his stations were knocked off the air when that card was hit.

If he had wired his program outputs to different cards, he would have lost only the one station. The lesson is to plan for the possibility of things going wrong and design your system with work-around or damage-control strategies in mind.

With the Wheatstone Bridge system, we can provide redundant automatic-failover CPUs for the router, and for each control surface as well. Additionally, redundant automatic-failover DSP engines can be installed. Our control surfaces and associated studio racks will continue to mix and operate even if they become disconnected from the main system. In fact, entire subsystems of the overall router — say for example, all your air studios — can be equipped with failover CPUs so that they can function independently.

Finally, because the Bridge system is based on embedded hardware, audio will continue to flow through the system even with the control surfaces turned off and unplugged. This doesn't sound significant until you get called at 6 in the morning because of a coffee spill into the control surface.

A system like this can be made as fault tolerant as anything else in your plant.

What new technology trends do you see affecting the broadcast console market, or broadcasting in general?

I hear HD Radio being talked about a lot. Also, more and more signals in the plant are becoming digital. Just a few years ago the majority of our digital consoles were shipped with mostly analog inputs; in fact many digital consoles were shipped that had no digital inputs at all [input channel type is configurable on Wheatstone consoles]. This is no longer the case. All of this points to the increasing role of digital technology in radio.

This brings up an interesting point: the switchover to digital technology has created the necessity for software development.

Digital devices need software to run. As a result, our development of digital products requires many hours of software development, testing, debugging and support, none of which was necessary to make an analog console. And because software is never "done," it can always be modified and enhanced — Microsoft spends billions of dollars and employs many thousands of talented engineers on software development, and still their products are never finalized and totally bug free — the effort continues after the products are sold. For reference, Wheatstone pays a large amount of money just for software support contracts to maintain the software that we use to design and build our products.

As digital radio consoles of all types are becoming more and more software-based, they will offer more options and configuration settings. A lot more decisions need to be made to design a studio system. Careful preplanning and close communications with the manufacturer will go a long way to getting these consoles to work the way you want.

As a manufacturer, what has changed in the last 10 years or so in the way that you produce new equipment?

The radio — audio — industry is a small part of the professional audio industry, which in turn is a very small part of the electronics industry. We're all familiar with the changeover to surface-mount technology and the proliferation of microcontrollers that has been going on in the electronics industry. The cell phones, DVD players, and computers we all use today would not even be technically possible, let alone affordable, without this changeover — although some might argue this is not a bad thing!

Automated production and assembly equipment, coupled with the ever-increasing power of microcontrollers, make possible meeting the expanding needs of customers.

This trend affects radio too. The feature set of Wheatstone's premier radio console of years ago, the A-500, is surpassed today by the simplest of our Audioarts analog consoles. Wheatstone has embraced this technology with integrated in-house computer-aided design/computer-aided manufacturing and automated surface-mount assembly for our circuit boards. We use computer numeric control machining, bending and laser cutting for our metal and woodworking. Just as surface-mount lines allow us to implement the complex circuit designs required in today's products, our laser allows us to execute designs

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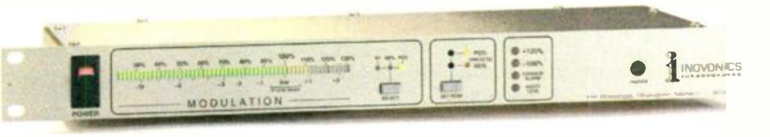

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in metal that weren't previously possible.

This CAD/CAM design capability gives us several important benefits. First, it makes possible design details that would be totally impractical otherwise. A modern Field Programmable Gate Array or DSP chip has hundreds of pins typically spaced twenty-thousandths of an inch apart and requires a four- or six-layer PCB just to function properly. These chips are necessary to create a digital console design. These types of chips are not available in through-hole packaging, and even if they were, trying to design a two-layer circuit board manually that connects all of these hundreds of pins at such small spacing would be essentially impossible.

Phil Owens and Andrew Calvanese, Wheatstone Corp.

Likewise with the mechanical parts, the laser and other CNC machines allow our designers much more flexibility in choosing details such as our custom speaker grill openings and radial bends for wrist rests without requiring expensive and long lead-time tooling.

Second, both our surface-mount and metal working areas are tightly integrated with our design departments. What this means is that as soon as the design is done in the computer (be it a PCB or a console frame), the design files can be directly sent to our manufacturing equipment and built without any secondary digitizing, set-ups or reprogramming necessary. This gives us excellent manufacturing control and repeatability, and also allows us to be extremely responsive. We normally have completed, working circuit cards in our hands the second day after the design is finished in the computer (the two-day delay comes from the manufacturing and shipping of the raw PCB itself) and new metal parts within an hour after design finalization.

Another benefit this technology gives us is that it becomes practical to manufacture products cost effectively in the relatively small batch sizes appropriate for the radio market, where a few thousand consoles a year might constitute the whole market. Our surface-mount equipment allows us to build 10 or 20 different types of circuit boards in a single day; our metal working equipment performs similarly.

All this allows us the quality control, production efficiencies, capabilities and responsiveness we need to manufacture the higher-performance, feature-rich, complex products in demand today. By bringing all these processes in-house we have created a vertically integrated company that is not at the mercy of outsourcing vendors. ■

Products & Services SHOWCASE

Grounding

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used SSM-2141 balanced line receiver will degrade some 25 dB with only a 1 ohm imbalance in the driving source.

However, balanced receivers having common-mode input impedances some 1,000 times higher are essentially unaffected by imbalances as high as several hundred ohms! A balanced input using either a quality input transformer or the InGenius IC is virtually immune to this degradation.[5] Each typically achieves 90 to 100 dB of CMRR, virtually unaffected by real-world balanced output imbalances, and typically over 80 dB of CMRR when driven by an unbalanced source!

CMRR Measurements vs. Reality

CMRR of balanced inputs have traditionally been measured in ways that ignore the critically important effects of driver and cable impedances. Therefore, actual CMRR achieved in real-world systems is often far less than that touted for the balanced input itself.

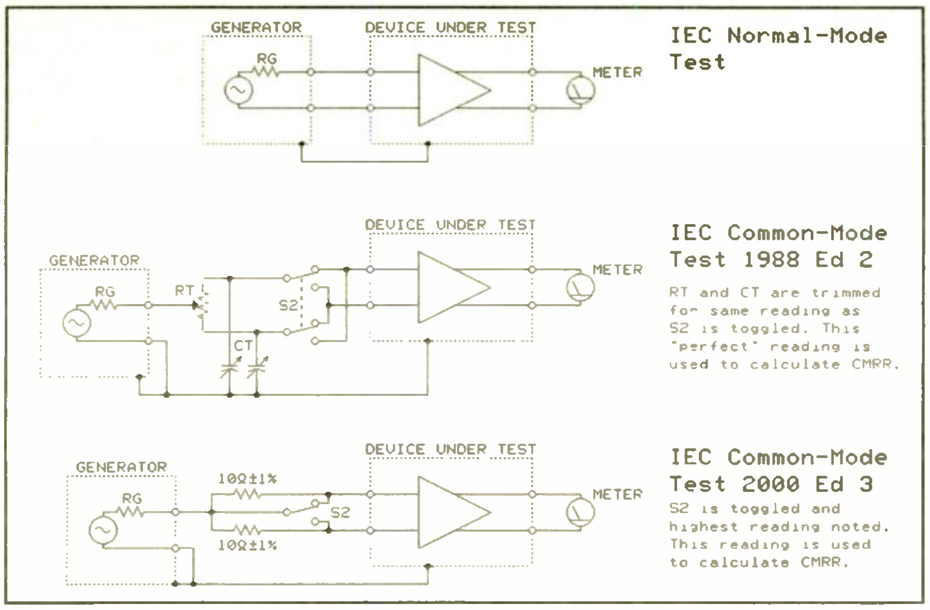

Fig. 3: Old IEC test, center, tweaked source until perfect.

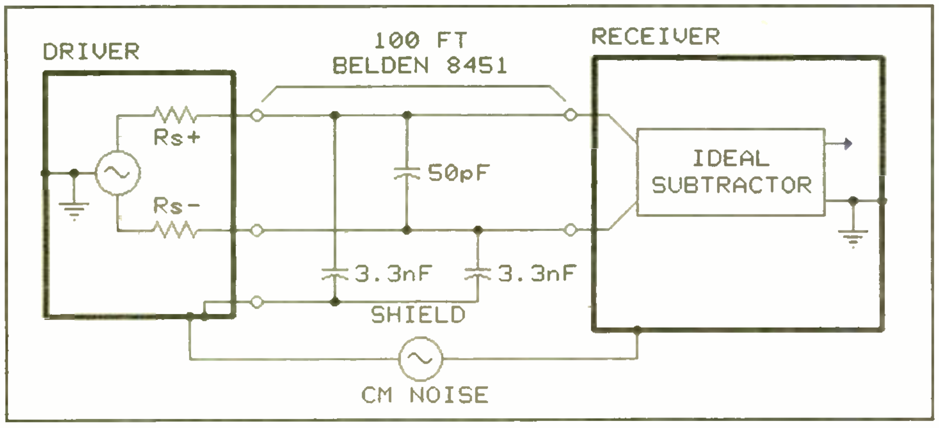

Fig. 4: Grounding shield at receiver forms filters.

As shown in Fig. 3, the old IEC method essentially fine-tuned the driving source impedance until it had perfect balance. Other widely used techniques, simply shorting the two inputs together or using high-precision resistors (as in most test instruments), are equally unrealistic measures.

This author is pleased to have helped persuade the IEC to change its measurement standard for balanced inputs and outputs. The third edition of "IEC Standard 60268-3, Sound System Equipment - Part 3: Amplifiers" was issued in August of 2000. The new method involves placing a 10-ohm imbalance, first in one line of the test source and then in the other. The lower of the two calculated CMRR figures is then used.

Effect of Cables Interface CMRR

It's important to understand that noise rejection in a balanced interface isn't just a function of receiver CMRR. Actual performance in a real system depends on how the driver, cable and receiver interact.

The shield, if present, should always be grounded at the driven end of a balanced cable. There are two reasons.

The first involves the cable capacitances between each signal conductor and shield, which are mismatched by 4 percent in typical cable.[6] If, as shown in Fig. 4, the shield is grounded only at the receiver end, these mismatched capacitances and mismatched driver common-mode output impedances (often by 5 percent or more) form a pair of low-pass filters for common-mode noise. The mis-tracking of these filters converts a portion of common-mode noise to differential signal. If, as shown in Fig.5, the shield is connected *only* at the driver, this mechanism doesn't exist because all filter elements are at the driver ground potential!

The second reason involves the same capacitances working in concert with signal asymmetry. If signals were perfectly symmetrical (equal and opposite voltage swings) and capacitances were perfectly matched, the two capacitively-coupled signal currents in the shield would cancel. However, imperfect symmetry and/or mismatched capacitances will cause net signal current in the shield. This current should be returned directly to the driver from which it came. If the shield is grounded at the receiver, all or part of this current will make its return via an undefined path that can induce system crosstalk, distortion or oscillation.[7]

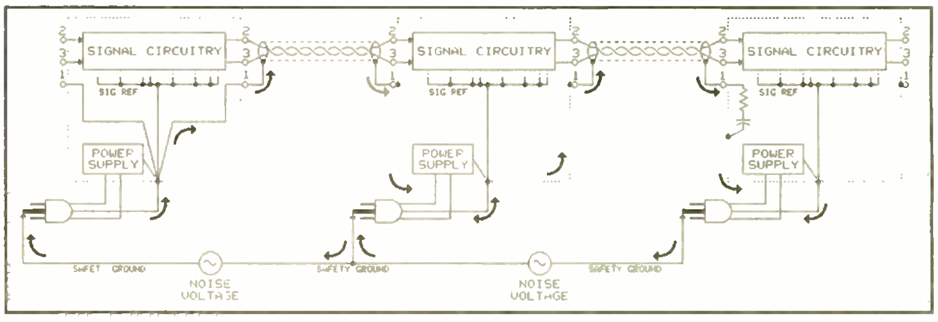

Fig. 5: Grounding shield at driver eliminates filters.

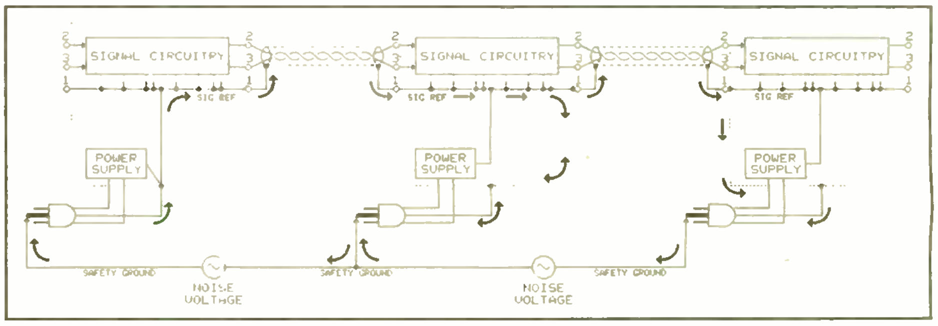

Fig. 6: Equipment Having the Pin 1 Problem

At audio frequencies, the ultimate in interface CMRR is achieved by grounding cable shields only at the driver. Ground voltage differences between system devices can cause considerable current flow in cable shields, creating a magnetic field extremely close to the signal conductors — unless the induced voltages are precisely equal, differential noise, called shield-current induced noise or SCIN, is created. Cables using *drain wire* shield construction have the worst SCIN.[8,9]

However, to guard against RF interference where typical system cables become a fraction of a wavelength, grounding the shield at more than one point is desirable. Therefore, a high-frequency treatment must be superposed on the low-frequency treatment.[10,11] This is a conflict between star and mesh grounding methods. One approach, called *hybrid* grounding, couples additional points on the shield to ground at high frequencies only through capacitors. After years of debate, the merits of hybrid grounding finally are being considered for incorporation into preferred practices being formulated by an AES Standards Committee working group.

In practice, I recommend never grounding only the receive end of a balanced shielded cable, and generally recommend grounding at both ends for two reasons: 1) if the equipment input has marginal RF suppression, grounding the shield at the input will usually reduce the problem; and 2) it doesn't require the use of a specially wired cable, which might, if not clearly marked as modified, find its way into another system and cause unexpected problems.

AC magnetic fields will induce voltages

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Grounding

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along the length of each signal conductor, and the magnitude of each voltage is a strong function of distance to the field source. Twisting of signal conductors is a first-order technique to minimize induced differential voltage by making the average distances to the conductors equal. So-called *star-quad* cable is a refinement that parallels opposing pairs of conductors, further reducing susceptibility by about 40 dB. Remember that wiring at terminal or punch-down blocks and inside XLR connectors is vulnerable because the twisting is opened up, creating a magnetic pickup loop.

COUPLING BY EQUIPMENT DESIGN DEFECTS

In a system, noisy ground currents flow from one piece of equipment to another through cable shields. The design of some equipment allows this current to flow in internal conductors, such as PC board traces, that are shared by high-gain circuitry. The result is common-impedance coupling into the signal path, which causes noise at the equipment output.

When this defect exists, the equipment behaves as if the shield connection were a low-impedance audio input. Since pin 1 is the shield connection for XLR connectors, Neil Muncy called this defect the "Pin 1 Problem" in his famous 1995 AES paper.[12]

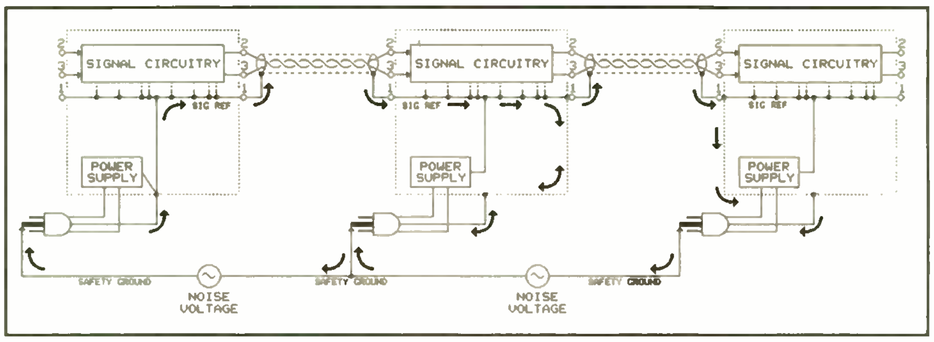

Fig. 7: Equipment Properly Handling Shield Connections

Fig. 6 shows a system where each piece of equipment has the problem, while Fig.7 shows several pieces of equipment that do not. The right-most device in this figure shows an example of the hybrid shield grounding technique mentioned earlier. At the time he wrote the paper, Neil estimated that up to 50 percent of commercial equipment is affected. Fortunately, the simple *hummer* test can reveal the defect.[13] This problem frequently exists in equipment with unbalanced interfaces, too.

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[13] *Windt, J., An Easily Implemented Procedure for Identifying Potential Electromagnetic Compatibility Problems in New Equipment and Existing Systems: The Hummer Test, Journal of the Audio Engineering Society, June 1995, pp. 484-487. For more information see www.jensentransformers.com/as/as032.pdf* ■

AM Noise

CONTINUED FROM PAGE 7

amplitude modulation with a bandwidth less than 100 kHz, therefore standard coax can carry the detector output a long distance to the monitor without any compromise in the reading accuracy. Modulation monitors connected to a sample using coaxial cable cannot be expected to produce meaningful synchronous AM readings.

IMPORTANCE OF SETTING UP AN ACCURATE SAMPLE

It may seem that I am over-stressing the importance of an accurate sample position and level. It is imperative to plan sampling and detection properly because an inaccurate sample can prompt severe mistuning of the transmitter and yield incorrect overall indicated levels of AM noise. When the sample is established properly, it can be relied on for real-time monitoring of transmission system integrity for many years.

The linearity of the detection circuit is equally as important as the RF sample. The optimum detector employs full-wave carrier rectification with LC filtering to produce an AM level that can be calibrated against the detected carrier level. The RDL DCF-100MB detector used with the ACM-1, ACM-2 and ACM-3 AM Noise Monitors relies on this method. Engineers have used half-wave diode rectifiers and suitable filter capacitors to produce samples that can be used as a relative tuning indicator when a quantitative value of AM is not desired.

HOW LITTLE IS ENOUGH?

Each station may establish its own threshold for maximum AM noise based on terrain, multipath in critical coverage areas and transmitter capabilities. In general, simply tuning for minimum AM without taking a calibrated reading can result in actual levels from -25 dB to -50 dB or better. Readings of -40 may be acceptable if there is minimal multipath in critical listening areas and no subcarriers are in use. The same value may be unacceptable in a competitive market with moderate-to-severe terrain or other reflection producing obstructions. Most systems can attain a sustained level of at least -50 dB with good maintenance. Levels of -55 dB or better will produce optimum station performance in any environment.

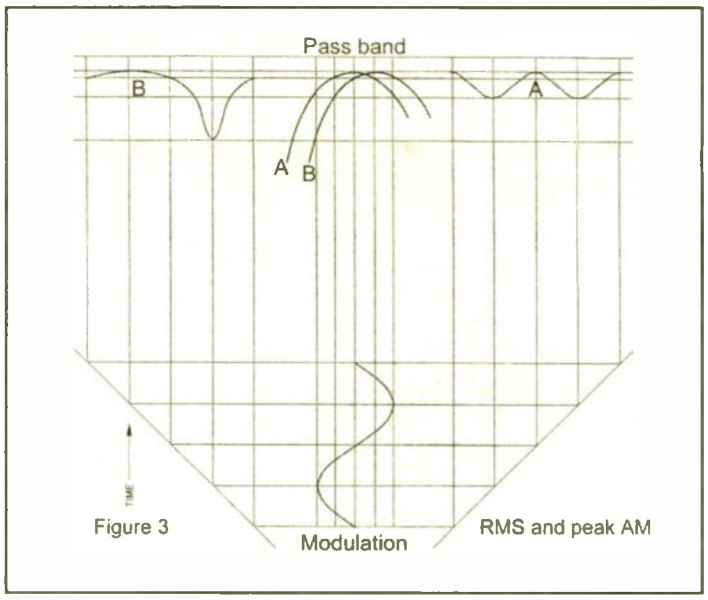

Fig. 7: Peak vs. RMS AM Noise

It is best if the station can monitor the level constantly, allowing a predetermined threshold to trigger an alarm. Changes in the bandwidth caused by any source such as tube aging, variations in power service voltages, temperature and transmission line or switch heating can alert the station of impending maintenance needed before the station's service area or subcarrier performance is affected noticeably. In selecting the proper threshold, it is normal to allow for a variation of 5 dB or more during each broadcast day.

Monitoring is always desirable though not always possible or practical. Today's monitoring system costs are within most expense budgets, but some facilities prefer periodic measurements by engineering staff. Other sites share common antennas with combiners that cannot sufficiently reject neighboring carriers to allow constant monitoring. At such sites, AM noise should be checked individually on each transmitter during a coordinated maintenance period. That period can also be used to check the antenna's effect by measuring and logging the AM noise on a *reflected* carrier sample. The low amplitude of the reflected signal should require a more sensitive sampling slug. Such a maintenance check is equally beneficial to stations that do not share a common antenna.

PROPERLY TUNED AND OPERATING PASSBANDS

When a transmitter and associated components are tuned and operating properly, a waveform similar to Fig. 5 will be produced by a detector or at the output of an AM noise monitor. With this display of centered tuning and a measured AM level of -55 dB or better, the station is assured of optimum performance.

Good performance may also be possible with a passband that is not perfectly centered, provided the measured AM level is sufficiently low. Frequently, offset passbands as indicated in Fig. 6 produce excessive AM noise levels.

QUALIFYING AND INTERPRETING THE READINGS

If an AM noise measurement method is being used that does not produce a calibrated level, care must be exercised in qualifying the results. An important characteristic to understand is that as AM noise becomes worse, typically the peak content increases while the RMS energy remains the same or decreases. Therefore, metering that produces RMS or averaged results will become more inaccurate as AM noise increases. Receiver performance, however, will degrade as peak AM increases. This makes it critical to monitor the *peak* AM excursions.

Fig. 7 details the problem. If the peak value of AM indicated by passband A were measured at -39 dB, an RMS meter would indicate the same waveform at a level of -45 dB. If the passband were then tuned for an RMS null, waveform B would result in an RMS reading of -46 dB. However, the peak value produced from passband B would actually be -32 dB. Nulling the RMS reading would actually result in performance degradation in AM noise from -39 dB to -32 dB!

Similarly, if a common diode is used as an RF rectifier, equally erroneous results can easily be produced. A 1N4148 fed with 10 V peak-to-peak at 100 MHz with 10 percent applied AM modulation will produce a DC/AC ratio of -20 dB, which is an accurate indication of AM. The same diode fed the same signal at 4 V peak-to-peak would result in a reading of -17.9 dB, while with a carrier sample of 1 V p-p it produces a reading of -9.8 dB. This is an error of more than 50 percent. Accurate, repeatable AM noise measurements are improved through the use of low-capacitance high-frequency diodes.

BENEFITS OF MONITORING AND CONTROLLING AM NOISE

Nearly any anomaly in the performance of the transmitter and associated RF plumbing has an immediate and measurable effect on AM noise. When the accurate AM level is monitored continuously against a threshold standard, the station can be assured of proper performance. Control of AM noise optimizes subcarrier operation and any broadcast services relying on signals in the upper spectrum of the baseband. Stereo separation relies on the 38 kHz subcarrier, and is materially worsened by increased AM levels.

Proper sampling and monitoring of AM noise levels produce coverage consistency, maximum service radius, improved stereo separation and satisfied subcarrier tenants. It provides the final "QC" for the station's carrier signal, program content excepted. ■

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Lessons in the Wake of Ivan

By Stephen Poole

Some people think that it would be fun or exciting to go through a tropical cyclone. They'll plan hurricane parties and gibber that it'll be something to "tell the kids about."

I assure you, that feeling lasts only until they've been through their first real major hurricane. Here in Alabama, Ivan left a large number of sadder-but-wiser folks in its wake.

Sandy and I moved to Birmingham from coastal North Carolina, and we had been through several storms while we lived there. The last major hurricane we experienced was Fran; my memory of that one is spending the night huddled in the dark, listening to the wind howl, to the freight-train rumble of tornadoes, and to the sound of trees breaking all around. The next morning, we drove around and were stunned at the damage. God had obviously looked after our little home. There were twisted and mangled trees directly in front of and behind the house, evidence that at least one storm-spawned tornado had "jumped" over our residence.

We lived in Hoke County, N.C., to the west of the I-95 corridor. We didn't even go through the worst of Fran, but what we experienced was bad enough. We were without power for several days. There were downed trees and power lines blocking many of the roads. Some of our friends and neighbors had suffered significant damage to their dwellings.

IVAN THE TERRIBLE

Ivan caused catastrophic damage along the Gulf Coast when it came through in mid-September. It knocked out power to over 800,000 people in Alabama (the highest number ever recorded). Among other damage to broadcast facilities in the state, Channel 15, in Mobile, lost its 1,850-foot tower, which caused a chain reaction that knocked several other stations off the air.

Ivan had weakened before it reached Birmingham, but still caused significant problems here. Some local stations went off early in the storm. The WDJC(FM) site on Red Mountain was one of them. We lost power around 10:30 a.m. on Thursday and didn't get it back until Sunday morning. The BE FM-30's exciter and IPA both have 50 ohm outputs, so my assistant Kenny and I tried to rig for low power on a small generator.

We soon had to give up and run because the wind had become so strong. A good thing, too. Once the storm began to subside and we were able to head back up Red Mountain, we discovered several downed trees, including one right across our road. We would have been trapped if we'd stayed.

As I write this, everyone is back on the air and all is well. We only suffered minor damage here, for which I'm grateful. But since it's fresh in my mind, I'd like to share some severe weather tips. I realize that most of you are in areas that don't normally experience tropical cyclones, but much of this is general, common-sense advice that applies for most severe weather scenarios.

Stephen Poole

The author is a chief engineer for Crawford Broadcasting Co. and is based in Alabama.

NUMBERS CAN FOOL YOU

The number one tip that I can give — urge upon — you is not to fall into the same traps that the newsies do — in the case of hurricanes, they concentrate on wind speeds and where the storm will make landfall.

Ivan caused catastrophic damage along the Gulf Coast when it came through in September.

First of all, hurricanes are huge. A tropical cyclone can be the size of a state, meaning that you may feel its effects for many, many hours as it slowly moves past you. The storm-force winds can extend hundreds of miles from the center.

Thursday morning, Sept. 16, the day of the storm, I woke up early so that I could get started before the storm got really bad. We were already feeling a strong, gusty breeze here in Birmingham from Ivan, while the center was still on the Gulf Coast, hundreds of miles away.

The newsies, naturally, get more excited about strong storms, Category 3 and higher. The major storms make better pictures because they're more likely to take off a roof or smash a building. (Plus, they get to put idiots on camera to stumble around and scream gibberish in the high winds.)

That's unfortunate, because it can create the impression that "weaker" storms — in quote marks for a reason — aren't as dangerous. Remember, especially once that storm moves inland, the biggest dangers are flooding, wind gusts and spawned tornadoes, in that general order.

Tropical storms also vary greatly in size and speed. Andrew was a small, intensely powerful system when it hit Florida back in the early '90s. The damage was catastrophic, but was limited to a wide swath across the peninsula. Ivan was one category weaker than Andrew, but caused damage over a much wider area because it was much larger.

More on those wind gusts. You might watch your local weather channel and relax because the meteorologists are saying that you're "only" going to have 40 to 50 mph winds. Remember that those are sustained, one-minute averages. The gusts can be much stronger, especially at higher elevations. (That applies in general, too, not just to tropical cyclones.)

Now you know why your STL dish could be honked out of alignment by a "mere" 40 mph wind down at ground level — and you also know now why you should always require that your installer put a stabilizer bar on that dish, even if they insist that "the clamps are tight enough!"

The bottom line is, wind gusts are completely unpredictable and can vary according to terrain. What you're experiencing on the ground might be no big deal, but that antenna atop your 1,000-foot stick might be hanging on for dear life. As mentioned, tropical systems regularly spawn tornadoes too, which just adds to the chaos of swirling eddies, sudden gusts and the ever-present, constant, sustained wind pressure.

PLAN YOUR ESCAPE

One other tip, one that again applies across the board to hurricanes, winter weather, you name it (and this is another one that the newsies don't cover adequately).

Your location might not experience the worst of the storm. Maybe you're in a cul de sac or a cove that blocks most of the wind, for example. But plan for the worst. Have several routes mapped out to all of your sites. It is common for roads to be blocked after a severe storm, especially in rural areas. Don't just assume that once the storm passes, you'll be able to jump in your truck and drive directly to a tower site. That road may be underwater or have trees and downed power lines all over it. Plan for that!

Special mention, once again, goes to my assistants: Kenny, who ran around with me during the peak of the storm, and Todd, who stayed late the evening of the storm to keep things running after Kenny and I collapsed from exhaustion.

Finally, be prepared for the usual idiocy.

My favorite moment from the storm occurred here at the studios, just as the wind was at its peak and we were operating three stations on a portable generator. The engineering department was running like mad keeping things up, working around outages, going to backups as needed, and so on. One of our show hosts came into the control room to complain.

"Why do the phones keep dropping out from time to time?"

I couldn't resist. I looked at him, smiled and yelled, "I don't know ... could it be that we're *in a hurricane?!?*" I don't know if he got the point, but he didn't complain about the phones again. Thank heaven for small favors.

Until next time!

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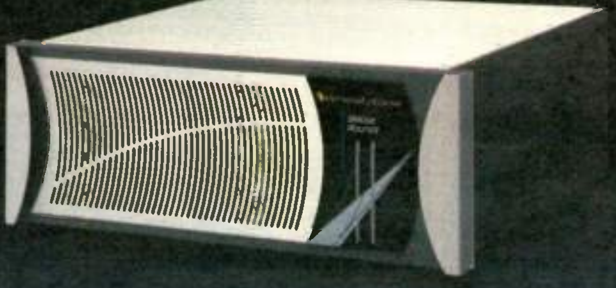

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