ENGINEERING EXTRA

Cris Alexande...

RADIOWORLD

FEBRUARY 17, 2010 | In-Depth Technology for Radio Engineers | $2.50 | RADIOWORLD.COM

NewBay Media

A Low-Loss IBOC Combining Method

Small Says Myat's Filter Technique Is Ideal for Higher HD Radio Power

BY DEREK J. SMALL

The author is director of filter products for Myat Inc.

Increasing the IBOC digital sidebands from –20 dBc to as high as –10 dBc presents more challenges for broadcasters wanting to transmit from one antenna.

WHITEPAPER

At –20 dBc, transmitting from one antenna requires digital and analog signals to be combined one of several ways: low-level or common amplification, high-level (couple digital power to transmission line at 10 dB), or mid-level, which uses a combination of common amplification and hybrid coupling to minimize FM and digital losses.

Common amplification is the most efficient and cost-effective way to combine FM and digital sidebands; however, with increased digital sidebands, it becomes difficult to use common amplification in a linear mode at powers above 20 kW due to peak voltages.

At the 2009 NAB Show in Las Vegas, Myat introduced a patent-pending technique for combining FM and digital sidebands for IBOC transmission to a single antenna. The system uses filters to combine digital sidebands operating in MP1 or MP3 more efficiently than high- and mid-level techniques, and is ideal for increased digital power.

The low-loss IBOC combining system utilizes the classic directional filter circuit shown in Fig. 1, and consists of two quadrature hybrids, two identically

(continued on page 10)

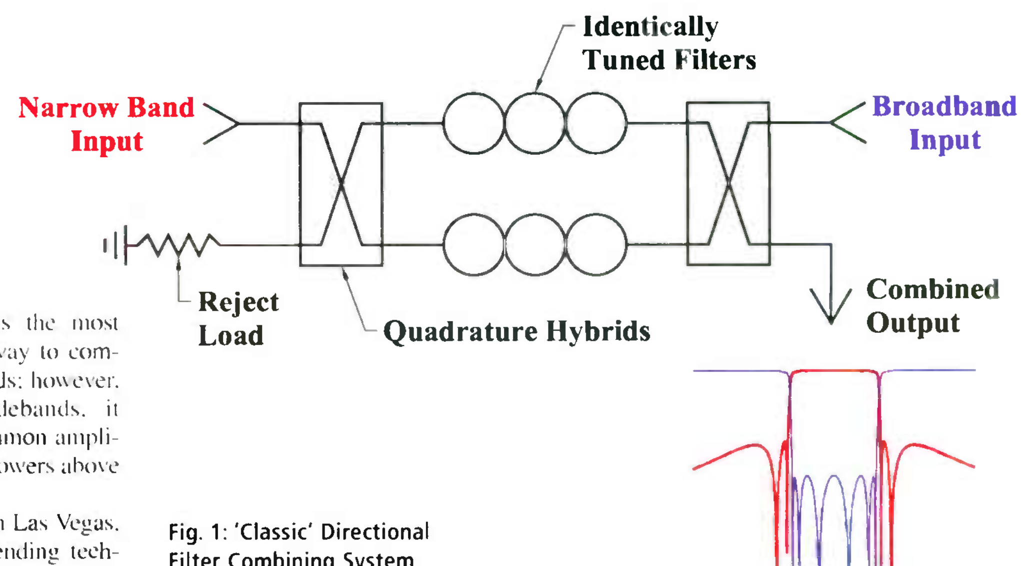
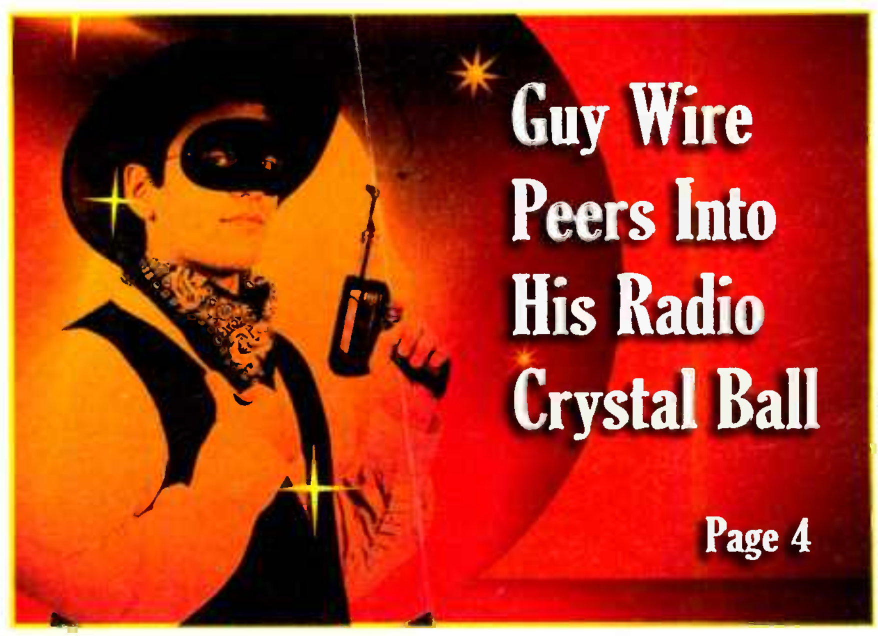


Fig. 1: 'Classic' Directional Filter Combining System



FINALLY...FAILSAFE NETWORKED AOIP.

WH ATST N

When it comes to networking your facilities, there's only one best choice. Wheatstone. For years, we've been the network/control surface choice of top broadcasters. And with good reason – we care.

Wheatstone's Audio-Over-IP product is the best in the business. Here's why:

1. WheatNet-IP is easiest for a station to implement and configure. It is, hands down, the easiest in the industry. No need for Wheatstone to provide factory on-site assistance unless you really WANT us there. The manual and app notes will have you up, running and stable in less time than any other system.

2. WheatNet-IP is a natural for large facility multi-station networking (and for smaller facilities too!). It uses the IGMP features of Ethernet Layer 3 switches to identify a multicast packet, see which ports are requesting that packet, and send it only to those ports. Traffic control is maintained and system bandwidth is optimized.

3. Redundancy is critical. A typical WheatNet-IP installation has multiple levels of redundancy. Each BLADE holds the complete map of the entire system within its onboard memory – we call it distributed intelligence – a system with 50 BLADEs has 49 backups with failover in the event of a failure. Cisco Stackwise technology provides redundancy in the central core TOC switch. A WheatNet-IP/E-Series console studio complex can stand alone, even if the TOC goes down, with backup analog or digital program audio feeding a back end router independent of the core Gigabit infrastructure.

4. Modular is better. Why would you want to combine your switch, mix engine and I/O into one box? Beats us. With WheatNet-IP, you install only what you need, where you need it. We believe in not overselling.

5. Manufacturing quality is very important. Wheatstone is proud to have the best track record in the business for build-quality, reliability and intelligent functionality. With far more up-and-running installations than anyone else, this is where we really shine. An investment in WheatNet-IP and E-Series control surfaces today will reward you with a future-proof, failsafe networking/control environment that's infinitely updatable and in for the long run.

6. WheatNet-IP has an advantage. Take a look at your entire environment. Wheatstone is a perfect partner because we are always there, always innovating. Built into every WheatNet-IP BLADE are features others just didn't think of – handy utility mixers, silence detection, crosspoint routing control, headphone monitoring of any source, lots of logic GPIO, and comprehensive metering of audio I/O, not just signal-presence indicators. And, in the hugely unlikely event that a BLADE needs to be replaced, you just plug in a new one and enter the BLADE number. That's it.

7. Wheatstone is local. WheatNet-IP and the E-Series, just like ALL Wheatstone products, are designed, engineered and built from start to finish in our New Bern NC USA facility. Everyone who works on our products is 100% knowledgeable and immediately available. You can relax – as with the famous insurance company, you ARE in good hands.

With WheatNet-IP, we think we've done our homework. In fact, we know we have. And we're happy to say that we've got the best product on the market. To learn more, and there's a LOT more, get us on the phone or visit us on the web. We'll be happy to meet with you and get you everything you need.

Wheatstone

Audio Networking—Simply Evolved

phone 1.252.638-7000 | www.wheatstone.com | sales@wheatstone.com

RADIOWORLD
ENGINEERING EXTRA
ol 34 No. 5 February 17 2010

Next Issue of RADIO WORLD March 1, 2010
Next Issue of ENGINEERING EXTRA April 14, 2010

E mail: rwee@nbmedia.com
Web site: www.radioworld.com
Telephone: (703) 852-4600
Business Fax: (703) 852-4582
Editorial Fax: (703) 852-4585

or address changes and subscription renewal, please visit adioworld.com and click on the "Subscription" button. o submit letters or story proposals, to request writer's uidelines, or for other editorial matters, e-mail the editor t radioworld@nbmedia.com.

Radio World Founded by Stevan B. Dana

Radio World (ISSN: 0274-8541) is published bi-weekly with additional issues in February, April, June, August, October and December by NewBay Media, LLC, 810 Seventh Avenue, 27th Floor, New York, NY 10019. Phone: (703) 852-4600, Fax: (703) 852-4582. Periodicals postage rates are paid at New York, NY 10079 and additional mailing offices. POSTMASTER: Send address changes to Radio World, P.O. Box 282, Lowell, MA 01853.

REPRINTS: Call or write Caroline Freeland, 5285 Shawnee Rd., Ste. 100, Alexandria, VA 22312-2334; (703) 852-4600; Fax: (703) 852-4583

Globe graphic ©iStockphoto.com / Edward Grajeda

How It All Gets Done

A Planned Studio Renovation Has Me Thinking About Project Design

BY MICHAEL LECLAIR

It looks like I will be taking on a substantial studio renovation this year. Some money has opened up, after a couple of years in which capital projects could not be funded. Here's hoping more of my colleagues will hear similar good news soon.

For me, and probably you too, large projects are rewarding work. We get a chance to show what we can do with the right resources and time to plan, unlike the daily repair cycle, which is mostly just reacting to rapidly changing circumstances with troubleshooting skills, flexibility and efficiency.

On a big project, we become designers rather than maintainers.

WHERE TO START

Studio projects are not like other engineering work.

Only a relatively small number of people will use or even see your new transmitter or automation system. Few want a tour of a tower site. The main goal in such projects is to build a reliable, high-performance system while trying to keep costs at a reasonable and efficient level.

> **On a large project, we become designers rather than maintainers.**

Studios, on the other hand, are used by a lot of people and for many purposes. These facilities present a public face of the organization during listener tours (which are very popular at our station). Other factors, such as ergonomics, come into play, because a studio is meant for people, who perform better when they are comfortable with a particular space.

There can be many competing goals and corresponding need for tradeoffs. Even the possibility of future programming changes should be considered, since typical studio builds have a life of around 10–15 years before new equipment begins to make sense. That is a long time.

If a project affects a large group, I like to start by offering to meet with as many of the studio users as possible to talk about what they need and what they would like. This step is crucial to my understanding of exactly what people need the technology to do for them.

The question is by no means straightforward. When I get a group of users together to talk about their programs, I always am surprised by the kinds of comments I hear about what works, what doesn't and why. While I pride myself on being up on the latest in technology, this doesn't mean I am working with the details of how a radio show is produced on a daily basis.

Users amaze me, both with their technical sophistication and their ambitious ideas on how to create good radio. Well OK, not all users; but as a designer it is my role to offer out the right palette of technological tools that helps them accomplish their goals. Without an open exchange of ideas, it is hard to know what is needed and what the goal of the project should be.

Some of the best ideas come out of these open meetings.

DON'T FORGET THE OWNERS

Meeting with the users is important but only a part of helping to define the goals of a project.

The central task in completing any successful project is to understand the interests and requirements of the owners, who are working to provide the money and staff resources that allow it to happen.

In classic project management, the term for this group is "key stakeholders." Key stakeholders tend to be at the managerial level rather than users (although not always, as in the case of highly paid air talent). Typical key stakeholders might be the general manager, program director or head financial officer. Your direct supervisor usually is a key stakeholder in the project. Without the support of at least most of this group, a project can be delayed or even founder completely. It is essential to understand this group and to work with them to provide what they need.

And this is where things can get tricky.

Key stakeholders are people, not roles in an organizational chart. They have their own competencies and insecurities.

A particular individual may want to be involved in ways that do not reflect his or her position in the organization. Some key stakeholders have more authority than others, meaning that in a conflict of ideas, the higher authority position generally will prevail.

Meeting with users to find out what works and what doesn't is a crucial step in designing a new space.

A sophisticated designer develops the skill to figure out what the key stakeholders want and which ones matter most, sometimes to the point of having to read between the lines of what is said or written. This is never an easy task, especially for engineers.

To make matters more complicated, key stakeholders tend to be the busiest and most distracted people. If asked directly about what they think, they may not have time to tell you much of anything. Most are used to delegation and practice it in every aspect of their lives.

Key stakeholders rarely want to participate at the ground floor, but want to come in when a lot of planning and thinking has been done already. This allows them to use their expertise most efficiently. It also virtually guarantees

(continued on page 6)

THIS ISSUE

FEBRUARY 17, 2010

Predictions for a New Decade

Are We in for Another Rough 10 Years?

BY GUY WIRE

What a relief it was to see 2009 and its tumultuous decade pass into history. The past few years and the worst recession since the Depression have been especially painful for all broadcast media. But the turn of the calendar always offers renewed hope, new opportunities and a chance to let the crystal ball reveal the future.

Your masked radio soldier of solder and syntax usually makes predictions this time of year. Let's not worry about the bad news and the doom-and-gloom stories about radio dying. Let's concentrate on what can happen if we all keep focused on adapting our business to the trends and currents moving our listeners forward.

By almost all recent surveys and polling accounts, terrestrial radio continues to enjoy very wide appeal and is used daily by about 228 million American consumers. The Internet may be our most vaunted competitor but it's also a valuable partner that extends our over-the-air services. In 2009, the growth of wired Internet connections in the United States started to level off.

OBAMANOMICS TO THE RESCUE

The new economic order of government bailouts, tight money and declining asset values has changed the way business does business for just about everyone. No longer do big failed companies liquidate and just go out of business. Nor do competitors buy them up at fire-sale prices to save their brands. Lenders are not using their bailout and stimulus money to make loans, so there's very little buying and selling.

What this means for radio is that companies like Citadel/ABC, Cumulus, Clear Channel, Regent and a few others should just reorganize and let the investors take the hit. CEOs and managers who made bad deals and incurred mountains of debt still keep their jobs. Their stations, already stripped down to the basic necessities of reduced staff and resources, just keep on keepin' on. As long as radio can generate decent cash flow, we'll see more debt-ridden groups reorganize.

It also means Sirius XM satellite radio is here to stay. While its subscription growth has slowed to a crawl and they've never made money, there is just enough demand and promise to keep the product available as a dashboard option. That's enough for investors who are willing to keep supporting it.

When wireless IP radio reaches critical mass, Mel's successors will turn off the satellites and rely on the Internet. The delivery platform is not the key ingredient in their product. Programming content is. When IP does become fully viable, Sirius XM will merely drop "satellite" from its name and turn off the birds, saving a ton of money.

Terrestrial radio will also encounter a similar crossroads, possibly even sooner than satellite, but it's not all that important for us to worry about when it will actually happen. Successful stations that have already developed their digital assets with a strong Internet and new media presence are well positioned to be in the mix.

The challenge, now and forever, is keeping the content fresh and compelling while continuing to superserve our local markets. That job will be easier if we properly leverage the technologies available to us today.

EVENTUALLY, MAYBE, SOMEDAY

Almost everybody is predicting wireless IP ultimately will be the delivery platform for all mass media. But there are too many issues that have yet to be sorted out before we'll have any clear idea of when that's going to be possible.

Most of the challenge is driven by economics. Even in the large test markets, the nascent efforts of companies like Clearwire and the various cellular carriers trying to build out the needed infrastructure are gaining traction much more slowly than most observers expected.

There will likely be several more rounds of consolidation in the wireless arena and another generation or two of technology advance beyond 4G needed before we get there. Realize what's required: seamless coverage and huge bandwidth capacity that can sustain literally millions of high-rate simultaneous connections in most all significant markets.

> **Almost everybody** is predicting wireless IP ultimately will be the delivery platform for all mass media. But there are too many issues that have yet to be sorted out before we'll have any clear idea of when that's going to be possible.

The demands of streaming video content likely will swamp radio's need for streaming audio bandwidth. My crystal ball says making this all work the way consumers will need it is still several decades away.

THE HERE AND NOW

We can best effect change and improvement for our stations by how we attack the immediate challenges and opportunities at hand. There are too many radio stations that have yet to deploy many of the digital technologies available now to augment their core products. Along with RDS, interactive Web sites, Internet streaming, social networking tie-ins and audio/video podcasting, HD Radio has emerged as a key opportunity.

After almost two decades of development, I'm convinced HD has finally turned the corner of acceptance outside the industry. Say what you will about increased interference and cannibalizing ourselves with more channels, HD Radio for FM offers entirely too many impressive new improvements and features to be ignored or rejected. The newly approved digital power increase will only help ensure its long-term success.

There are now well over 100 HD Radio models available for purchase, many at prices under \$100. Several pocket models are among them, including the impressive Insignia, now selling at discount for under \$40. There really isn't much of an excuse anymore not to buy an HD Radio receiver.

IBiquity's Bob Struble knew all along the key for HD success was getting the automobile industry fully on board. The CES last month offered a convincing display of additional support. Ford is leading the way, along with 15 other car companies that will add HD in the factory-installed standard radios of many of their 2011 models.

If your station's transmission system is at or near the end of its life cycle and you haven't yet converted to HD or updated for the power increase, buying a new transmitter or antenna is still a wise investment. There's plenty of time for the current generation of equipment to be fully amortized before you have to worry about wireless IP taking over.

RADIO DARWINISM

Despite the opportunities available to FM stations that will help us stay competitive in this Internet-dominated world, our AM brothers are left holding the short stick. Even though HD offers higher fidelity, AM HD is not likely to gain many more converts. There are simply too many forces conspiring against the senior radio band.

The new wireless handheld devices that are finally including radio like the Zune, iPod Nano and new cell phone models don't play AM. And none of the combo Internet/FM wireless desktop radios do either. By law, it used to be that all radios had to include both AM and FM. By necessity and design, AM is now being engineered right out of most of the truly exciting and attractive

(continued on page 6)


Crystal Ball © iStockphoto / bulent gultek

PREDICTIONS

(continued from page 4)

electronic gadgets consumers are buying.

Natural attrition and survival of the fittest will continue to whittle down the ranks of AM stations. The smaller privately owned stations that have not maintained vitality and relevance with their communities and have not kept up with technology trends truly are an endangered species. Any business that loses relevance and the ability to compete should be allowed to fail and let a superior technology platform replace it.

The best that AM can hope for as a valuable service going forward this decade will be for the stronger stations to seize every opportunity they can find to extend their brand presence. Some will improve coverage by expanding directional patterns and buying off other stations that limit their allocations potential.

Look for more of the successful news, talk and sports AM stations to also appear as FM HD2 and HD3 stations. Some will replace marginally performing FM formats in co-owned clusters, while others will add suburban FM simulcasts. The quiet migration has been underway for a few years now.

The proposals from the Minority Media and Telecommunications Council and the Broadcast Maximization Committee calling for TV Channels 5 and 6 to be reallocated for radio expansion and AM migration are logical and compelling.

But don't hold out any hope this will ever be adopted. The TV counter lobby will neutralize the effort and radio will be stuck with its existing bands. The FCC will spend most of its time and attention carving up UHF TV channels for wireless broadband providers.

Arbitron's PPM revolution already has spawned a number of major changes in the markets where PPM has become the audience measuring stick. Middays are now as important as mornings and many expensive morning shows have been jettisoned. More music and less talk seem to be imperative to hold audience; yet successful talk stations in most markets are holding their own.

We can only hope radio's decision makers will not let their cost-cutting knives cut too deeply into the last remnant of what separates us from jukebox automation and Internet radio: live and local real personalities who connect with their audiences in refreshing, enlightening and entertaining ways each and every day.

I predict the smart ones will use restraint.

Guy Wire is the pseudonym for a veteran broadcast engineer.

Comment to rwee@nbmedia.com.

STUDIO

(continued from page 3)

that a design process will involve repeated iterations as ideas are collected, assembled and then reviewed by the various layers of key stakeholders. That's another way of saying that a lot of design work is done and then gets thrown out if the key stakeholders don't end up liking it.

However, the more iteration, the further the design can get from the original intentions of the users. Communicating final plans to the end users needs to be handled delicately, since it will not be possible, or even desirable sometimes, to include all of their ideas in the final design. The outcome, if not handled properly, can be a studio that no one wants or likes, even though the key stakeholders signed off on it.

SUCCESS IS POSSIBLE

In practice, while what I just described above can be discouraging, the fact is that the process of managing a successful project design is rewarding.

Yes, there are moments when you have to throw out a lot of work and alter your thinking, but this is how projects work. Key stakeholders may be frustrating in some ways, but most did not get to where they are without being fairly intelligent, and they understand the most fundamental limits on a particular project. They usually will listen to a well-stated argument and come around to a different point of view if it makes sense. It does not seem to be too steep a requirement that, as designers, in the end we need to make sense.

Design for studios boils down to making the best possible facility to accomplish as much of what you want to do as is feasible. As such, there generally will be significant tradeoffs in the final product. For us as engineers, this is actually understandable and reasonable. We are used to the limits of physics, and tradeoffs are just another way of looking at physical limits.

Right now, I am just in the beginning stages of this project. As this year goes on, I will revisit our progress in studio renovation to let you know how the project is going, some of the key limitations on the design and how it all turns out.

How to Avoid a Cluster of Failures

And What to Do When Everything Breaks at Once

BY CRIS ALEXANDER

In my company we have a first-rate bunch of market chief engineers.

Each is a veteran and has been in the business for many years. Each has tremendous experience with studio equipment, transmitters, antennas, towers, STL systems and the other components that make up the broadcast technical infrastructure. Some have a crew of engineers working under them. One has a large engineering staff. I am proud of our people and their technical abilities.

But even with all their experience, qualifications and staffing, each of these folks faces something that *every* broadcast engineer will confront on occasion: multiple fires to put out at the same time. We may wish things would always happen at a metered pace, but the reality is that when it rains, it pours.

It doesn't much matter why this happens. The problem becomes one of priorities: What do we fix first, and in what order do we address the rest?

There is no easy answer. Each situation is different and dynamic. And sadly, whatever you do, it often will be the *wrong* thing to do, from at least someone's perspective.

THE NIGHTMARE

Consider this situation: Severe thunderstorms march across the city, bumping the power at the studio and scrambling the brain on the automation, taking out Internet service and locking up the AM's on-air console.

When it snows, it pours. This FM antenna coincidentally failed at the same time as an audio processor and a whole-site UPS unit.

As the storms roll on, lightning hits the power lines at one of the FM sites, killing the power there and doing some other unknown damage — the generator is running but neither transmitter will come up. And the digital STL at the other FM site dies when the storm passes over.

To each station's PD, his or her station should have priority. They don't need PPM data to tell them that their listeners are all tuned to the competition now and that many just might not come back.

In recent years, I have observed competent, capable and committed engineers terminated because of situations very much like this one.

"But that's not fair!" you say; and you're right! Yet everyone needs someone to blame. In many cases, blame for poor performance in ratings and sales will be (unfairly) shifted to the chief engineer for any number of stated reasons. When you boil it all down, the problem often is poor communications. We can truthfully say, "Communication is our business — it's not our policy."

So how do we avoid this? In what order does our overloaded chief engineer address the myriad of problems and issues in this or a similar situation?

The process involves regular communication with a management-level decision maker, usually the GM or owner, the one person who has a dog in every fight in the cluster.

It really is up to him or her to help you set priorities, sometimes on the fly, and deal with the issues in the order that makes the best sense for the operation. This will be good for some of the people involved and bad for others, but that is unavoidable. But with the GM making or helping you make the calls, the blame is shifted to where the buck should always stop: at the top.

(continued on page 8)

FAILURES

(continued from page 7)

It is an excellent idea to sit down with the GM during a quiet period and talk about scenarios like this one. You know your market and your facilities. You know where the vulnerabilities are. The GM knows where the money is and what facilities need protecting, perhaps at the expense of another. You and he can flesh out a "what-if" list that you can use to prioritize things without a lot of additional consideration.

For example, most clusters have a billing leader, the one station in the cluster responsible for the lion's share of the revenue. It is a simple fact that the air time of that station is worth more than the air time of the other stations in the cluster. Accordingly, priority should always be afforded the lead station. The GM can identify the pecking order and help you set the respective stations' priorities.

DEVELOP A PRIORITY LIST

In other cases, the situation may be more dynamic.

Consider a cluster that has a billing leader that would normally get top priority, but during baseball season a sister station that would otherwise be at the bottom of the list might actually be assigned the top slot.

There can be all sorts of factors that figure into this, things to which the chief engineer would not normally be privy. For example, there may be contract negotiations in process for next season that could be jeopardized by an outage during a game. And because the situation is dynamic, regular communication with the GM is required to stay on top of it.

Some GMs are all over this. They're always thinking, always managing, always communicating. Others — and these are more typical — don't give engineering a thought until something goes seriously wrong. It's up to you to figure out which type you're dealing with and adjust the way you interact with them accordingly.

With the first type, you likely have a good and regular flow of communication already. With the second, the burden will be on you to initiate and maintain this flow.

Now we've considered the worst-case nightmare scenario, what about the everyday stuff? Like most engineers, you have your project list, your regular maintenance duties and a bunch of sleeve tugs in the hallway. Projects and maintenance are fairly easy to prioritize. Sleeve tugs easily can derail your priority train if you let them.

I would suggest the following as a starting point for the priority list for just about any station or cluster:

1. Compliance (FCC, FAA, etc.)
2. Transmitter Operations
    a. Transmitters (main and auxiliary)
    b. Antennas and Lines
    c. STLs
    d. Audio Processing
    e. Remote Controls
    f. Generators
3. Studio Operations
    a. Mixers
    b. Automation Systems
    c. Routers
    d. Source Equipment
4. Production
5. Remote Equipment
6. Facilities (building, etc.)

Identify vulnerabilities and don't even think about deferring maintenance on that aux transmitter!

Looking at this list, you probably can see some conflicts already, but that's unavoidable. Your priority list should be developed with the help of your GM or director of engineering, and it will in most cases follow the money.

Starting at the top, your licenses are critical to your operations and they must be protected at all costs. Stations simply can't afford trouble with the FCC or other regulatory agencies, especially trouble that can affect license renewal.

Your signals are the delivery mechanism for your product (programming) to the consumer (the listener). If you can't deliver, you're done. It won't matter a whit how great the programming is or how good the audio from the studio sounds. The transmitters main and aux, the antennas, transmission lines, STL links, processors and remote control systems have got to work all the time. And don't even think about deferring the maintenance on that aux transmitter. That's like neglecting to get the flat fixed on your spare tire — sooner or later you will be walking.

Much the same argument can be made for studio facilities; without programming, all you'll have is a dead carrier. But you typically have more options at the studio. Most automation systems, for example, can operate in some sort of emergency mode for a day or more. On-air consoles, mixers and routers can be bypassed altogether if necessary to get back on the air. But otherwise, studio facilities are high on the priority list.

Production is important, but you can be on the air without production for awhile, and again there are often other options (multiple production rooms, control rooms not in use, etc.).

You get the idea.

Once the priority list is developed, the trick is to stick with it. That's where the sleeve tugs come in. You're walking down the hall and one of the producers calls your name, telling you that Mic 2 in Production B isn't working. The temptation is to run in there and see about the mic problem, but you can't do that if there are other things in play that are higher on the priority list. You have to be disciplined here. You have to explain to all sleeve-tuggers that they must use the established discrepancy reporting system and write up the problem; you'll get to it as soon as you can. You can't allow the urgent to get in the way of the important.

DON'T FAIL TO COMMUNICATE

I hope that if you've gotten anything out of this discussion, it's that you must *communicate*. Good communication with your GM or owner, PDs, producers and others is key to a smooth-running, properly prioritized technical operation.

Let me add one more thing. Once a week, write a detailed report of your activities to the GM. Copy everyone else that might be interested — PDs, producers, etc. Describe in easy to understand terms what you have done this week, provide the status of each item (complete, parts on order, etc.), and give a brief overview of the plans for the next week.

This report will do more for you than anything else you say or do. Each person in the operation sees only his or her own part of the operation. It's easy for them to think that you give all your attention to station "B," or that "He spends all his time at the transmitter site goofing off." Your professional, well-written, detailed weekly report will put an end to all that, document your activities for the record and — here's the best part — help you evaluate your own performance and priorities. If they don't sound right when you write them up, they probably aren't.

Cris Alexander is director of engineering at Crawford Broadcasting Company, a longtime RW contributor and a past recipient of the SBE's Broadcast Engineer of the Year Award.

RoadWarrior LC is a new full-duplex, two channel (Program & talkback) audio codec. Its new design, robust, compact and with a flat control surface, prevents accidental damage to the controls and makes it easier to use. It is a portable audio codec with all Suprima functionality built in.

RoadWarrior LC

- **IP/ISDN**
- **2-channel input mixer with line/mic levels and phantom power**
- **Lightweight & rugged design**
- **Can be controlled remotely from its web page**

Rear panel of RoadWarrior LC

Suprima

- **Includes LAN, ISDN, U & ST, and X.21 interfaces Standard**
- **Auto backup to ISDN from IP or X21**
- **Built in Web Browser for control and monitor from remote location**
- **Comes fully loaded with every available algorithm included**

Visit us at NAB Booth #C11533

CCS™

MUSICAM USA™

670 North Beers Street, Bldg. #4
Holmdel, NJ 07733 USA
732-739- 5600
732-739-1818 fax
email: sales@musicamusa.com
web: www.musicamusa.com

FILTER

(continued from page 1)

tuned filters and a reject load. The directional filter in Fig. 1 has been used for years to combine multi-station FM and TV signals to one transmission line while providing adequate isolation between transmitters.

COMBINER OPERATION AND LIMITATIONS

A signal at the narrow-band input port is split in quadrature (the split outputs have a 90 degree phase difference), passes through the bandpass filters, and is recombined at the output hybrid due to the phase relationship of the signals at the hybrid. Out-of-band signals delivered to the broadband input port will split in quadrature at the output hybrid, reflect off the bandpass filters (the amount reflected is a function of filter rejection), and be delivered back to the output hybrid where they are recombined and delivered to the output terminal.

Isolation from broadband input to the narrow-band input port is a sum of filter rejection (typically 27–30 dB) and hybrid directionality (typically 36–45 dB). Isolation from the narrow-band input to the broadband input port is dependent on hybrid directionality, 36–45 dB, and proper phase matching of the bandpass filters. The transmission response from each input to the output is shown in Fig. 1 and color-coded. Note the transmission response for each input is similar to the S-parameters of the bandpass filter (i.e. the narrow input is the filter's transmission response, S21, while the broadband input is the filter's reflected response, S11 or S22). This makes analysis of the directional filter simple.

There is nothing new about the directional filter layout in Fig. 1 other than the type of filtering provided to perform the combining function of analog and digital signals for IBOC transmission. Illustrated in Fig. 2 is the typical frequency response (S21 and S11) of a three-section filter used for combining co-located FM channels with 1.6 MHz separation and the same filter with 800 kHz separation. Remember, the filter transmission response, S21 in red, represents the narrow-band input to output frequency response of the directional filter in Fig. 1. The filter return loss, S11 in blue, represents the frequency response of the broadband input to output. The limited rejection for 800 kHz spacing (approx. 7.5 dB at 600 kHz from filter center frequency, i.e. the lower band edge of channel to be combined) results in higher loss (approx. 1.2 dB at the lower band edge of channel to be combined) at the broadband input.

Module insertion loss from the broadband input port to the output port is plotted as a function of filter rejection in Fig. 3, and illustrates the need for high rejection to obtain low insertion loss from the broadband input to output port.

This is way too much loss for high-power FM combining. Transmitters don't typically have the headroom to waste this much power; filters would run exceptionally hot and the low efficiency results in high operating costs. As a result, a four-section filter (or three sections with cross-coupling) that provides more rejection is required for 800 kHz channel separation.

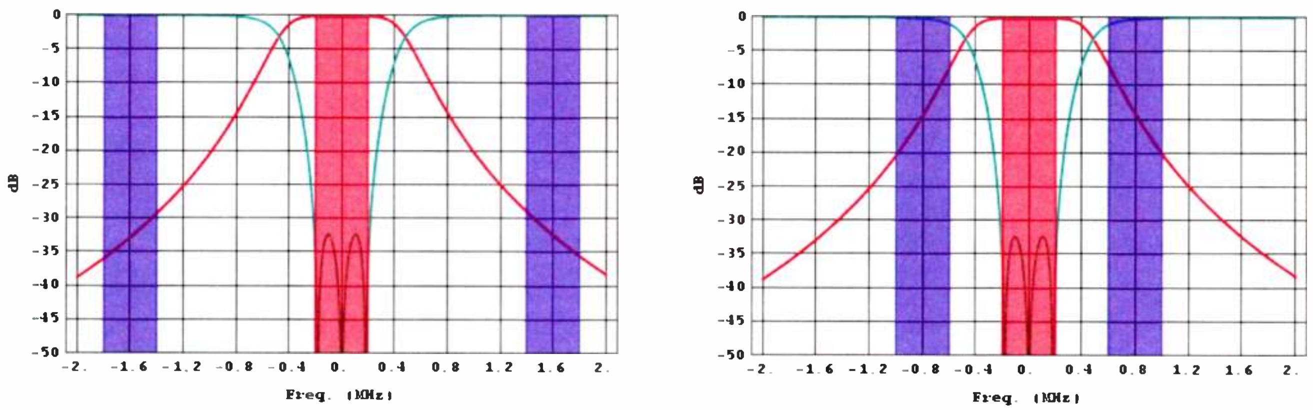


Fig. 2: Three-Section Filter Response, S21 and S11, 1.6 MHz and 800 kHz Channel Separation

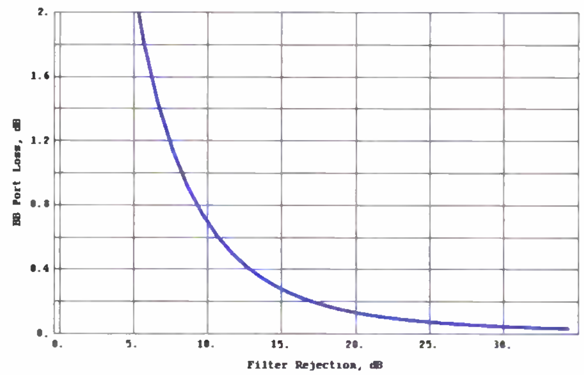
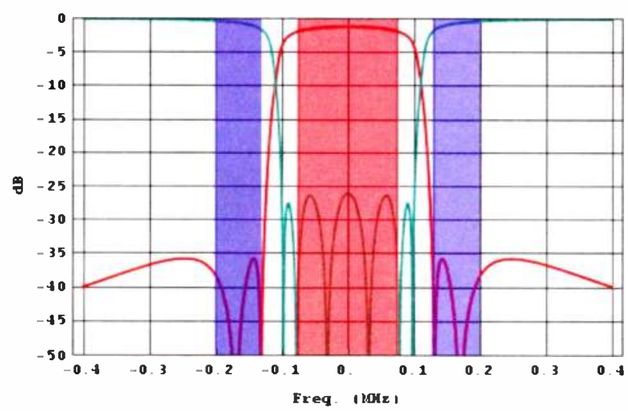


Fig. 3: Relationship Between Filter Rejection and Broadband Port Insertion Loss



IBOC LOW LOSS COMBINER MODULE LOSSES, 6-SECTION SHARP TUNED FILTER				
Mode	**Min. Loss (dB)**	**Max. Loss (dB)**	**Variation (dB)**	**Integrated Loss (dB)**
FM	-1.13	-1.8	0.67	-1.31
Digital, MP1	-0.34	-1.67	1.33	-0.71
Digital, MP11	-0.34	-28.69	28.35	-2.04

Fig. 4: Sharp-Tuned Pseudo-Elliptic Function Filter Response for Combining Digital Sidebands to Host FM

UNSUITABLE FOR IBOC?

The use of filters to combine FM and digital sidebands was discussed early in the rollout of HD Radio, but quickly ignored due to the filter order required for 27–30 dB rejection at the digital sidebands, its related cost, size and resulting FM loss. Referring to Fig. 1, the analog or FM input is applied to the narrow input, and the digital signal applied to the broadband input. A six-section sharp-tuned filter with multiple cross-couplings can achieve the required rejection at the IBOC MP1 digital sidebands. Six-section pseudo-elliptic function filters that incorporate multiple cross couplings are not difficult to design; however, significant losses are incurred due to the narrow bandwidth and filter order (compared to "classic" tuned FM filters).

Higher losses lower the power handling capabilities for a given size filter. A six-section, pseudo-elliptic function, sharp-tuned filter response and table summarizing loss results is illustrated in Fig. 4. The shaded red represents the host FM channel +/– 75 kHz while shaded blue represents the MP1 mode digital sidebands. Integrated or average loss of this narrow sharp-tuned filter using high Q 24 inch square cavities that provides 30 dB rejection at the digital sidebands is approximately 1.31 dB. Note however, that due to the finite cavity Q there is also significant loss roll-off at the digital sidebands inner edge. Using this size cavity in a classic four-section filter with wider bandwidth (400 kHz plus) would have approximately 0.25 dB loss and handle 30 kW. The sharp-tuned filter in Fig. 4 using same cavity geometry can only handle approximately 7 kW without a lot of additional cooling.

It's easy to see why sharp-tuned filters were quickly discarded as a solution to combine FM and digital sidebands.

INTENTIONAL LOSSES

A directional filter combining solution that *purposely* exhibits loss was overlooked. Reducing filter order and requiring less rejection at the digital

(continued on page 12)

MINSTRUMENT MATRIX

Sophisticated Minstruments from NTI give you comprehensive test capability... and these flexible audio instruments fit in the palm of your hand

DL1 Digilyzer
Digital Audio Analyzer

A handheld digital audio analyzer with the measurement power & functions of more expensive instruments, the DL1 Digilyzer analyzes and measures both the digital carrier signal (AES/EBU, SPDIF or ADAT) as well as embedded digital audio. In addition, the DL1 functions as a smart monitor and digital level 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 connect to an analog line.
In addition to customary audio, carrier and status bit measurements, the DL1 also includes a comprehensive event logging capability.

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

DR2 Digirator
Digital Audio Generator

The DR2 Digirator not only generates digital audio in stereo & surround, it is a channel transparency and delay tester as well, all condensed into a handheld package. Delivering performance & functionality challenging any digital audio generator made today, it produces all common audio test signals with sampling frequencies up to 192 kHz and resolution up to 24 bit. The Digirator features a multi-format sync-input allowing the instrument to be synchronized to video and audio signals. In addition to standard two-channel digital audio, the DR2 can source a comprehensive set of surround signals.

- AES3, SPDIF, TOSLink, ADAT outputs
- 24 bit 2 channel digital audio up to 192 kHz SR
- Sine wave with stepped & continuous sweeps; White & Pink Noise; Polarity & Delay test signals
- Dolby D, D+, E, Pro-Logic II, DTS and DTS-HR surround signals
- Channel Transparency measurement
- I/O Delay Measurement
- Sync to AES3, DARS, word clock & video black burst
- User-generated test signal files

AL1 Acoustilyzer
Acoustics, Audio & Intelligibility Analyzer

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

- Real Time Analyzer
- Reverb Time (RT60)
- Delay measurements
- High resolution FFT with zoom
- Optional STI-PA Speech Intelligibility function
- Automatic Distortion analyzer (THD+N)
- Frequency, RMS Level, Polarity measurements
- Requires optional MiniSPL microphone
- Includes MiniLINK USB interface & Windows PC software for storing tests and PC transfer

MR-PRO Minirator
High performance
Analog Audio Generator +
Impedance/Phantom/Cable measurements

The MR-PRO Minirator is the senior partner to the MR2 below, with added features and higher performance. Both generators feature an ergonomic instrument package & operation, balanced and unbalanced outputs, and a full range of signals.

- High (+18 dBu) output level & <-96 dB residual THD
- Sine waves & programmable swept (chirp) and stepped sweeps
- Pink & white noise
- Polarity & delay test signals
- User-generated custom test signals & generator setups
- Impedance measurement of the connected device
- Phantom power voltage measurement
- Cable tester and signal balance measurement
- Protective shock jacket

ML1 Minilyzer
Analog Audio Analyzer

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

Add the optional MiniLINK USB computer interface and Windows-based software and you may store all tests on the instrument for download to your PC, as well as send commands and display real time results to and from the analyzer.


MiniLINK

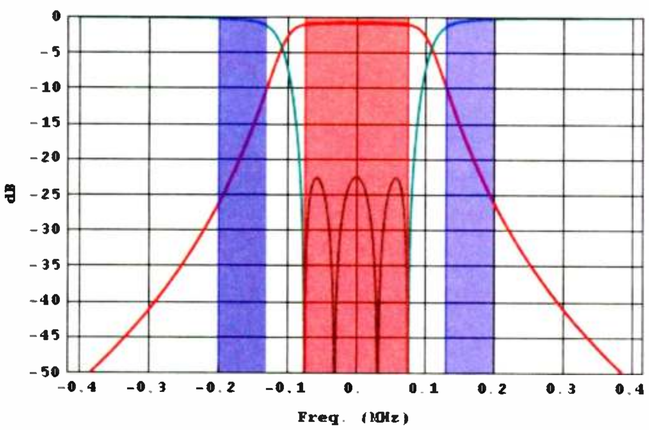
- Measure Level, Frequency, Polarity
- Automatic THD+N and individual harmonic distortion measurements k2 – k5
- VU + PPM meter/monitor
- 1/3 octave analyzer
- Requires optional MiniSPL microphone for SPL & acoustic RTA measurements
- Frequency/time sweeps
- Scope mode
- Measure signal balance error
- Selectable units for level measurements

MR2 Minirator
Analog Audio Generator

The MR2 pocket-sized analog audio generator is the successor to the legendary MR1 Minirator.
It is the behind-the-scenes star of thousands of live performances, recordings and remote feeds.

- Intuitive operation via thumbwheel and "short-cut" buttons
- New higher output level (+8 dBu) & low distortion
- Programmable Swept (chirp) and Stepped sweeps
- Sine waves
- Pink & White noise
- Polarity & Delay test signals
- Illuminated Mute button

NTI Americas Inc
PO Box 231027
Tigard, Oregon 97281 USA
503-684-7050
www.minstruments.com
info@ntiam.com



IBOC LOW LOSS COMBINER MODULE LOSSES, 4-SECTION MILD TUNED FILTER				
Mode	Min. Loss (dB)	Max. Loss (dB)	Variation (dB)	Integrated Loss (dB)
FM	-0.806	-1.05	0.244	-0.87
Digital, MP1	-0.19	-1.29	1.1	-0.48
Digital, MP11	-0.19	-7.31	7.12	-1.23

Combining Method	Efficiency		
	FM	Digital, MP1	Digital, MP11
IBOC LOW LOSS	83%	90%	75%
-10dB Injector	90%	10%	10%
-6dB Mid-Level Injector	99%	25%	25%

Fig. 5: 'Mild' Tuned Four-Section Filter Response for Combining Digital Sidebands to FM

FILTER

(continued from page 10)

sidebands has less impact on the host FM (lower loss, lower delay and loss variation), and the digital loss incurred is significantly less than current high- and mid-level combining solutions. Fig. 5 illustrates one solution using a four-pole filter with a Chebyshev response. The shaded red represents the host FM channel +/– 75 kHz while shaded blue represents MP1 mode digital sidebands. Note the integrated loss and variations over FM (+/– 75 kHz) are significantly less than the sharp-tuned filter.

The loss numbers in Fig. 5 are calculated using the unloaded Qs of 24 inch square cavities. The higher loss as compared to a "classic" tuned four-section filter is due to the narrow bandwidth. For example, the bandwidth of the filter response in Fig. 5 is approximately 160 kHz, compared to 400 kHz minimum used in "classic" FM combining applications. Fig. 6 illustrates the effect of bandwidth on losses for a four-pole filter. Note a 400 kHz wide four-pole filter used in a "classic" combiner would have approximately 0.25 dB mid-band insertion loss.

Even with the reduced integrated loss (0.87 dB) the bandpass filters will still require significant cooling apparatus. Three section filters with cross-coupling provide a more optimum solution when considering loss and degradation to the host FM with only slightly higher integrated loss on the digital sidebands. Instead of bandpass filters, notch filters could also be used in the complementary circuit.

The directional combining solution that *purposely* exhibits loss does not require lossy and unreliable circulators for isolation. However, isolation is a function of filter rejection as described above, and is reduced compared to the isolations achieved in "classic" FM combining modules due to the reduced requirement for digital combining. Classic multi-station FM combining modules will have approximately 30 dB of rejection at the channel to be combined, and therefore, provide approximately 65 dB to 70 dB isolation when hybrid directionality is added. IBOC combining modules provide 40 dB–45 dB isolation between all ports, significantly less due to lower filter rejection.

As presented at the 2009 NAB Show in Las Vegas, a four-pole combining module was tested using a Harris HPX Transmitter with a FlexStar HDx Exciter operating in analog mode at 24.1 kW applied to the narrow input of Fig. 1. A Harris Z16HD Transmitter using a FlexStar HDx Exciter at 2.7 kW was applied to the digital input, or broadband port of Fig. 1. The combined output of 20 kW FM with 2 kW (–10 dBc) digital sidebands is shown in Fig. 7, MP3 mode of operation.

There was moderate degradation to the FM signal without linear pre-correction. The uncorrected degradation to the FM signal was not severe enough to prevent this combining solution; however, linear correction can further reduce degradation. The 1.4 dB roll-off slope impressed by the filter on the inner MP1 (or the 7 dB slope impressed by the filter on the inner MP3) HD Radio sidebands had no impact on uncorrected Block Error Rate, introduced no change in the CD/No of 72.5 dB, and the signal was fully receivable on an external consumer receiver in both MP1 and MP3 modes. Combiner isolation was more than adequate to suppress RF IMD products to meet NRSC-5B emission mask requirements (see Fig. 7) without RF circulators or isolators.

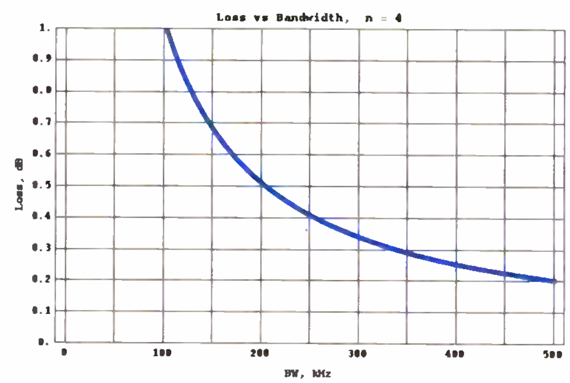


Fig. 6: Mid-Band Filter Loss vs. Filter Bandwidth

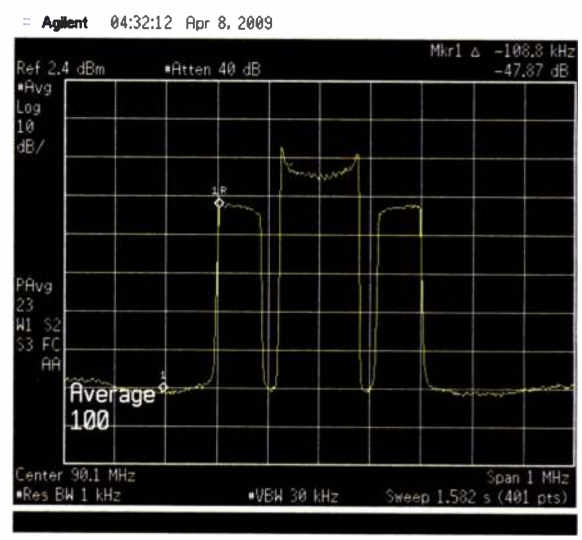


Fig. 7: Combined Output, 20 kW Analog, 2 kW Digital

The combining system is ideal for the early adopter wanting to increase digital power up to –10 dBc without the need to invest in another transmitter.

The impact on HD Radio Modulation Error Ratio (MER), a new standardized method for determining transmission quality, needs further investigation. Myat is working with major radio groups and transmitter manufacturers to clarify complete proof of performance.

Although narrow sharp-tuned filters were initially discussed for IBOC combining, high cost, large size and excessive loss and loss variations prohibit their use. The combining system described here that uses wider, mild-tuned filters has less impact on the host FM and has significantly better digital efficiencies (see Fig. 5 for efficiency comparison) than other high-level combining techniques. The combining system is ideal for the early adopter wanting to increase digital power up to –10 dBc without the need to invest in another transmitter. The combiner also has application for higher-power stations (approximately 20 kW TPO and above) where common amplification is not practical because of the excessive peak voltage in the transmitter.

Comment on this or any other story at rwee@nbmedia.com.

Often Used But Often Misunderstood

Exploring the Basic Operation of Compressors

BY DAVE MOULTON

I thought it might be nice to take some time to consider the primary tool we use to cope with audio levels. That would be, of course, the audio compressor.

This is a fascinating device. We use it a lot. However, our understanding of it is a little, ah, limited.

The compressor has some fairly tricky controls. Its action can be hard to hear if it's used well. Mostly, it isn't understood all that well by those of us who use it. It has the further characteristic of coming in a range of shapes, sizes and flavors, sometimes described in rather exotic terms that are often hard to understand.

It often is cloaked in retro mysticism (as in "Oh man, that old tube [insert brand name here] compressor does better on everything than any modern compressor. They should never have stopped making it!"). Sometimes, it is an extremely complex device (as in multiband compressors) and sometimes it is pretty tricky as well (as in "look-ahead" compressors).

SO WHAT DOES A COMPRESSOR DO?

A compressor is a device that regulates the gain or level of an audio signal as a function of (usually) the changes in amplitude of that signal, according to a fairly complex set of rules.

Sometimes we use it to prevent overly loud signal peaks from distorting, and sometimes to smooth out the level variations to make a signal (particularly a voice signal) more continuously and easily audible. Sometimes we use it to reduce the overall dynamic range of an audio signal.

All compressors work by sending the incoming audio signal through an active gain stage, usually a voltage-controlled amplifier or its digital equivalent. At the same time, the signal is also sent, in parallel, to a so-called level detector, which studies the signal and converts it into a control voltage, or its digital equivalent.

Said control voltage is manipulated in a variety of clever ways (here's where each compressor gets "its own sound") and then used to regulate the gain of the active gain stage. What could be simpler?

For instance, if the control voltage is inverted — so that as the amplitude of the audio signal gets greater, the control voltage is reduced — the net result will be that as the audio signal gets louder, the gain stage will make it softer, hence "compressing" its dynamic range. Got it? Good.

If only it were that simple …

In actual fact, we want the compressor to do a bunch of other things for us.

First let's consider the Threshold control, which is probably the most important control on any given compressor.

WHAT DOES THE THRESHOLD LEVEL CONTROL DO?

Over the years we've learned that what we really want most is for the compressor to leave the level of the signal alone, except for various embarrassing peak levels that are causing overloads and distortion. Hence, we came up with a control called the "Threshold."

This Threshold control sets the level above which there will be acts of compression occurring. Below that threshold level, there will be no changes in the level of the signal due to the compressor. Got that?

In an analog compressor, the threshold level is usually expressed in dBu (or dBm), and calibrated from something like +20 dBu down to –20 dBu. In the digital realm, threshold level is given in dBFS, from 0 dBFS down to –40 or even –60 dBFS.

Let's study the implications of this for a moment.

What Happens With a High Threshold Level?

Let's set the Threshold control in a digital compressor at –6 dBFS for a moment, to discuss what would happen. When the signal level is at or below –6 dBFS no compression of any sort happens. What comes in goes out unchanged.

However, when the signal level goes above the threshold set at –6 dBFS, compression occurs, and the compressor reduces the level of the signal by some amount (determined by another control called Ratio, which I'll talk about later). This means that as signals approach 0 dBFS they are turned down in level, so hopefully they won't distort.

With this setting, then, very little would happen except when peak signal levels exceeded –6 dBFS, in which case they would be turned down to avoid overloads. Make sense?

What Happens With a Low Threshold Level?

However, if we set the threshold at, say, –20 dBFS, a lot more compression will happen.

First, everything above –20 dBFS (which is probably almost the entire signal trace) will be turned down by the amount determined by the ratio control. So virtually all the audio will be compressed in range, which is quite definite-

(continued on page 16)

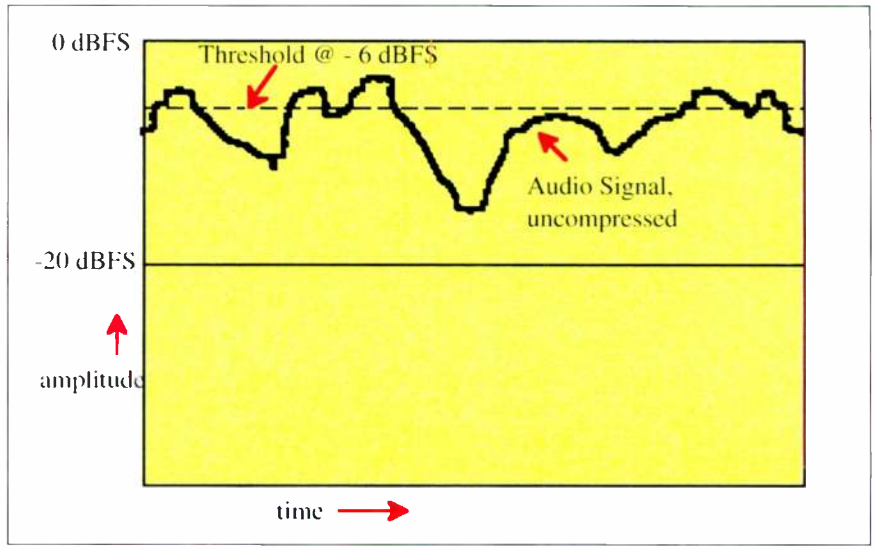


Fig. 1: A hypothetical uncompressed audio signal ranging between approximately –3 dBFS and –16 dBFS

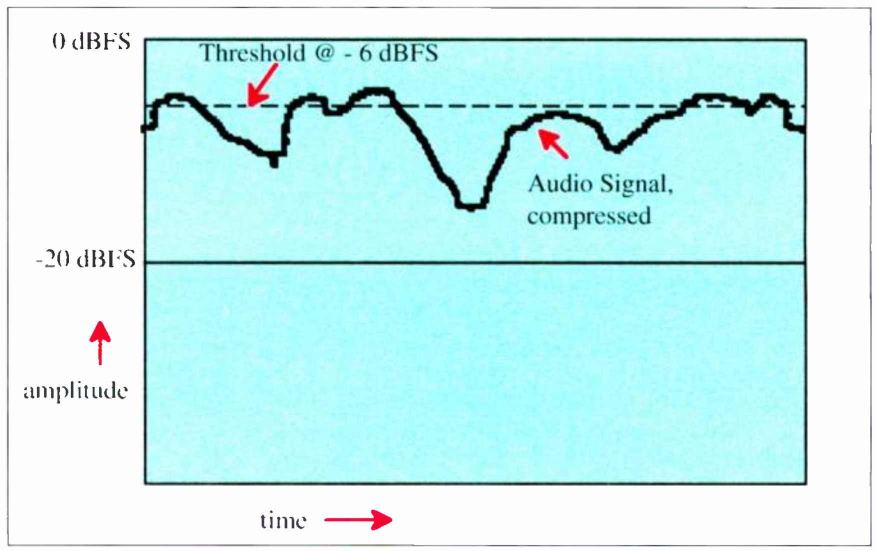


Fig. 2: This is the same audio signal in Fig. 1, now compressed above the –6 dBFS threshold.

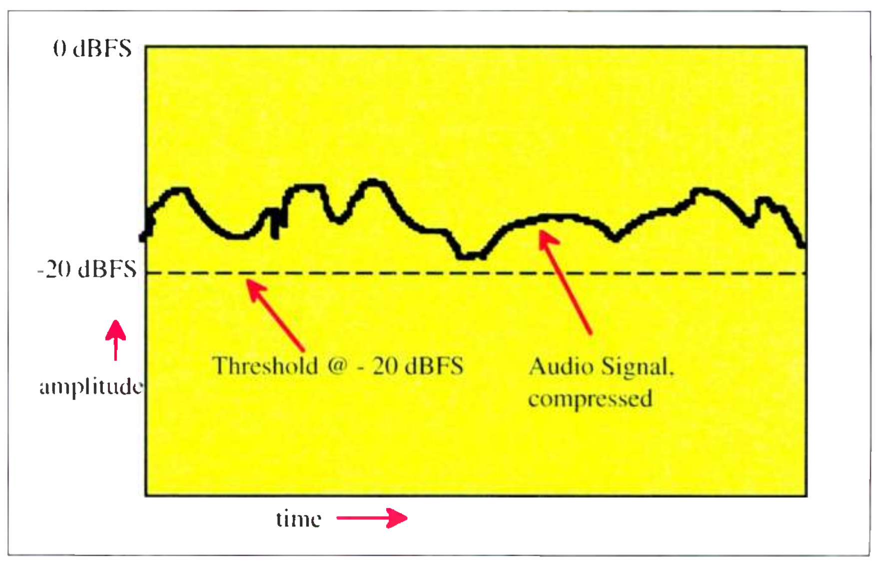
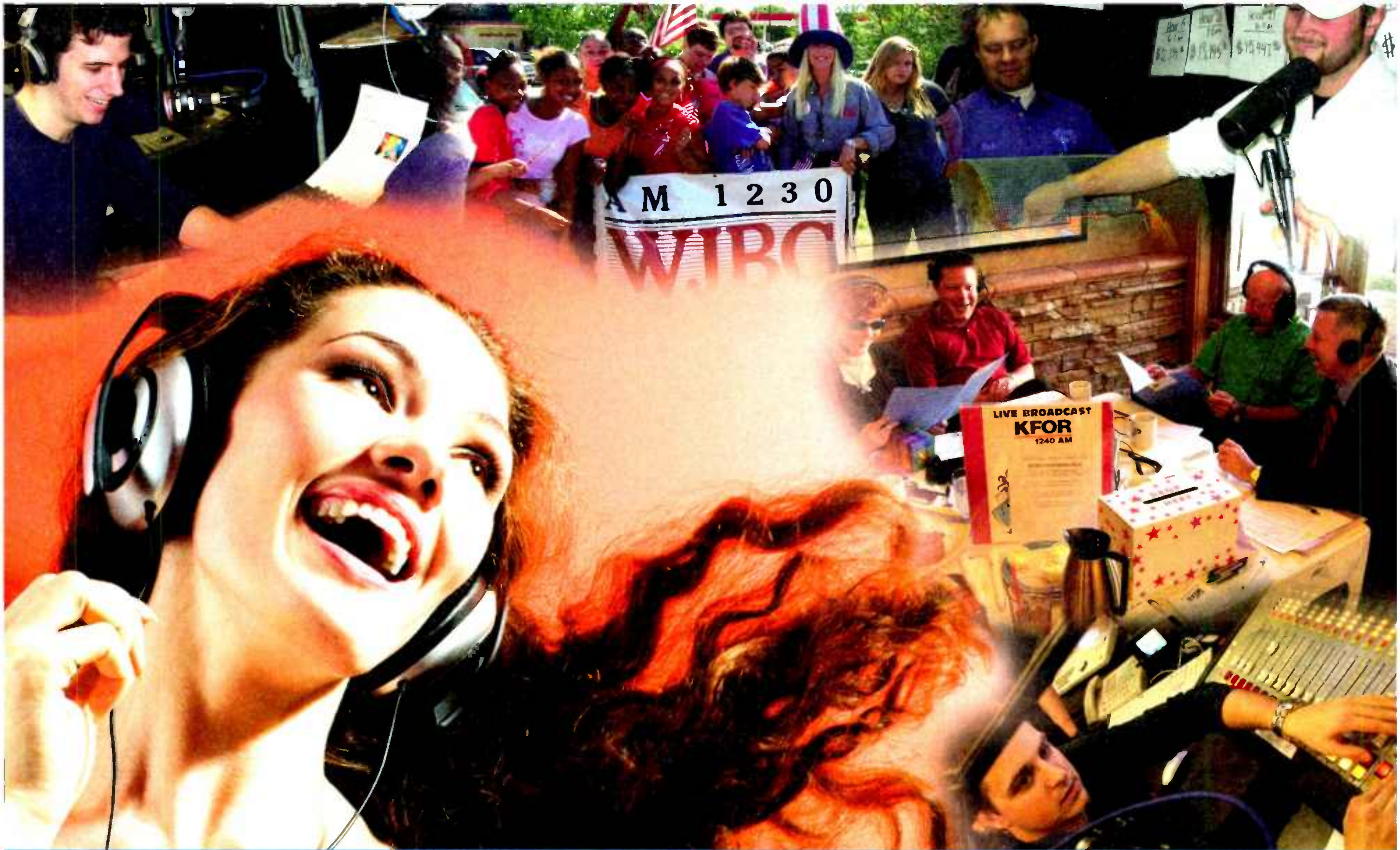


Fig. 3: This is the same audio signal as in Figs. 1 & 2, but with a –20 dBFS threshold.



HERE TODAY. HEAR TOMORROW.

Your passion for creating great radio got you where you are today. Now it's time to take advantage of the techniques and technologies at the NAB Show™ to ensure your programming will be heard everywhere your listeners are tomorrow. This is the ultimate venue for exchanging strategies and identifying sound solutions at the global level. And the smartest way to guarantee the programming you create reaches more consumers — at home, at work and at play.

At the NAB Show you'll find virtually every innovation driving production, editing, programming, transmission, ad sales and new media solutions for your thriving radio enterprise. Visit the unparalleled exhibit floor to see, touch and test advanced technologies that enable HD, live audio streaming, video archiving/streaming, revenue-generating Web sites and more. Then experience a wealth of hands-on educational opportunities presented by top talent, producers, advertising managers, PDs and Web developers, the stars influencing today's best radio content.

Join professionals who share your passion for radio excellence at the NAB Show this year. For more information, visit **www.nabshow.com.**

FREE
EXHIBITS-ONLY
PASS
USE CODE A520

NABSHOW™

Where Content Comes to Life™

Conferences April 10–15, 2010 :: Exhibits April 12–15

Las Vegas Convention Center :: Las Vegas, Nevada USA

COMPRESSORS

(continued from page 14)

ly audible in most cases. In crude or excessive usages this is referred to as "squashing."

So, depending on how the threshold is set, a lot or a little can happen to the signal, from nearly inaudible to extremely audible.

Take a look at Figs. 1–3.

Fig. 1 shows a hypothetical uncompressed audio signal ranging between approximately –3 dBFS and –16 dBFS. The threshold is set at –6 dBFS, but is not active.

Fig. 2 shows the same audio signal, now compressed above the –6 dBFS threshold. Note that there is very little change, except for the peaks; what was –3 dBFS is now –5 dBFS. Very subtle, but it means you could turn up the level by 5 dB without trouble if you wanted.

Fig. 3 shows the same signal, but with a –20 dBFS threshold. Now we've done some serious squashing. What was formerly –3 dBFS is now –12 dBFS and what was formerly –16 dBFS is now –18 dBFS. The dynamic range of the signal was originally 13 dB. It is now 6 dB; dramatically different. If we wished, we could turn it up by 11 dB without overload. It might not sound so good, but we could do it.

Gain Reduction

In a conventional compressor, the only time that gain reduction happens is when the audio signal is above whatever threshold level we've set. When the threshold is set at a high level, little gain reduction happens. When the threshold is set at a low level, a *lot* of gain reduction happens.

In the latter case, the overall level gets turned down quite a bit. In my previous example, it had been turned down by 9 dB, which is a lot. Take a look at Fig. 4, which shows a composite of Figs. 1 and 3 from above. It shows the heavily limited result of a low threshold applied to a typical fairly loud audio signal. We've limited the gain all right, but we've also made the signal much softer.

That leads us to the use of two other controls on a compressor, Make-Up Gain and Ratio.

MAKE-UP GAIN

Often, our goal in using compression is *not* to make the signal softer, but to make the signal stay fairly close to some particular level. So, at the output of the compressor, we put in another gain stage with a level control, which allows us to turn the signal that we've squashed back up in level (see Fig. 5).

Fig. 5 shows several things. First, the squashed signal has been turned up so that now it hovers just below 0 dBFS. This was done by adding 11 dB of make-up gain in the compressor. As a function of doing that, of course, we have also turned up the noise floor of the audio signal by the same amount: 11 dB. This is a polite way of saying that we have reduced the dynamic range of the signal by 11 dB.

So, the benefit of this sort of compression is that we've made the signal level much more stable, and we've made it all pretty loud (and audible). The downside is that we've increased the level of noise, which could be (and often is) annoying. It comes with the territory, as Custer used to say.

The point of make-up gain is to compensate for the amount of gain reduction we've done while squashing the signal, with the negative side effects of increased noise and a certain lack of dynamic expressivity.

SO WHAT IS RATIO?

There is another way to approach this, which is to control the *amount* of gain reduction that occurs above the threshold level. This is done with the control called Ratio.

The downside is that we've increased the level of noise, which could be (and often is) annoying. It comes with the territory, as Custer used to say.

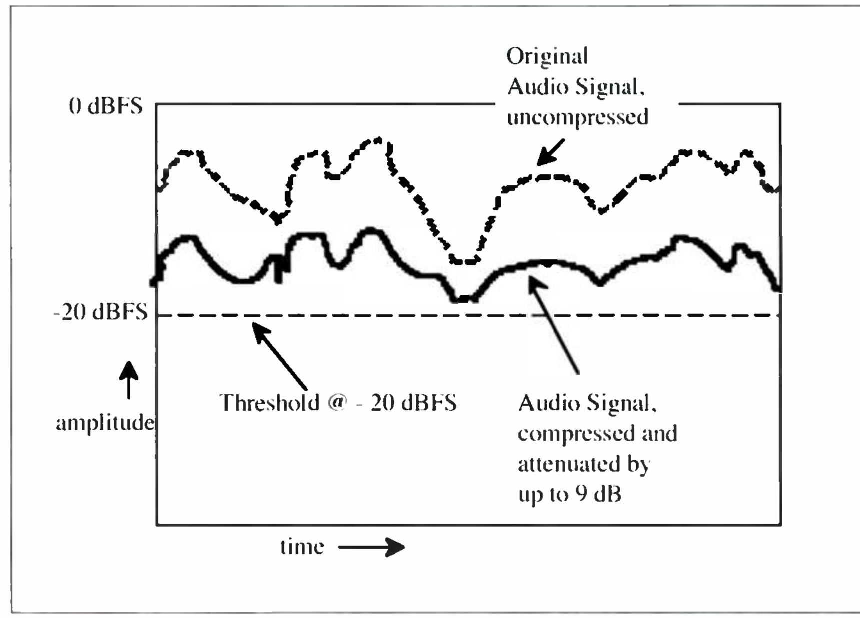


Fig. 4: The heavily limited (squashed?) result of a low threshold applied to a typical fairly loud audio signal. We've limited the gain all right, but we've also made the signal much softer.

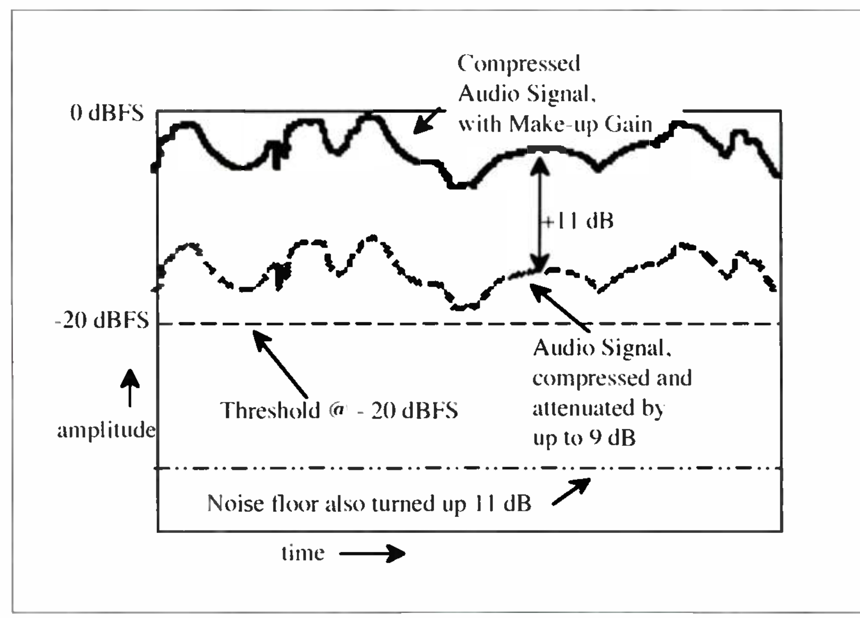


Fig. 5: This figure shows several things. First, the squashed signal has been turned up so that now it hovers just below 0 dBFS. This was done by adding 11 dB of make-up gain in the compressor. As a function of doing that we have also turned up the noise floor of the audio signal by the same amount: 11 dB. This is a polite way of saying that we have reduced the dynamic range of the signal by 11 dB.

The Ratio setting expresses the amount of input amplitude above the threshold that is needed to yield 1 dB *output* above the threshold. For instance, assume the threshold is at –10 dBFS and the Ratio control is set at 3. This means that an input of –7 dBFS (3 dB over threshold) will result in an output of –9 dBFS (1 dB over threshold). If the input goes up to –4 dBFS (6 dB over threshold), the output will be –8 dBFS (2 dB over threshold). Got it?

Suppose the ratio is set at 20, with the threshold still at –10 dBFS. An input of 0 dBFS will result in an output of –9.5 dBFS (0.5 dB over threshold). Usually, any ratio setting greater than 10 is thought of as limiting.

For the illustrations above, I used a ratio of 3.

For solo acoustical musical tracks like vocals, very gentle ratios (maybe 1.5 or 2) are most appropriate. Sometimes though, with a high threshold setting, you'll want a ratio that is also high, particularly if a lot of the program is above threshold, to keep all the overshoots from distorting.

Threshold and Ratio are often used together to manage the dynamics of a given track or mix. You work on the threshold level to get it to a point where it is at or slightly below all of the troublesome levels on the track, while tweaking the ratio to get just the amount of gain reduction you want for effect. In plug-in compressors, you can even automate these two functions, to very gently massage the track as it goes along, variably compressing what needs some help while leaving the rest of the track pristine.

In a compound compressor, there may even be both a limiter function (at a high threshold level) and a gentler compressor function at a lower level. I find this handy to protect myself against overloads while getting some really nice smooth gain management over the meaty musical content of the middle levels of the dynamic range (say, from –8 dBFS down to –25 dBFS).

ATTACK AND RELEASE

All compressors work by sending the incoming audio signal through an active gain stage, usually a voltage-controlled amplifier (VCA) or its digital equivalent. At the same time, the signal is also sent (in parallel) to a so-called level detector, which studies the signal and converts its amplitude into a control voltage (or digital equivalent). This, in turn, is used to regulate the gain of the signal via the VCA. That level detector and the resulting control voltage have some pretty tricky aspects, having to do with time.

Compressor designers usually include two special time controls called Attack (to control how rapidly the compressor 'attacks' the level-over-threshold to turn it down), and Release (to control how rapidly the compressor stops reducing the gain after the level-over-threshold has gone away).

Three things often happen, and you need to know about them. The first one is a distortion-like sound that can be created by the control voltage being changed too quickly. It's called amplitude modulation. The second one is a change in spectrum, due to the attack and release times emphasizing one portion of the spectrum of the program, or attenuating another portion. The third one is called "pumping," a gasping quality that relates to the level being returned to its uncompressed state too quickly or too often.

All of these have to do with time. In most cases compressor designers have included two special time controls called Attack (to control how rapidly the compressor "attacks" the level-over-threshold to turn it down) and Release (to control how rapidly the compressor stops reducing the gain after the level-over-threshold has gone away).

Amplitude Modulation

You'd think, intuitively, that we'd like the compressor to track the "level" of the signal just as quickly as possible, right? That way, we could compress just the offending elements and nothing else. It'd be the most accurate, no?

Unfortunately, the level of the signal is actually the wave trace itself. If we track it very closely, the control voltage will become an audio signal itself (because it goes up and down within the audio range), and cause the voltage-controlled amplifier to also generate an audio signal. This will modulate the actual audio in the low-frequency realm. It sounds just like fairly nasty, low-frequency harmonic distortion.

So, we use two strategies to head off this nasty. Set either the attack or the release control slow enough so the compressor can never modulate in the audio realm. "Slow enough" means a time *greater than* 50 ms.

Just so you know, 50 ms is the period for a 20 Hz tone, and if we keep things longer than 50 ms, they'll never get into the audio range above 20 Hz. Clever, eh?

If we want a fast attack to catch some sudden spikes of energy, we need to make sure the release is a good bit slower than 50 ms. If we want a fairly fast release, we may want to set the attack slower than 50 ms. As always, *use your ears!*

Changes in Audio Spectrum

An unintended consequence of compressor use can be an alteration of the balance of the audio spectrum. This happens as a function of the specific program material, and is more of a problem with music than voice.

Often, for example, we'll have extremely loud bass or kick drum signals. They can be (and often are) the loudest component in the program. They can trigger the compressor's gain reduction, turning down the bass, but then (as the release occurs) let the higher frequencies through with little or no attenuation. Occasionally, the behavior will be just the opposite, depending on the settings and the program.

The problems also occur as a function of the attack time and/or the release time changing the envelope (shape of loudness) of individual sounds, and thereby changing their timbre. Solving these problems takes some practice and careful listening to the effect the attack time is having, and then the effect the release time is having.

And remember, it varies with the program material. Heavy metal will suffer differently than will a nice female jazz ballad. But they both will suffer from compression abuse.

Pumping

Pumping is a real problem in situations where we have both a voice and background noise. We feel the need to limit the level of the voice, to make it consistent and distortion-free. So we compress with a fairly low threshold and large ratio. Also, we use a fast attack, so that the beginning consonants of words don't spit at the listener.

If we use a slow release, once the compressor is invoked, the level stays down, maybe too far. So we speed up the release time, and as a result, during the spaces between words or sentences, the release of the compressor pulls the background noise back up, maybe by 15–20 dB, creating a kind of a sudden sucking "sheeeuppp!" leading into the next word spoken, which then punches the level back down. When this happens over and over, it is fatiguing and annoying.

The answer here is judicious balancing of attack time and threshold together, against the release time, so that the attack of the compressor is fairly mild and not overdone, while the release is slow enough that the background noise just begins to start pulling up before the next word (and attack) arrive. Sometimes just reducing the total amount of gain reduction (via threshold adjustment) can solve the problem best.

Taken together, the effects of attack and release adjustments suggest a little more about the complexity of compression. Much of what we need to do is more concerned with time than level.

Compressors go back and forth between the realm where the level change can occur to each individual sound (fast attack and release, which changes the envelope of each sound, a key determinant of timbre) and the realm where we are changing the overall level (which is

(continued on page 18)

COMPRESSORS

(continued from page 17)

what happens with a slow release).

Getting so you can hear the difference between compression as envelope and compression as level control is a big step in developing your hearing skills.

To conclude this review of compressor basics, let's consider the nature of the level detector itself, and how it affects the sound quality of the compressor.

THE NATURE OF A LEVEL DETECTOR

The level detector is a circuit that observes the incoming audio signal and derives from it a DC control voltage that will, after manipulation by the Ratio, Make-up Gain, Attack and Release controls, regulate the gain of the voltage-controlled amplifier that is at the heart of the compressor. Exactly how that detector derives such a DC control voltage is an important determinant of how the compressor will regulate gain and how it will sound.

At the heart of such concerns are the concepts of peak, average and RMS levels and their detection. These days, digital compressors often give us a choice between these modalities.

Peak detection derives a control voltage from the highest signal peaks encountered. The detector senses the highest audio level and determines the control voltage from that, usually slowly decaying until another peak is encountered. Such detection concentrates on preventing overloads or clipping, by emphasizing quickly the maximum signal levels encountered.

Average detection is derived from an average of the changing signal level over some time period, usually between 100 ms and 3 seconds. Such detection of amplitude is fairly slow (depending on the signal itself) and gain regulation is comparatively restrained.

RMS (Root Mean Square) detection is derived from the relative power variations of the signal. Power changes as the square of the amplitude, so RMS detection is more volatile over time for a given time constant compared to average detection. It is fairly well correlated to how we humans hear, but it is also fairly complex to compute in real time, so it is not used as much as it might be, particularly in low-cost compressors.

Loudness is a subjective sensation, not a physical value. Our perception of loudness varies as a function of amplitude, frequency, frequency bandwidth and time. It is a complex sensation. It defies direct physical measurement.

With compressors we are attempting to manipulate relative loudness and the behavior of the level detector can have a significant effect on such perception of loudness. In the case of peak detection, a squashing of loudness range occurs for any program with regularly occurring peaks, while for programs with only one peak there is little reduction in loudness except for the few moments following that single peak.

Rupert Neve Designs Portico Series 5043 Compressor/Limiter Duo

What should be clear from this is that peak detection will have a distinct set of behaviors and effect on loudness, which may not be appropriate, but will be distinctive. At the same time, such peak detection does head off overloads and clipping pretty definitively.

In the case of both average and RMS detection systems, we are actually taking an average of the level. Often, we integrate that average over time, so that recent events have a greater effect on gain regulation than do not-so-recent events. But, in any case, such detection is of an average level, *not* a maximum level. This means overshoots remain possible (governed, of course, by the Attack and Release controls).

The difference between average and RMS is that average values respond to relative amplitude while RMS values respond to the relative power of those amplitudes. That power curve is exponential, so we obtain a different range and nature of behaviors.

Which is the correct one? Neither. They sound different — you gotta use your ears to decide what "sounds best" for you in any given application. Dang!

THE CREST FACTOR

One final complication to all of this is the crest factor, which describes the variable relationship between peak and RMS values.

Compression and limiting both may change the crest factor of a signal, depending on how their time constants are set. At the same time, crest factor is often surprisingly large. For random noise the crest factor is slightly less than 12 dB, meaning that the measured peak level of a pink noise signal will be almost 12 dB greater than the RMS measured level. For some voice signals or certain types of percussive music, a crest factor of 20 dB is reasonable to expect.

Crest factor is not only audible, but it is also a basic determinant of audio character, which is to say that a program with a large crest factor has a distinctly different sound character and sonic and musical meaning than a program with a small crest factor. When we squash the crest factor with a fast-acting limiter, we change something fundamental about that sonic "meaning" of the signal. Not necessarily good.

SUMMARY

Compressors and limiters are complex devices. Their action is audible in ways that are often unexpected and sometimes hard to describe. We need them in our work, but we often don't understand all the sonic implications of what they are doing.

One of the key things to keep in mind about a compressor is that we don't normally hear gain reduction as such, unless it is massive. The track may feel warmer, more stable and secure, easier to listen to for reasons that don't seem obvious, and so on. As a general rule, we just don't hear the level itself bobbing up and down.

I personally think this is why compressors are so hard to hear, and to understand.

The behavior of the level detector in each given compressor is a big part of this. And beyond the parameters I have already described, designers can add all sorts of things to manipulate the detector. Unfortunately, such designs and the capability of any given level detector are usually not revealed (except in marketing terms such as "hyperacoustic temporal sensing"), so we are limited to guessing at what they are doing, which can be annoying and difficult.

Such confusions require that we use our ears, which is a good thing, because that's all that our end-users have.

Thanks for listening. And thanks to Eddy Bogh Brixen ("Audio Metering," Broadcast Publishing) for some really useful information for this article.

This series on the basics of compressors was published in TV Technology magazine. Reach Dave Moulton via his Web site moultonlabs.com.

Fuse vs. Circuit Breaker?

Choose Your Protection to Fit Your Application

BY CHARLES S. FITCH

Society of Broadcast Engineers certification is the emblem of professionalism in broadcast engineering. To help you get in the certification exam taking frame of mind, Radio World Engineering Extra poses a typical question in each column. Although similar in style and content to the exam questions, these are not from past exams nor will they be on future exams in this exact form.

Protect Me, CB!

Question posed in the Dec. 9 issue

(Exam level: CBRE)

When is a fuse more desirable than a circuit breaker as an overcurrent protection device?

a. A fuse is never more desirable in a circuit
b. A fuse is always more desirable in a circuit
c. A fuse is more desirable when the current is over 100 amps for economic reasons
d. A fuse is always more desirable because it means you get a service call
e. A fuse is more desirable when instantaneous interruption is needed

The correct answer, or more aptly the most correct answer, to the question at right is the final one.

A circuit breaker (CB) is an electromechanical device. Due to the intrinsic mechanical transit time needed for any action to take place, a fast-blow fuse can provide faster circuit interruption than a CB.

A circuit breaker senses over-current in two ways: First, over the long haul, by thermally heating a sensor; second, in the relative short term, by magnetic trip, where the excess current flow builds a field in a magnet until the magnet pulls in a trip actuator opening the circuit.

A fuse ("fusus" is Latin for melted) can be constructed of various conductive materials in many different enclosure formats. In any given fuse, the conductive material is selected in type and sizing such that when a predetermined, rated maximum level of current is exceeded for a forecast amount of time, the conductive material fails by melting thus opening the circuit. Ultra-fast fuses can "blow" in under 0.1 second.

> **A fast-blow fuse** can provide a faster circuit interruption than a CB.

To review the other answer choices:

Answer (a) is wrong on its face; as noted, the faster-acting fuse over the slower CB can be more desirable.

Answer (b) is wrong as both of these protection devices have many unique desirable qualities. For instance, a CB can be reused by simply removing the overload and resetting the CB. In contrast, more choices of incremental current values and response curves are available in fuses to protect delicate or expensive components closer to their operating values.

Answer (c) is wrong as both CBs and fuses have ratings to 1000 amp plus and the economics involved are not that far apart. The decision for which device to use is really a judgment based on device design and application.

Answer (d) is pure humor and if you don't think it is, you're in this profession for the wrong reasons.

ONE-SHOT DEAL

Actually, humor aside, an important point is reflected in answer (d): the one-time-use feature of a fuse.

When a fuse is used in the devices found in broadcast plants, a blown fuse

(continued on page 20)

A fuse for every muse. The lineup starts with the ubiquitous 3-AG format and a typical holder on the left, to a 480 volt motor protection one-time use (as opposed to a cartridge with replacement fuse strips) on the right. Two Society of Auto Engineering (SAE) standard automobile fuses are on the bottom. Color coding indicates ampere rating.

FUSE

(continued from page 19)

signals a failure circumstance. Replacing the fuse and restarting the device before an evaluation can be made may enlarge the damage or reactivate a dangerous condition.

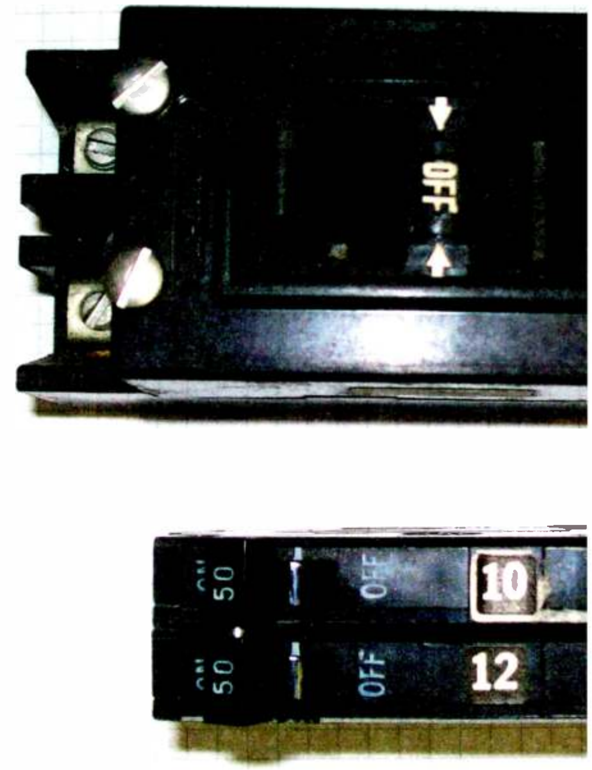
The designers of many devices put the fuses inside for this very reason, and your designs and construction may want to emulate this arrangement so that your overly ambitious staff members don't keep replacing the fuses until they have fried everything back to the input line cord.

Since we're mentioning fuses in equipment you build, let me note some configuration conventions.

For chassis-mounted fuse holders, the supply is always on the back pin and not the ring connection point. This places the supply potential furthest from your possible contact. When you make your fuse and holder decisions, size them for the expected maximum current and voltage potential present.

In this age of shared stations, where someone as unfamiliar with your plant as a competitor's engineer may be covering for you in an exigent circumstance, always mark the type and current rating for any critical fuse next to the holder. Also helpful is to note the normal current that would be flowing through that fuse to help in troubleshooting.

A closing comment related to me by an instructor at a Motorola two-way radio school: Be wary of people who try to sell you "100 percent pre-tested fuses."



Two pedestrian circuit breakers: a bolt-in type with lugs as often found in transmitters, and a panel 'snap-in' version.

The deadline for signing up for the next cycle of SBE certification exams is March 26, 2010 for exams given at the NAB Show on April 13, 2010.

Missed some SBE Certification Corners or want to review them for your next exam? See the "Certification" tab under Columns at radioworld.com.

Charles "Buc" Fitch, P.E., CPBE, AMD, is a frequent contributor to Radio World.

PRODUCTS & SERVICES SHOWCASE

Watch That Drop

Question for next time (Exam level: CPBE)

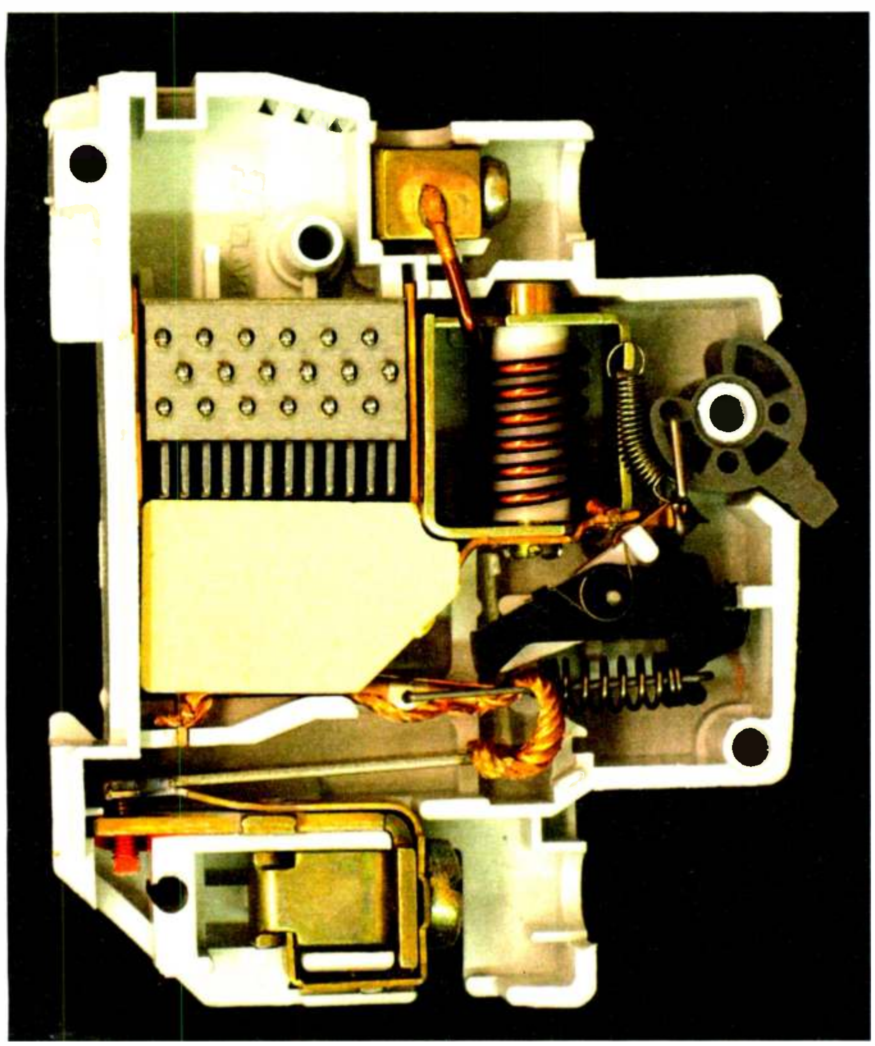
You are preparing an executive summary for management outlining the projected cost of various aspects of new facilities at your station. You need to calculate the wire size for the main power feed to the remote truck.

This truck will be used as the ultimate studio backup in the event of a long-term utility power outage. The HVAC and terminal equipment in the truck make this a notable load, so the wire is a large part of the cost. An NEC requirement is to limit voltage drops at the load to under 5 percent of the nominal supply voltage.

The worst-case load, with both HVAC units running on a 95 F day and all lighting and gear running, is 62 amps at 240 volts. The wire distance to the trucks reserve parking space is 250 feet.

Ignoring power factor (PF) and incidental issues, from the panel CB to the local disconnect supplying the truck, which size wire will limit the voltage drop to under 5 percent?

a. #18 copper TW
b. #16 copper TW
c. #12 copper THHN
d. #8 aluminum USE
e. #4 copper THHN



Exposed View of Inner Mechanism of a Circuit Breaker

DEBT

(continued from page 22)

Various news services have reported examples of the restructuring of debt in our industry.

Citadel Broadcasting Inc., the third largest broadcast company in the country, renegotiated its loan agreement twice in the last year and received a waiver on some of its loan covenants. Even so, the company skipped a $2 million payment on its loans.

According to a recent article in the Wall Street Journal, Citadel Broadcasting was preparing to file a prearranged Chapter 11 bankruptcy in order to convert $2 billion of debt into a combination of debt and 99.5 percent of the equity in the reorganized company. If approved, current stockholders effectively would be wiped out and some 90 lenders would become the new owners. The debt rose dramatically when Citadel acquired Walt Disney's ABC stations in 2006.

Similarly, Emmis Communications Corp. amended its debt agreements twice this year, and Regent Communications Inc. fell into default. Clear Channel Communications is struggling with about $20 billion of debt following a leveraged buyout last year.

Robin Flynn, senior analyst at SNL Kagan, has commented, "In recent years, radio has suffered from a 'leverage hangover.' Back in 2002, equity [stock] made up, on average, 76 percent of total market capitalization. However, that flipped in 2008, with 73 percent of total capitalization representing debt obligations."

If bankruptcy would make the radio industry stable, even at a reduced financial level, why isn't that a good option? The answer is that each of its stakeholders is in a very different position relative to the past sins. Investors and senior executives would find that their stock became worthless. Investment banks would be forced to write off bad loans and would become owners of radio assets without any knowledge about how to manage those properties.

Investment banks and equity owners have therefore taken the only rational path (for them) that could lead to their salvation: Squeeze expenses by reducing staff, equipment maintenance and facilities. Regardless of the damage to the long-term viability of radio, "burning the furniture to stay warm" becomes their preferred choice. Conversely, employees who provide valuable and substantive services to listeners would benefit if past financial sins were written off.

Evolution did not design our brains to handle time.

Like many others, I believe that commercial radio is, and will be, viable in the long term. But we cannot get to that point until the past sins are paid for by someone.

In the meantime, try to protect yourself from the combat among the financial titans, avoiding being an innocent victim of the battle that arose from human foibles of the past two decades among those who were in charge.

I would not dare to predict who will eat the hot potato because I, too, have no ability to envision the future. But at least we can watch the show as a perverse kind of entertainment.

Barry Blesser is director of engineering for 25-Seven Systems. Visit his Web site at www.blesser.net.

Who Pays for Radio's Past Sins?

When Debt Becomes Too Large to Pay, Someone Eventually Has to Take the Loss

BY BARRY BLESSER

Looking at the financial aspects of the radio industry provides insight on the behavior of its many stakeholders. For those of us who took a basic economics course years ago, we learned that it was a science based on rational assumptions, with such concepts as elasticity, supply-demand ratio and optimized self-interest. The assumptions, with supporting data, easily could be programmed into economic models that would predict the future of a business. Like engineering, models provide predictions. But are they valid?

Recently, the assumption of predictable rationality has been challenged by a group of behavioral economists who make a compelling case that economic activities, like all human endeavors, are often or mostly irrational.

In his New York Times bestseller "Predictably Irrational," author Dan Ariely says people do not make economic decisions using rational assumptions; even the smartest and best-trained professionals are emotionally biased by their unconscious cognitive framework. For example, holding a cup of hot coffee or a glass of cold soda influences the decisions that people make 10 minutes later. People are unaware of their unconscious biases and distortions.

Ariely's premise is directly relevant to the current mess in the radio industry.

Part of the problem is that human beings tend to extrapolate from the present without realizing that their interpretation of "now" is not a good predictor of the future. Boom and bust cycles arise from our flawed view of the future. Evolution did not design our brains to handle time. Rather, we are designed to handle the opportunities and threats in the present, using an often distorted sense of history.

To illustrate the role of the future in economic behavior, let's look at a simplified example of what is called leverage.

LEVERAGE

Assume that an investor has $100 cash to buy a stock or business, and assume that such investments have historically been increasing in value by 10 percent per year. At the end of each year, the value of the investment has increased by an additional 10 percent.

But what happens if that same investor can borrow $900 from a friendly bank (ignore interest), which allows him to purchase $1,000 of stock? At the end of the first year, a 10 percent return on $1,000 of stock produces $100 gain relative to the original $100 cash invested, or a 100 percent return. At the end of 8 years, the investment is worth $2,100, a $2,000 gain on the initial $100 of invested cash, a 20-fold increase. This is the power of leverage: using borrowed money.

The radio industry is now in the early stages of 'Who holds the hot potato?'

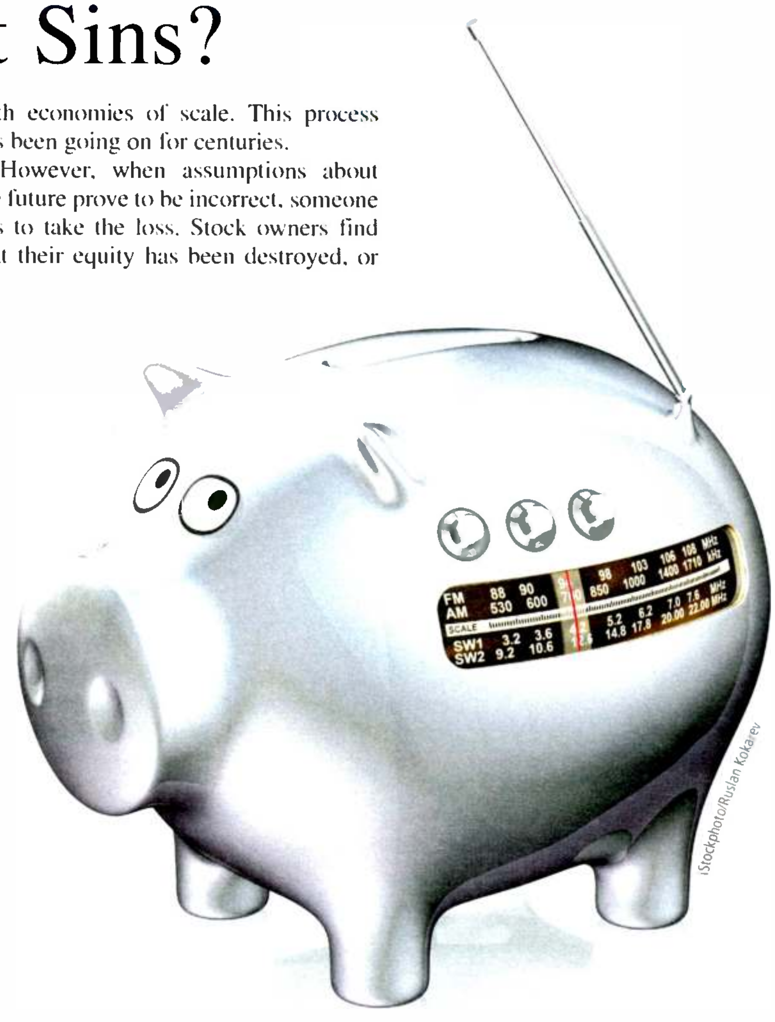
However, if the investment goes south, the reverse dynamic takes place. If the investment value plummets to $900 in the first year, the investor's original $100 cash is wiped out. If it further decreases to $500, the investor is in debt with no means of salvaging the bad decision.

Leverage is always a two-way street, and the outcome depends on assumptions about the future.

During the boom years of broadcast radio, many people assumed a growth model, and they assumed that this growth would continue indefinitely, albeit with temporary ups and downs. Most of the large radio networks were created with borrowed money — leverage — that was used to support mergers, acquisitions and expanded facilities. The investment banking industry provided loans because they collected healthy fees. The same dynamic played out in other industries.

There is nothing intrinsically wrong with mergers and acquisition because large organizations can raise productivity with economies of scale. This process has been going on for centuries.

However, when assumptions about the future prove to be incorrect, someone has to take the loss. Stock owners find that their equity has been destroyed, or investment banks find that their loans have to be written off. The same dynamic is playing out in real estate, with homes that are worth less than their outstanding mortgage. The radio industry is now in the early stages of "Who holds the hot potato?"



iStockphoto/Ruslan Kokarev

Most countries created bankruptcy laws to allow the courts equitably to distribute the pain of bad decisions among various stakeholders. When assets have less value than liabilities, the organization is at a financial dead end.

But there are two types of dead ends. In the one case, the company must be liquidated because there is no way to restore a balance between income and expenses. For example, Polaroid simply disappeared because there would never again be a market for their instant cameras. In other cases, a company can be reorganized to become viable if past liabilities are written off, stock is made worthless, debt holders are forced to become stockholders, and life goes on for employees and customers.

United Airlines went bankrupt in 2002, but through reorganization, it is still flying passengers around the world. This financial dynamic is as old as recorded history. With the boom in railroad construction in the 19th century, holding companies went bankrupt at a regular rate. But the railroads continued to run, since they provided a valuable service for transporting freight and passengers. Investors, especially the investment trusts in England, held the hot potato. Bankruptcy washed out the bank loans and investor equity, but that did not stop railroads from functioning.

Commercial radio stations may well be viable in the long term, even with reduced advertising revenue, as long as income still exceeds the cost of keeping the station on the air: namely, electricity and minimal labor cost.

What happens if a positive operational cash flow is not sufficient to also cover paying interest on loans and eventually to repay the principle? If nothing is done about continuing deficits, eventually liabilities exceed assets. Something has to give.

REAL WORLD

There are numerous examples of how this scenario is now playing out.

(continued on page 21)