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# Contents

BE Radio
September/October 1997
Volume 3, Number 6

## FEATURES



## DEPARTMENTS



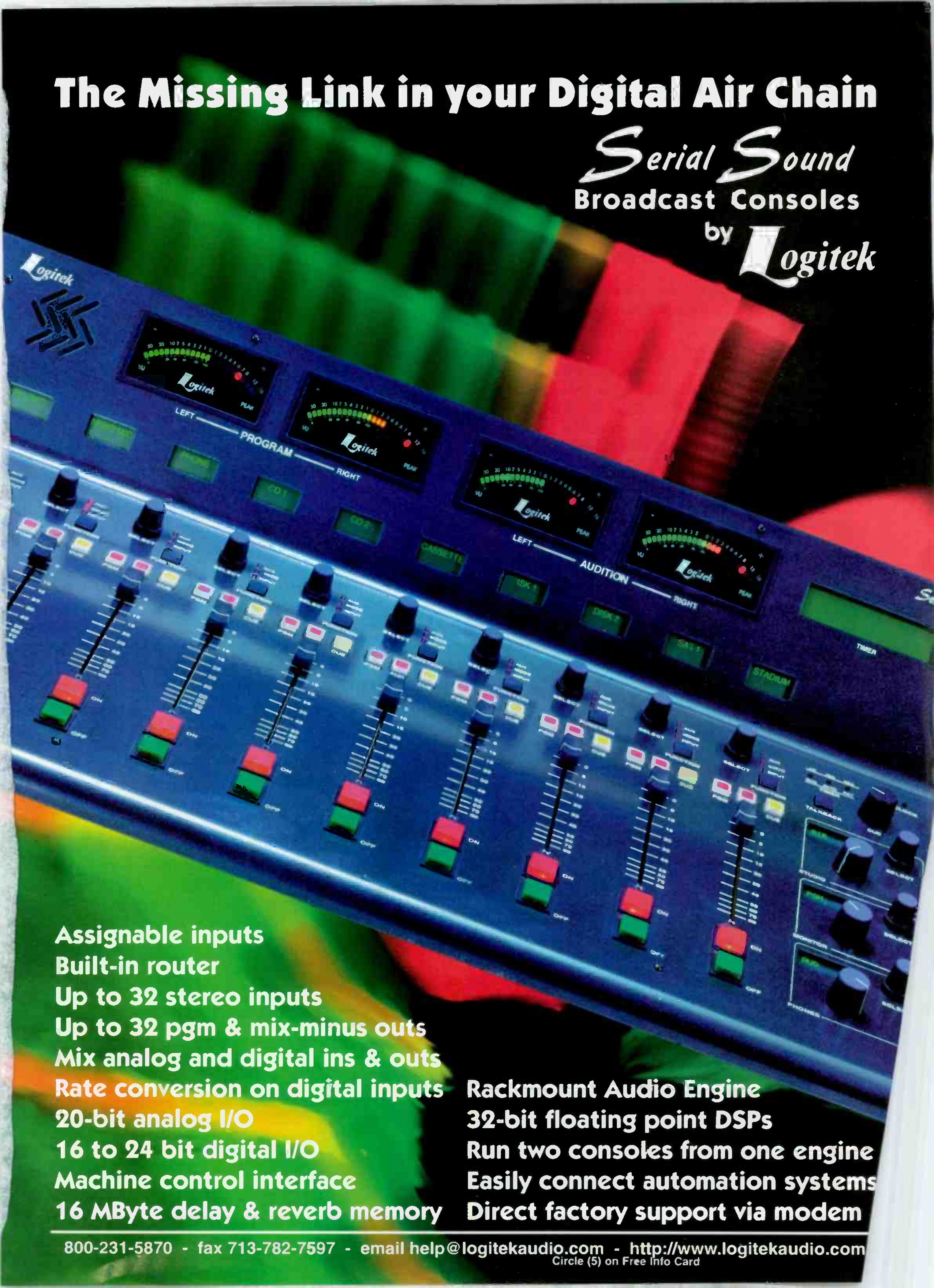
24

46

72

18

**ON THE COVER:** Light up the phones and stir up the conversation. Talk Radio is a unique format that presents its own special challenges. "Go ahead caller, you're on the air..." (Cover design by Stephanie Masterson, *BE Radio* art director.)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# Editorial

# Getting to know your TCO

From my new vantage point in the Silicon Forest, it's easy to keep track of all things computer-based. So, when a new report crossed my desk the other day, I realized it had strong implications for radio broadcasters.

The subject was *total cost of ownership* (TCO) for Windows-based PCs, a hot-button issue in the computer industry right now. TCO studies are intended to help an organization understand and manage the costs of owning and operating PCs and related equipment. This new report was developed by Interpose, Inc. (a leading TCO analyst based in Orlando, FL), in cooperation with Microsoft for its Windows operating system.

The report first divides TCO into direct and indirect costs. Items considered as direct costs include hardware and software purchases, plus system management, support, development and communications. Indirect items attempt to quantify the "opportunity costs" of lost production due to downtime and other computer-related problems.

The first thing that jumped out at me was the fact that total hardware and software expenses amount to only *11%* of TCO. That means that the true cost of operating a PC is nearly an order of magnitude (i.e., one decimal place, to the bean counters) greater than the initial purchase price of the machine, including its software. A recent ad by Hewlett-Packard addressed this issue nicely, using the visual metaphor of an iceberg. You get the picture.

The next important fact showed that fully 18% of TCO is attributed to indirect costs of "user downtime." This means that the system is up, but users can't do their jobs due to lack of training or understanding of the computer system's proper operation. So, they spend time unproductively by "futzing" with their PCs. In some cases, this is the fault of poor user-interface design or steep learning curves, while in others it stems from insufficient support — either from the software vendor or the computer owner's IS department.

Also interesting was that the single biggest item (at 24% of TCO) was spent on support. Combined with system management (at 19% of TCO), this shows that nearly half of a company's computer costs — or more than four times its hardware/software expenses — go to the personnel, either inside or outside the firm, who keep the computer system running. Yet, even this may not be enough, given the 18% futz-factor noted above.

Summarizing these expenditures, the report concludes that the average Windows-based networked system costs a company $7,251 per desktop per year. (Other reports have placed this figure as high as $12,000.)

For general administrative applications, there's got to be more to the equation than this cost analysis, otherwise it would make more sense to go back to typewriters and adding machines. Indeed, those other elements are the productivity, quality and creativity that computer-based operations allow and encourage — but there's no guarantee these will automatically occur. Management must stimulate the appropriate use of the computer by staff to get its money's worth, along with hiring people with the requisite skills — both among users and support personnel.

On the other hand, computers are used in radio for functions other than word processing and spreadsheets, and TCO should be considered from a company-wide, comparative viewpoint. The computer systems used for audio production and program-stream assembly are often substantially *less expensive* than equivalent traditional audio equipment, so you're in a positive position from the start. Annual support costs for these operations also may not be significantly different from conventional systems (even though the actual support *people* might be). Moreover, the cost of keeping production on conventional hardware continues to go up, while the overall cost-effectiveness of computer-based operations is improving.

The conclusion? Radio has good reason to turn to computers, both financial and creative. But make sure you know what you're getting into. Understanding and applying TCO principles will help you keep long-term computer costs under control. Most important, be sure to budget support levels properly — including user training — and hire the right people or organizations to carry it out. Otherwise, you're better off sticking with carts and reels.

**Skip Pizzi, editor in chief**

*For further information on TCO, check the Interpose web site at http://www.interpose.com.*

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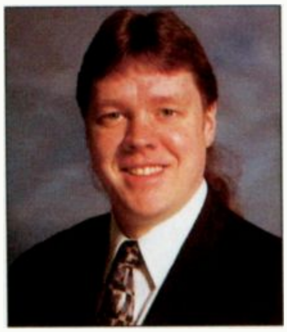
# Road trip!

Labor Day has passed and the tradition of summer radio remotes is behind you. As you prepare for the fourth quarter of 1997, there is only one thing to do: take a road trip.

Being able to take a break and get away from the normal routine is always healthy. It helps clear your mind and can give you a fresh perspective when you return.

With the fall comes the NAB Radio Show in New Orleans. Now that the World Media Expo is gone, the show returns to its original format. Although the spring show is a great event, it covers so many areas that it is easy to miss certain aspects. This four-day fall event has one big advantage over the spring NAB: *it's just radio. BE Radio* will be there to cover all the news and new technology presented.



One of the myths about the fall show is that it is a programming-only event. This is simply not true. Although there are sessions for programming, there are also sessions for management, owners and engineers. Three technical workshops are being offered that cover AM directional arrays, radio frequency radiation and AM/FM transmitters. If you can attend any of these, I recommend it. The sessions are a great opportunity to learn more or maybe just get a refresher course. There is no such thing as too much education.

The exhibit hall is also the best way to go for a test drive or to at least kick the tires on all of the new equipment. There is still plenty of information on the exhibit floor to keep you apprised of the current trends.

One week after NAB Radio is the 104th AES convention in New York City. The main focus of this show is generally geared toward studio recording and live sound, but there are plenty of attractions for radio as well. On Friday, Sept. 26, there will be a forum on DAB. Representatives from the leading proponents of the various DAB systems will be there to discuss the current trends and research. The recording studios know it's coming, and they are getting ready for it. This is the one forum that I most want to attend to see how the allied audio fields are viewing this upcoming technology. A digital standard and deadline has been given to television, and you need to be aware of what is happening in DAB so you can be prepared for it when it's our turn.

The AES has four full days of sessions planned on a variety of topics, including digital audio encoding, ISDN and, of course, Internet audio delivery. There are also several tours planned, including a visit to WQHT/WRKS, New York, for a facility tour of the first all-digital air chain from CD to transmitter.

With the face of change constantly looking at you and your facility, the trade conventions are an increasingly better way to learn about and see the technology that is coming and how it will affect your professional career. It may not always be in the station's budget (or your own budget if you are a contractor) to go to the larger conventions, but there are alternatives. Several regional SBE conventions and equipment vendor expos often offer informative sessions and workshops that provide an excellent opportunity to advance your knowledge and, hopefully, your career.

As more stations consolidate, the need for qualified engineers will only grow. There may be fewer at a given facility, but the responsibilities will not decrease at all. Look at the impact computers alone have made to the job of the broadcast engineer. The technical ability of the average station employee is not increasing to keep up with the new technology being implemented, so your ability must increase. Greater ability means more value to the station and hopefully more security to your career. As the pool of new recruits continues to dry up, attending conventions is one way to keep yourself up-to-date and in demand. Be sure to stay on top and aware of the changes that are affecting us every day.

The consolidation of duties of station personnel is everywhere. Engineers need to pick up on some of that by becoming their own marketing department. Add to your arsenal of tools with knowledge and understanding. Make the effort to become certified or trained to show a higher proficiency in some relevant area, be it computers, RF or audio. If you don't toot your own horn once in a while, no one else will.

**Chriss Scherer, editor**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# LAN basics, part 2

By Kevin McNamara, CNE

Last month, in part 1, we looked at the *open systems interconnect* (OSI) model and some of the basic LAN building blocks, including channel access methods.

## Topologies

Of all the layers defined by the OSI, we should be most interested in the *physical layer*. This is the level that we, as engineers, will have the most involvement wtih. The physical layer deals with the cabling, connectors and interfaces required to physically attach the computers on the network. The types of topologies that you will most likely run across are:

- *Bus*. Consists of a linear section of media, such as coaxial cable, terminated on each end and are connected through a "tee" connector.
- *Ring*. Similar to the "bus," except the ends are joined, hence the name. In practice, the ring configuration actually consists of dual counter-rotating concentric rings. Devices attached to a ring network will have "in" and "out" ports.
- *Star*. More commonly referred to as a hub configuration. The signals are carried from each device on the network to a central point (the hub). These hubs can be passive or active.

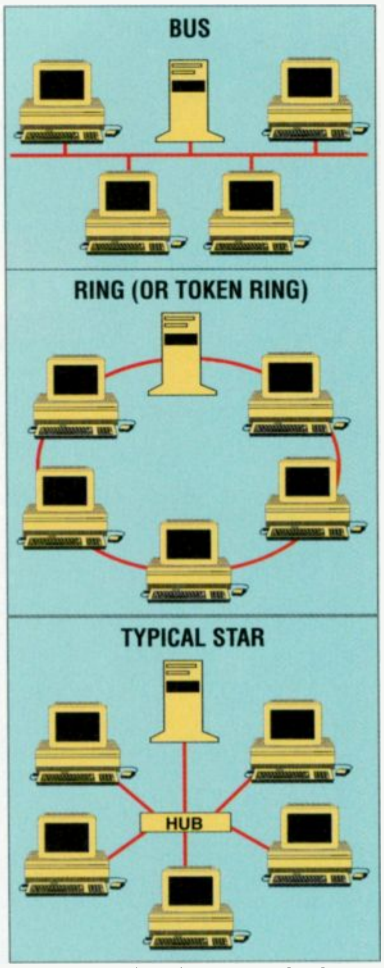


**Figure 1. The three standard network topologies (bus, token ring and star).**

## Network protocols

By definition, *network protocol* refers to the rules that govern how any two or more elements on a network communicate. The OSI is defined by seven protocol layers. Working committees of the IEEE are responsible for defining and ultimately standardizing the various LAN protocols at the datalink and physical layers. For our purposes, the primary differences of these standards relate to data throughput, transmission media (cabling) and channel access method.

Currently, the 100Mb/s Ethernet protocol has become the LAN protocol of choice for most new installations. Keep in mind that standards are currently in development for 1,000Mb/s Ethernet.

Three types of media have been specified to transmit 100Mb/s Ethernet signals:

1. *100BASE-TX* (Fast Ethernet). Essentially a faster "contention"-type access method.
2. *100BASE-T4* (100VG-AnyLAN). Uses a new approach to access the network called *demand priority*. This system supports *token passing*. One of the real advantages of this method is that it uses four pairs of telephone voice-grade cable. The VG stands for voice grade.
3. *100BASE-FX*. Fiber-based "Fast Ethernet" similar to FDDI.

Here are a few more high-speed protocols you should be aware of:

- *Asynchronous transfer mode* (ATM). A technology based on high-speed packet switching, ideal for multimedia and other complex applications. ATM is capable of data rates from 51Mb/s to 622Mb/s.
- *Copper data distribution interface* (CDDI). FDDI using copper-based media instead of fiber.
- *Fibre Channel*. Developed jointly between Hewlett-Packard, IBM and Sun Microsystems. Fibre Channel requires the use of fiber cabling and is capable of passing data at a rate from 266Mb/s up to several Gigabits/s.

These are some of the high-performance protocols available. All of these offer data throughputs of 100Mb/s or more, however, most of the PC-based networks in existence today are still using protocols that pass data at approximately 10Mb/s. In terms of performance, these will work just fine for many business-type applications, but generally are not suitable for installations requiring high throughputs or passing real-time information. Some of the 10Mb/s Ethernet standards are:

- *10BaseT.* Uses UTP cabling in a hub arrangement.
- *10Base2.* Uses "thin" coax (RG-58), arranged in a bus configuration.
- *10Base5.* "Thick Ethernet" uses heavier coax (RG-8), allowing longer line distances.

Although not considered a high-speed protocol, *wireless Ethernet* is becoming a popular media for connecting portable workstations (laptops) to a LAN or for connecting LANs between different buildings where cabling is impractical. The FCC has permitted unlicensed operation of these systems at 902-928MHz and 2.4-2.483GHz using spread-spectrum modulation. Most of these systems will provide data rates from 2Mb/s to 10Mb/s, at distances of approximately 30 miles with directional antennas.

For high data rates via a wireless media, consider using a standard point-to-point T1 microwave system. This approach is typically more expensive and requires FCC licensing.

## The LAN infrastructure

Proper planning and installation of the *LAN infrastructure* is the single most important task in the building of a reliable and upwardly expandable LAN. Given the current pace of technology, this is the one area that probably won't need to be replaced in the next five years if the wiring and power systems are designed and constructed properly. Strangely, many companies (particularly broadcasters) skimp on this phase and, consequently, spend a great deal of money and time resolving related problems.

Coaxial cables, once the de facto standard to connect devices on a LAN, are becoming less common. Unshielded twisted pair (UTP) and fiber cables are currently the most popular transmission media. Depending on the size of your LAN and the distance it must cover, the cabling and installation costs will represent a significant portion of the project. Each type of media has pros and cons. Let's take a look at each cable type and the connectors for it.

The typical fiber media used for LANs consists of a graded index multimode fiber-optic cable with a 62.5 micron fiber-optic core and 125 micron outer cladding. The light source used to drive the fiber media operates at a wavelength of 1,350nm. Single-mode fiber is used in applications requiring wider bandwidths and longer distances.

Three types of connectors may be used to connect the fiber cable to an interface:

1. *FDDI media interface connector* (MIC). These connectors are keyed in various ways and care should be taken to identify the proper keying arrangement. Connectors push in to the mating receptacle.
2. *Duplex SC connector.* Designed for simplicity, the connector is just pushed onto its mating section and the connection is completed.
3. *ST connector.* A spring-loaded bayonet-type connector. Both the outer bayonet ring and inner sleeve are keyed.

The losses on a fiber cable should not exceed 11dB for the entire run. You can expect 1dB to 2dB of loss per 1,000 meters of cable and an additional 0.5dB to 2dB for each connector, depending on how well it was put on. Dust and skin oils will substantially affect the system losses.

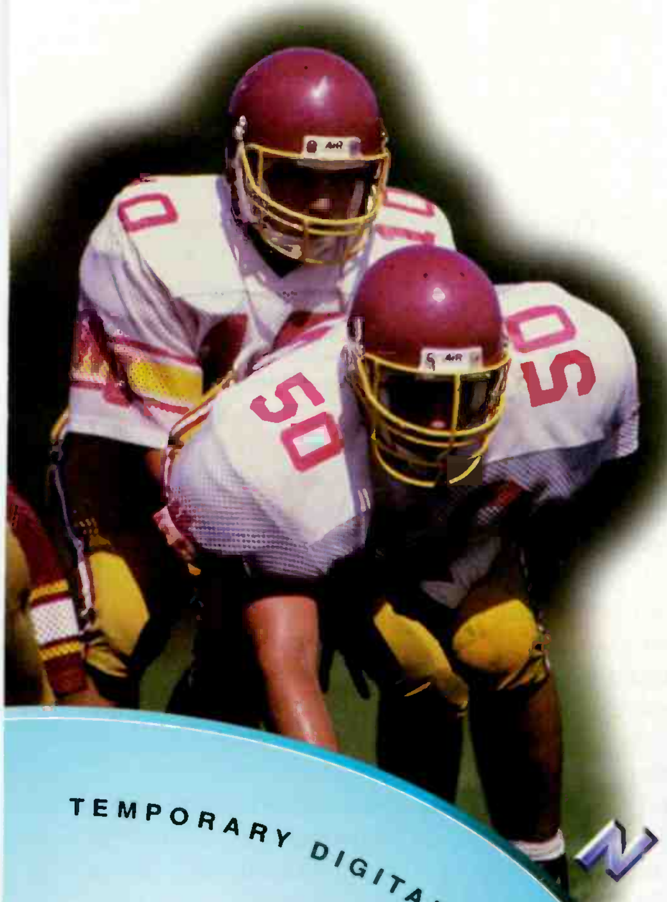
There is a variety of copper-based media to choose from that have advantages in cost, durability and, to some extent, ease of installation. These cable types consist of twisted pairs clad in an unshielded jacket and the primary difference between them is the capacitance between conductors that attenuates higher frequencies. This becomes a critical issue when the goal is to transmit 100Mb/s through long distances of cable.

Care must be exercised in the installation of fiber and copper cabling. Kinks, extreme bends and improper attachment of connectors can wreak havoc on the best-planned installations. Indications of poor installation practices appear as slower throughputs, communication errors and poor reliability. In order to

*Continued on page 83*

# Digital audio links

for broadcasters



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# Datacasting

**David Maxson, CBRE**

Datacasting is alive and growing in the United States and FM broadcasters have been a part of the industry from the beginning. Where is the business going? What will be the role of the radio broadcaster in datacasting?

Miles Beam's article, "Data Broadcasting: The Next Frontier," in the November/December 1995 issue of *BE Radio* defined datacasting as the transmission of data over a one-way link from a single point to multiple receiving points. This simple definition now encompasses a vast array of communications services. Key players in the datacasting industry are buying up smaller ones to gain market share and distribution channels. Companies, like WavePhore and Data Broadcast Corporation (DBC), are defining just what it is to be a *datacaster*.

A look at some of the datacasting companies will show how datacasting by broadcast radio fits into the overall datacasting picture. WavePhore claims to be "the leading operator worldwide of data broadcasting networks" and a "provider of proprietary products and services for low-cost, high-speed data broadcasting systems for distributing digital data via the existing worldwide TV, radio and satellite broadcast infrastructures and the Internet."

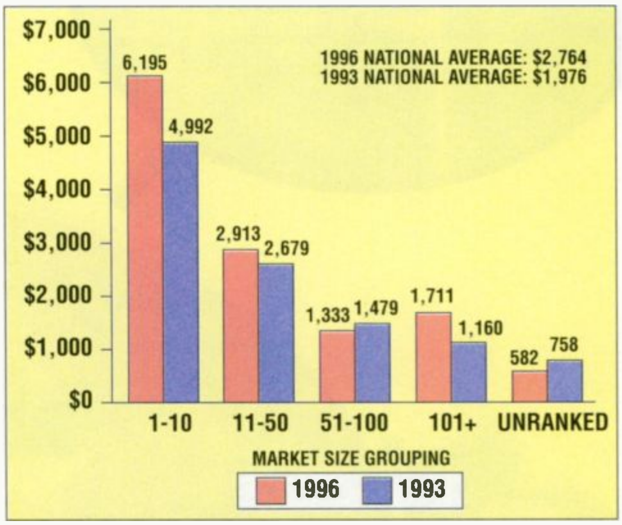



**Figure 1. Average monthly lease rate for a station's first subcarrier by market size.**

One of its growth acquisitions was Mainstream Data, which is known to many FM broadcasters as a tenant on their subcarrier spectrum. Primarily providing services to businesses, WavePhore enters the consumer marketplace this fall. This new service, called WaveTop, will use auxiliary TV bandwidth to deliver worldwide web-like functionality to TV set-tops.

Another big datacaster rose from the ashes of the Financial News Network. Through its information services division and its business services division, Data Broadcasting Corporation did $114 million in business in 1996. It acquired the company that developed the Quotrek stock market data service that is delivered on FM subcarriers. The service provides timely financial data to subscribers with special portable or computer-connected receivers.

DBC provides information services to approximately 40,000 subscribers via 76 FM subcarriers in the United States and Canada, as well as by satellite, TV vertical blanking interval (VBI), cable television, telephone and Internet. Its business services division includes Instore Satellite Network and CheckRite International, a check verification service.

The trend in datacasting is becoming pretty clear. Companies specializing in data acquisition and delivery are taking advantage of a host of different transmission media. Factors, such as cost, bandwidth, immediacy, geography and reliability, affect decisions on what medium to use.

## Leasing your subcarrier

What is an FM subcarrier worth? The newly published "FM Subcarrier Market Report/Technology Guide" from the National Association of Broadcasters (NAB) sheds light on the current state of the medium. Surveying a large sample of stations, NAB developed average monthly fees for FM subcarriers in different size markets. In top 10 markets, the first subcarrier leased on a station averaged $6,195 per month. In markets 11-50, that figure dropped to an average of $2,913 per month.

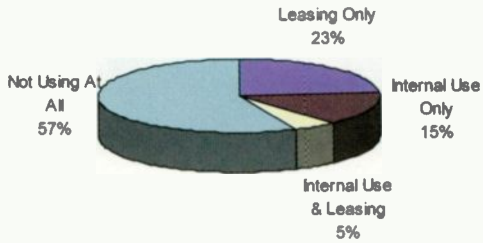
The NAB reports that less than 30% of all FM stations in the United States lease any subcarrier capacity. Demand is greater in the larger markets. About half the top 50 market stations lease some subcarrier capacity.

The report goes on to explore why subcarrier penetration remains fairly low. The most reasons given for not leasing subcarrier capacity were related to lack of demand for it. Coming in a distant second were concerns about effects of subcarriers on station quality.

Among those reporting subcarrier use, only half of the subcarriers mentioned are related to datacasting. The other half contain background music, foreign language services and reading services. Of the datacasting subcar-

riers, nearly two-thirds are used by paging services. The rest involve the now-familiar financial data and utility load management data, plus the more recent addition of Global Positioning System differential corrections data.

What's keeping datacasting's use of auxiliary FM spectrum so low? It might be because we think of the subcarrier space on FM signals as just that, "auxiliary." Many datacasters have their own proprietary waveforms and/or data protocols. As a result, subcarrier spectrum is sparsely populated and is plagued with incompatible data transmission systems.




Source: 1996 NAB FM Subcarrier Survey

**Figure 2. Distribution of FM subcarrier usage.**

Two attempts at subcarrier standards setting have been made under the auspices of the National Radio Systems Committee (NRSC), a joint effort of NAB and the Electronic Industries Association (EIA). One standard, RBDS, was adopted in 1993. The other, a high-speed subcarrier standard, is a work in progress.

The NRSC adopted the voluntary Radio Broadcast Data System (RBDS) standard for a low-speed subcarrier capable of transmitting, among other things, data related to an FM station and its programming. It is based on the European Radio Data System (RDS). In 1995, the EIA launched a program to equip stations in major markets with RBDS encoders. According to the NAB, by late 1996 only one-eighth of the stations surveyed were broadcasting RBDS signals. The EIA's objective was to equip stations in major centers of population with RBDS. This would allow most radio-buying consumers to see the benefits of RBDS in the showroom.

The other NRSC effort in support of datacasting is being conducted by its High-Speed Sub-Carrier (HSSC) subcommittee. It is reviewing tests of systems provided by three proponents: Mitre Corporation's subcarrier technology was developed in conjunction with the Federal Highway Administration; Seiko delivers messages to receivers as small as wristwatches; and Digital DJ transmits menu-based information to hand-held consumer receivers.

The HSSC subcommittee is examining the technologies, not the content. Presumably, if a single technology is chosen as a standard, entrepreneurs will develop new content and services for it. FM broadcasters may see an increase in demand for their subcarrier spectrum after the subcommittee makes its recommendations.

Of the three proponents, Digital DJ is the only one providing services today where the host station can get in on the action. Consumer content is provided by Digital DJ. The host station can incorporate its own services and advertiser-supported content into the Digital DJ datastream.

Digital audio radio broadcasting promises to blur the distinction between datacasting and broadcasting. Proposed digital audio technologies may not be fully compatible with data subcarriers. The wise FM broadcaster will include an escape clause in subcarrier leases to provide for new digital broadcast standards when they are adopted.

### Blurring the border

The line separating datacasting and other forms of communication is getting fuzzy, too. Paging is perhaps one of the original forms of datacasting. According to Motorola, there are 40 million pagers in use in the United States today. These point-to-multipoint systems work on a variety of

*Continued on page 83*

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Circle (8) on Free Info Card

# Proof of performance

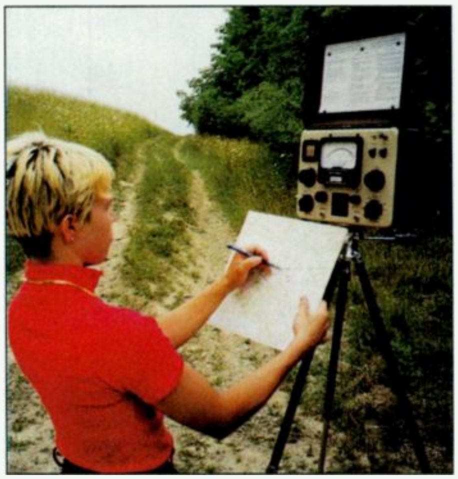
By John Battison, P.E., technical editor, RF

"We have to do a proof of performance." These words can strike terror into the heart of an engineer. When uttered by the FCC, it can mean that glaring discrepancies have been found in the pattern or a disgruntled competitor has filed a complaint. It can also be triggered by tower damage and subsequent rebuilding or a station facility change. In the case of an FM DA, it probably means that something has damaged the antenna.

## FM proofs

When an FM station is involved, the station engineer's work is usually less rigorous because FM antenna proofs are performed at the manufacturer's facility prior to installation unless required by an occurrence in the field. After antenna construction, it is clinically proven that the radiated signal meets the pattern constraints and the manufacturer's exhibit is accepted as proof.

The commission's rules do provide for making field-strength measurements. They involve a more-complicated procedure than for AM measurements, including a strip recorder for making runs of at least 30 meters with an antenna elevated 30 feet above ground. This last part can be dangerous. Engineers have been injured or killed while making these mobile measurements because they run the monitoring antenna into high-voltage power lines. If you must make this kind of measurement be absolutely sure of the location and height of power lines.


**Using a tripod to support the FIM gives a stable mount for multiple pattern readings.**

Sometimes, it is necessary to prove to management that the signal is going where it is supposed to. A convenient way to check this is by comparing the questionable signal with a known good one. Take a series of readings in a "good" area and then make similar ones in the "bad" area. If the station is non-directional with a circular pattern, there should not be much difference between them. Take into account circular polarization. It is often necessary to polarize the receiving antenna for local conditions. If there is a major difference between the "good" and "bad" signals, it is logical to go back toward the station on the same radial(s) to see where the discrepancy begins.

In extreme circumstances helicopter measurements are made. If the problem is great enough to involve such work, it can be performed by a consulting engineering firm that specializes in such work.

## AM proofs

AM directional antenna proofs have been a part of the radio engineer's life ever since the early days of broadcasting. Among the pioneers were Carl Smith, the late Andy Ring and Paul Godley. In fact, Andy Ring probably wrote most of the AM proof rules during his tenure at the FCC.

The rules haven't changed much over the years. Computers have helped make a relatively easy task of generating new antenna patterns from what was once a multiweek job of trial and error with scores of pattern calculations. Completely computerized proofs of performance have been proposed by the industry and development continues into ways of avoiding the laborious field proof of performance.

Sometimes, it is necessary to determine the effective field at 1km to establish that an antenna is producing the required effective field. The radial requirements in 73.186 are the same in terms of radial length and point spacing as for a proof of performance. The measurements are plotted and analyzed as for a proof. The effective field is then read from the graphs obtained.

## Preparing for a proof of performance

The first thing to do is check an up-to-date copy of the rules to be sure that there have been no changes since you last ran a proof. Read 73.151 and referenced paragraphs.

Be sure you have set up enough radials as specified in the CP. There will never be fewer than eight for a non-directional, more will be required for DAs. The more complex the pattern, the more radials will be required. Generally, three radials are needed for a major lobe. Whenever possible, avoid running radials through the side of lobes because it is hard to get accurate measurements on a curved line. Nulls and lobes are the most satisfactory points to measure, but sometimes the FCC requires measurements on a curved section of a pattern.

Be sure that you have enough topographic maps of

1:24,000 scale. Plot the antenna location *extremely* carefully. Your work will be wasted if you make a mistake here. A single non-directional tower can be located quite easily. In the case of a DA, the *center* of the array is used.

Laying out the radials is important. I've seen engineers fold over the edges of a map section and run a line across the join. At the end of 20 miles the radial is not where it is supposed to be! Don't cut the map margins off either; you may lose valuable data, and you may need the margin later.

I like to lay out my origin and take radials to the map edges. Then I measure off two or three reference points on the first map and locate the same point on the edge of the next (adjacent) map. With the protractor used to lay out the original radials, measure from this point on the edge of the new map to establish the radial direction on the second map. If followed for all maps, this method usually gives satisfactory results.

Some engineers prefer to pick measuring points as they go along because predetermined points don't always work. This is a reasonable argument, but I don't like this method. If points are chosen in accordance with the commission's required spacing and marked lightly on the map, you can adjust them as needed as the radial progresses and still be assured of taking enough readings.

Be accurate with the distance measurements, both when locating spots in the field and when measuring the map back in the office after runs are finished. If you say a measuring point is 100 yards into a field be sure it is 100 yards and not 150. With close-in and low-value nulls, a few yards can make a surprising difference. Monitor points should be picked based on suitable distance and freedom from overhead wires, train tracks and streams that may dry up or may be dry when you measure.

Record distance, time and measured values. If weather is extreme, a brief note may be helpful later when you are trying to repeat it. It is also absolutely essential to make a complete description of each measuring point location. Without this, it will be impossible to revisit the point and repeat. I've seen far too many older proofs with only a list of distances and no descriptions that had to be redone because it was not known where to measure. "Outside green house" is not acceptable. The house color may change with the next owner. Give the street name and number, for example, "North end of driveway at 4523 Green Street." It's far better to say too much than too little. Don't forget to take a photograph of each monitoring point with the FIM in the position used for measurement. The FCC requires this with the License Application Form 302.

**Careful and clear record-keeping is essential.**

Given a choice, it is often advantageous to run a proof in cold weather. Conductivity seems to increase in winter. If a proof is made when all readings are high, you may have a cushion later when the array ages and begins to cause trouble.

It goes without saying that your FIM must be in calibration. A calibration is accurate for about two years. If you are using more than one FIM identify the particular unit used on each radial. You should also record all serial and model numbers of all the meters in use.

Before starting runs check the calibration of all FIMs by measuring at a single point. There should be little difference between meters. In any case, record values measured and make more checks at different values to establish a calibration curve.

If you have to start close-in, you will need a chain 176 yards long to measure one-tenth-mile intervals. Using a shorter section and "leap-frogging" your distance will certainly lighten your load. Chain is preferred because rope can stretch. A sighting compass is essential to make sure the radials are exactly as planned. Remember, the FCC wants *true* azimuths, so correct your compass reading by applying the local variation to get true directions measured from true north. A two-person team is best for making field measurements: one to drive, take notes and read the map and another to hop in and out to make measurements and record locations and time. Doing both can be tiring.

Most field engineers hold the FIM on their shoulder to make readings, but a tripod is best for accurate work. This is especially the case when switching back and forth between DA and non-DA to get comparison readings at measurement points. It is more accurate to get measurements in both conditions at the same time and place rather than go back and recreate the "same" point at another time. If the FIM gives a maximum reading pointed far away from the antenna, try to find out why and consider using an alternate point.

Remember not to make measurements until two hours after sunrise and two hours before sunset. These are the critical hours and the ionosphere is a mess. The sporadic skywave can influence the measured fields.

After the field work comes the interpretation and determination of pattern shape and efficiency. Today, computer programs can analyze the measured fields and determine the conductivity and radiation. Otherwise, out comes the old "light table" and the adjusting of plotted radials vs. the FCC groundwave curves.

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*John Battison owns John H. Battison and Associates, a consulting engineering company in Loudonville, OH.*

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*Photos provided by Tracey Liston, WMIH-AM.*

# Who has the best Digital System for groups?


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Circle (11) on Free Info Card

# Congress authorizes broadcast spectrum auctions

By Harry C. Martin

The Balanced Budget Act of 1997, passed in August, requires the FCC to resolve conflicts among mutually exclusive broadcast applications through competitive bidding or auctions. Congress authorized the commission to conduct auctions to resolve mutual exclusivity among applicants who filed their applications on or before July 1, 1997. Also, Congress suspended the settlement cap rules beginning Aug. 5, to encourage settlements among competing applicants whose proposals were filed on or before July 1. Post July 1 applications will be auctioned, but the settlement caps won't be lifted for those applications.

The auction legislation raises questions that the FCC will have to answer in a rulemaking proceeding. These are some of the unresolved questions: It's unclear whether the FCC will proceed with auctions in any or all cases involving applications filed on or before July 1. The agency is authorized to conduct auctions with respect to these applications, but isn't required to do so.

It's unclear whether the auctions of post-July 1 applications will be open to anyone who wishes to bid rather than only to applicants who have reached cut-off status.

It's unclear whether the FCC will permit settlements among post-July 1 applicants, since the statute mandates auctions and no provision is made for settlements as a means of resolving mutual exclusivity among proposals filed after July 1.

The FCC has developed point-based comparative selection criteria that would have been universally implemented had Congress not approved auctions. Although unlikely, this methodology still could be used to resolve mutual exclusivity among applications filed on or before July 1.

## U.S. and Canada amend FM agreement

In July, the United States and Canada agreed to amend their bilateral FM agreement as follows:

- FM translators can now operate with up to 250W of effective radiated power (ERP) and with a 34dBu interfering contour distance of up to 60km.
- Concurrence by the other country is not required for FM translators where the interfering contour (34dBu) does not extend beyond the border.
- A 6kW maximum ERP level for Class A assignments was adopted.

By incorporating these relaxations into the allocations scheme, many translators within 175 miles of the border whose service potentials had been unduly limited, in spite of the non-existence of any interference possibility, can realize the maximum service capabilities for translators permitted under the FCC rules.

Another benefit is a relaxation in separation requirements to Canada for U.S. Class A FM stations. Until now, 6kW FM station allocation or upgrade proposals required notification as higher-powered Class B1 stations, leading to excessive separation distances, which in many cases, precluded a desired allocation or upgrade. By recognizing the 6kW maximum power authorized by the FCC, Canada's restrictions on Class A FM assignments in border areas have been significantly relaxed.

*Harry C. Martin is an attorney with Fletcher, Heald & Hildreth, PLC., Rosslyn, VA.*

## Changes by grandfathered FMs made easier

The FCC has revised the rules specifying how certain grandfathered short-spaced FM stations may improve their facilities or relocate transmitter sites.

Pre-1964 grandfathered stations are those stations that were short-spaced on Nov. 16, 1964, when the rules establishing minimum distance separations between stations became effective, and have remained short-spaced since that time.

The revised rules permit a co-channel or first-adjacent channel grandfathered short-spaced FM station to modify its facilities if the application demonstrates each of the following:

- There is no increase in the predicted interference area or the associated population.
- There is no increase in interference caused by the proposal to any individual grandfathered short-spaced station.
- Any area predicted to lose service as a result of interference has adequate service remaining.
- In addition, any applicant proposing interference caused in any areas where interference is not currently caused must serve a copy of the application with the licensee of each affected short-spaced station.

The rules also permit a grandfathered station that is short-spaced to a second-adjacent channel or a third-adjacent channel station to change transmitter location or other station facilities without regard to such short-spacing.

Finally, subject to the rules stated for co-channel and first-adjacent channel grandfathered stations, applicants previously required to obtain agreements to implement facility improvements or modifications are no longer required to obtain such agreements.

## dateline

On Oct. 1, 1997, radio stations in Alaska, Guam, Hawaii, Oregon, Washington and Samoa/Marianas must file their renewal applications.

Commercial radio stations in the following states and territories must file their annual ownership reports by Oct. 1: Alaska, Guam, Hawaii, Oregon, Washington, Samoa/Marianas, Florida, Puerto Rico, Virgin Islands, Iowa and Missouri.

Tower owners in Delaware, Kansas and Washington must register their towers with the FCC between Oct. 1 and Oct. 31.


810 KCMO

# TALK RADIO

## Talk radio is a proven and popular format regardless of the topic and market size.

**By Chriss Scherer, editor**

Serve the public interest through information exchange. This is what broadcasting is supposed to be all about. One format that exchanges more information than any other is talk radio. It is the oldest of formats and the way to get the job done has changed significantly over time. It's not just the farm reports on the daytimer, but the format covers signals everywhere with local and syndicated programming.

Talk radio thrives on news and controversy. When big news is happening, the talk stations draw the listeners in. Big elections are one of the more common draws, although more recently other items of war, scandal and crime have dominated the talk scene. On a nationwide average, almost 15% of all stations carry talk radio as their format.

### How it 'works'

What does it take to get a talk show on the air? Some studio equipment is common to music and talk formats, but heavier emphasis is naturally placed on higher-quality, better-sounding equipment required for the particular needs of talk radio.

The telephone caller is the best friend of any talk format. Because of this, the on-air telephone system has been given much attention. There are several elements that make up a quality integrated telephone system, and what works well in the office is generally terrible for on-air use. The two main components of an on-air telephone system are the *hybrid* and *interface*.

The interface used is based on several decisions that must be made by station personnel. How many lines will you use? What kind and size of expansion, if any, do you want? Will you want other studios to share the capability or will this be for only one room? These are just a few of the questions to ask when determining the size and type of system to use.

The 1A2 telephone system is one of the oldest multiline telephone system technologies around. It is a mechanical system that routes the raw telephone tip and ring through the facility. Once a controller is installed, additions and changes are not really that difficult. The use of multiconductor cable makes cabling a headache, but adding telephones or assigning lines is easily accomplished in the phone room. Once a popular system for offices, the 1A2 system has fallen from popularity for more recent digital telephone systems.

The faithful 1A2 telephone set.

The inherent advantage to using a 1A2 system is the theoretically unlimited number of lines that it can handle. The main frame of the system is limited, but more frames can be added as needed, even with separate music on hold feeds. There are many interfaces built to work as add-ons to the 1A2. These also have the expansion capability to support a great number of telephone lines. Interfaces made for 1A2 installations look like a telephone set to the KSU, which makes changes or additions much simpler.

As the popularity of 1A2 systems decreased for business use, other solutions were introduced. Proprietary controllers were developed that handled the telephone traffic as easily as a 1A2, but with cleaner, quieter and more reliable electronics. These systems did not interface with one another well, if at all, and were limited in their flexibility. The basic idea of those early systems is still used today, but with added features that make them more functional.

The all-in-one solutions range from a few lines and include an internal hybrid, to larger units that are strictly interfaces that feed a hybrid. Again, the choice is made depending on the current and future needs.

So, who has the largest on-air telephone system in use? That is tough to say. Some of the major nationally syndicated shows use systems that handle more than 30 lines. Of course, the systems used on these nationally syndicated programs are not the average, but they do give you an idea as to how large a system can become.

The telephone hybrid has the unique function of splitting a two-wire telephone line into separate send and receive audio lines. This has been done for many years in many forms and some are good at it, while others are not. It is possible to do this function with analog equipment, and there are several hybrids available that do it this way. The next step in caller/host isolation is to use DSP technology. More and more digital hybrids are used every day. The added advantages of automatic nulling during the call and being able to compensate for changes in the telephone line make digital hybrids the cream of the crop.

The call screener position is sometimes located in the control room.

The audio transmitted over a telephone line is limited to a frequency range of 300Hz to 3,000Hz. Unfortunately, this audio range rarely (if


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ever) presents a "flat" frequency response varying by 2dB or less. An adaptive hybrid will help to keep the null level in check, but equalization will also help in smoothing out the overall sound of each caller for a more consistent sound. Also, because no two telephones and no two callers are alike, some kind of audio compressor or *automatic gain control* (AGC) should be used to compensate for the differences in level. Some of the more advanced hybrids will do all of this for you. In some cases, this may need to be handled externally from the hybrid.

## Who's calling?

Now that you have callers and a way to handle their audio, there has to be a way for the show host and producer to communicate. Although there are many telephone systems available, there are just as many call-screening programs. Many of these screening programs interface directly with the on-air telephone system for simple routing and information handling.

Having immediate display information about the caller for the host to see is essential. The caller's name, location and topic of interest are the obvious items of interest. With the screener program, however, comes the advantage of creating a caller database instantly. How you use the database is entirely up to you and the marketing department, but additional information like telephone numbers, addresses, regular topics of interest, birthday, zip code and occupation can make the database even more useful for other station uses. If for no other reason, the next time that listener calls in, his or her information is already there to pass on to the host.


The host should have a clear view of the call-screener monitor, guests and the control room.

Caller ID is a service that is available almost everywhere. Several manufacturers have looked at implementing the Caller ID data into their call-screening software for even more flexibility. Before the call is even answered, the screener knows who it is, when he called last, how often he calls and all the other information stored in the database.

Advances in digital telephony have already been realized with ISDN. Remotes that were once noisy and

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intermittent now enjoy clean audio with ISDN service and codecs. Some manufacturers are looking ahead to the household use of ISDN for high-quality voice calling by planning on designs for ISDN hybrids. The use of multiplexed telephone service (where the incoming lines are all fed on a single pair instead of discrete wiring) will lend itself well to these plans.

Many stations use a digital PBX for the office telephone services and then route the calls to analog single-line ports for on-air use. Although this has cost advantages, most analog line cards were not built for the highest quality of audio, but rather

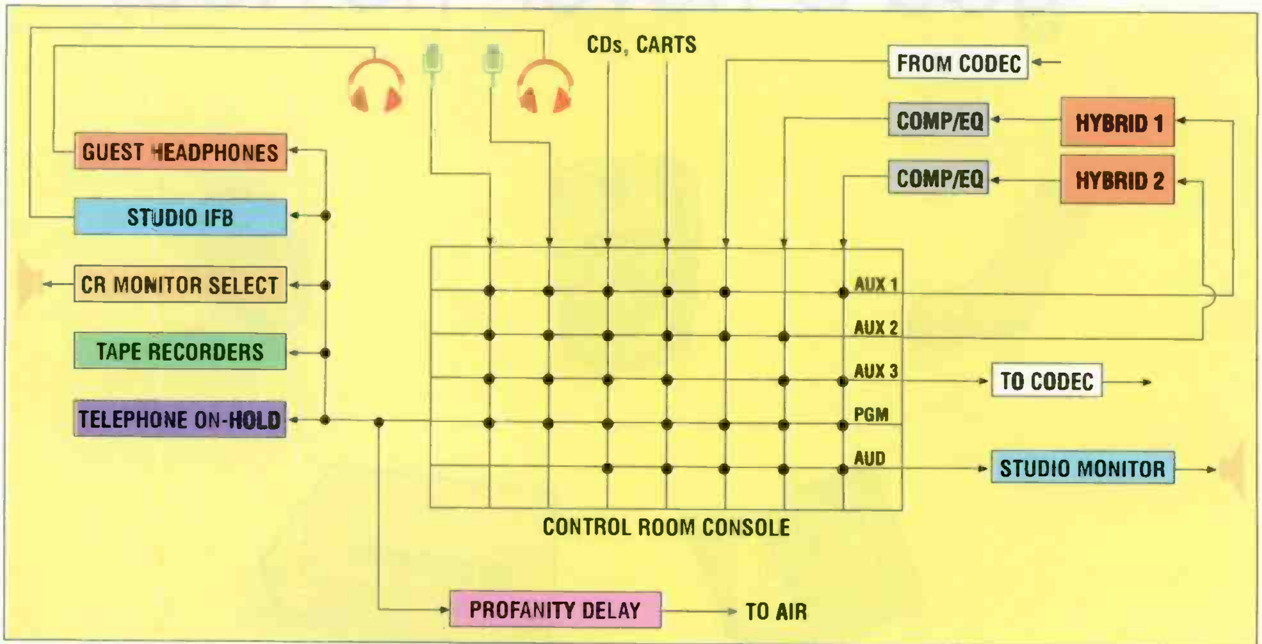



Figure 1. Typical audio signal flow for a talk-radio facility.

routing of fax or modem data, because most modern PBX equipment is digital. The savings on equipment cost may not be worth the loss in audio quality by routing through this low-quality audio equipment.

In some larger applications, basic intercom systems between the host, producer, screener and engineer will not work. Off-site guests, multiple hosts or even the operational format of the program may require additional attention to communications behind the scenes. In situations like this only some of the people involved may need to convey an idea without distracting everyone. There are a variety of intercom and routing devices that make this task much easier to handle.

Regardless of the method, any IFB or intercom system must be clean enough to be understood. The fanciest switching is worthless if you cannot understand what is being said. The intercom microphone and monitor must be able to faithfully reproduce the audio.

Another technical consideration may be the need for multiple mix-minus feeds. You already create one mix-minus feed for the telephone hybrid. With off-site guests or hosts, additional mix-minus capability may be necessary. With most consoles having only one, or possibly two, additional audio buses, you can run out of space quickly. There are ways around this though. Some interface products are available that can create the added mix-minus capability that you need.

## Technology advances

As was mentioned before, the low cost and easy availability of ISDN has made it easy to create a good-sounding remote quickly and easily. When RF remote pickup equipment is plagued with noise and interference, the ISDN codec is clear and clean. Taking a talk show on the road and having to suffer with the noise of a poor RF path or the costs of POTS multiline frequency extenders is rarely worth the effort and the chance of listener tune-out grows.

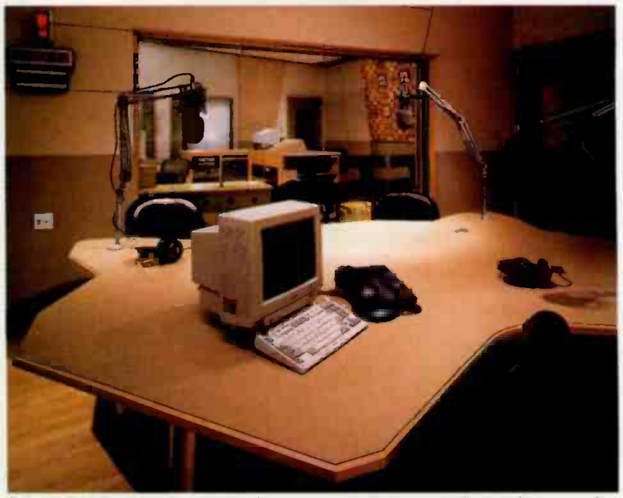
The ISDN codec technology has migrated to plain old telephone service (POTS) lines as well. Take the simplicity of frequency extenders and the quality of digital and get the benefits of both. Although not as wideband as their ISDN cousins, POTS

Studio windows help to open up a room.

codecs are another option, especially if ISDN service is not available.

Both of these methods give talk shows an advantage on the road. Within the datastream that carries the audio signal, an additional data path is usually present. The auxiliary path can be used to link the remote host to the station for call-screening information, telephone caller control or even just simple switching and tallies.

No matter how you get your remote audio back to the



Any talk studio should be large enough to comfortably seat the host and guests.

studio, the key to its success will be a familiar and easy-to-use environment for the talent. Any remote broadcast presents a challenge, but with a talk show, there is even more to keep in check.

### *Bleep* this

Running a live talk show without a profanity delay is like playing Russian roulette. Someone is going to get it eventually. When the budget is tight, this may seem to be an unnecessary cost, but the price of an inexpensive audio delay is certainly much cheaper than a fine from the FCC. Even if you don't get fined, the time and effort to please an offended listener makes it a wise investment.

Profanity delays originally were magnetic tape loops that delayed the audio by a short time factor. The delay time was determined by the tape speed and the distance between the record and playback heads. The biggest problem with these systems was the tape wear or, even worse, tape breakage, especially in the middle of a show. The next advance came with solid-state versions. The early units suffered from low audio bandwidth and you still had the problem timing to get in and out of delay. Having to "bleep" an offensive word was even more tricky, with the engineer having to switch over to the delayed signal and manually catch the offending remark or play an audio piece to fill the time until it was safe again.

DSP technology again came and added great flexibility to profanity delay designs. The ability to get in and out of delay at the push of a button makes the transition from live to delay and back a simple one. Adding delay time

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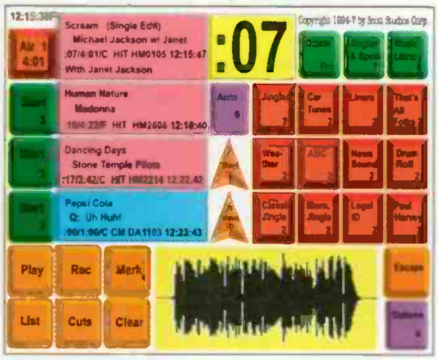

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slowly and inaudibly allows the program engineer more time to focus on the show being produced. When an objectionable word or phrase comes along, pressing *dump* clears the audio memory and the delay time begins to rebuild once again. A recent addition to these digital delays is the ability to dump only a portion of the audio memory, giving hosts the ability to get out of several consecutive on-air offenses.

## Monitoring

A clean monitor system is critical. The host needs to be able to hear the callers and all the program material. As was mentioned before, sometimes basic IFB systems will not provide enough flexibility for the program being produced. More elaborate schemes may be required if there is more behind-the-scenes work going on.

If telephone calls are an integral part of the program, you may consider having a separate telephone monitor setup in the studio. It is an added convenience that will give the host some physical flexibility by not always being tied down to a headset. The telephone monitor should be a direct feed from the output of the hybrid with no other program audio in it to avoid feedback. The added advantage of a telephone monitor is for in-studio guests.

When people that are not accustomed to talking into a microphone are asked to do so, they tend to speak

**Convenient placement of microphone and headphone level controls is an important design goal.**

softly and not focus their delivery. This is normal. Even though we all speak every day, being put in the spotlight, especially where the spoken voice is the highlight, can be a shock. When you take this same person and have him or her wear headphones, the distraction can be even greater with the concentration of sound directly into their ears. If they are in the same room with the host and the telephone monitor, they can be made to feel more at ease by holding a regular conversation without the constant reminder of being on the air.

If the guests must wear headphones, it is advisable to provide a monitor mix that contains no IFB. This is a distraction to many people. Most professionals can conduct an interview or discussion and monitor cues without skipping a beat, but for a guest to

*Continued on page 40*

## What is mix-minus?

The idea of a mix-minus feed is confusing to many people, especially when it is first introduced. In most applications, a program audio mix is fed for monitoring. This works well in cases where all the monitoring positions do not suffer from any time delays or other unique circumstances. The term mix-minus is a shorthand description of what it really means: a program mix, minus some specific element.

The most common use of mix-minus is feeding a monitor signal back down a phone line to a caller. If the caller's audio was sent back down the telephone line, a feedback loop would be created. This mix-minus can be created in a variety of ways. The most common way is by creating a separate mix within the console using an auxiliary send (aux.send) on the audio sources that are needed to be heard by the caller. By using a *pre-fader* send, the telephone monitoring system can be used off-line from the on-air setup, which most jocks and show hosts find convenient for hands-free use.

Another common way is to use a second program bus (audition or utility) to feed the phone line and de-select the sources that caller need not receive. This method has the advantage of allowing on-air level changes to be heard by the caller. This setup does not work well for off-air applications because of the switching required that usually leads to mistakes that are heard on-air.

Most newer consoles have provisions for telephone modules that enhance the function of the telephone mix-minus by switching the feed to take specific advantage of on-air or off-air use. When the telephone module is turned ON, the phones are fed from a mix-minus of the

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program chain, when the telephone module is OFF, the phone is fed from an alternate mix. Manufacturers may refer to this as an *off-line* or *telco* mix.

When an older console is the heart of the studio, the additional buses and aux sends may not be available to create the mix-minus. It is sometimes necessary to create the mix-minus with a differential amplifier. By taking a post-fader feed from the telephone level control, it can be inverted and then mixed to the program feed to subtract it from the mix. Adjusting the level for on-air use automatically adjusts the level to the differential amplifier, constantly nulling the telephone audio from the program mix. This is not as reliable as the other methods mentioned, because any phase shift of the signal will affect the quality of the final mix-minus.

There are telephone hybrids that can perform this function internally, as well. By taking a program mix for the input, it takes the incoming caller audio and does the differential mixing internally. This simplifies installation and the circuitry involved is advanced enough to do the job well.

The creation of these alternate monitor mixes lends itself well to the addition of IFB and other cue audio as part of the monitor feed. Because an alternate mix already exists, adding an audio source for cuing and other information can be easily accomplished.

The use of multiple mix-minus feeds has become more common with the increased use of codecs. Because most of these rely on some kind of data compression to squeeze the data into the reduced bandwidth of the ISDN or POTS line, there is an accompanying time delay through the system. This time delay can range from a few milliseconds in limited bandwidth mono applications to almost half a second for full-bandwidth stereo configurations. Monitoring yourself with a time delay of more than a few milliseconds is confusing. Typically, 35ms is the threshold of a perceived delay, called *Haas effect*. In these applications, a mix-minus feed to the remote site is crucial for a successful remote.

*Continued from page 38*

hear cues and other info can destroy the conversation and the listener will never know what happened.

### In our audience. . .

What about a studio audience? Having a studio audience presents a plethora of new issues and problems to tackle. It sounds like it would be easy to do, but there are plenty of considerations.

If the audience is more than a few people, having each person wear headphones for monitoring becomes impractical. When the audience members are going to be an active part of the program, microphone placement is the first critical issue. The physical layout of the room will determine how you will handle people moving around to the microphone. Find a suitable location that is easy to get to, but also isolated from other noise sources in the room.

Because headphones are not practical, there must be an open monitor or monitors in the room for the audience to hear. The source feeding these monitors cannot contain the microphone audio present in the room to eliminate

the possibility of feedback. Here is another application for a separate audio mix. The monitor should be run just loud enough for people to hear without letting too much of the audio leak into the microphone. The key to success here is the placement of the monitors compared to the microphone. Use the directional characteristics of the microphone to your advantage.

Use of a cardioid microphone will help with isolation, but be careful. If the mic pattern is too tight, you may end up losing the audience member's voice if he or she moves too far to the side.

If studio feedback is still a problem, there are feedback reducers that can automatically filter out the harmful frequencies. These operate by inserting deep, narrow-Q filters into the audio path as needed, as determined by the processor's circuitry. These products offer a range of capabilities. The basic units have only a few frequency bands that are set and remain static, while the more complex (and expensive) units have multiple bands up to 10 or more that are inserted as needed. The control circuitry can also be smart enough to sense the presence of the offending frequencies, filter them, and then after a period of time, reset the filters if necessary and re-assign them as needed.

## Audio processing

The main processing for the station is optimized for the overall program content. With so many other audio sources being pre-processed before they are put on the air, you need to allow for similar treatment to the mics. A microphone processor is optimized to handle the properties of a voice to give it a smooth, consistent sound. When properly set, it is possible to reduce the excessive peaks that are normal in the human voice and give a more consistent overall level for use in the program mix.

When a mic processor is improperly set, the resulting sound can have an interesting quality that may be suitable for an occasional effect, but an overcompressed, overequalized voice has a greater chance of becoming fatiguing to the listener.

Mic processors usually have a *downward expander* or *gate* built into them. This can eliminate some extraneous noise or assist in isolating open mics from multiple pickup paths. The level of the gate must be carefully set so as to not cutoff at too high of a threshold. When the threshold is too high, the audio will mute abruptly at the end of each sound, making it sound artificial. If it is set even higher, it can affect the audio, cutting off the person's voice in mid-sentence.

If you are using a profanity delay (which is always a good idea), you are going to have to monitor from a point in the program line before the delay. This audio feed will be routed to your control room monitor selector and any other device that needs a

real-time signal. This audio feed may need to have some kind of audio processing, because most on-air personalities are accustomed to a processed monitor feed off the air. The audio feed taken after the processing but before the profanity delay is often referred to as *pre-delay* and can be routed throughout the facility for use as a monitor feed.

## Room layout

The concepts behind studio design can fill an entire book, but some basic ideas should be kept in mind. The room should be large enough to be a comfortable work environment. Because of the constant open microphones, isolation from outside noise sources needs to be addressed. HVAC, traffic and office noise that may be permissible when the mic is open occasionally will quickly become annoying to the listener over time.

The sound of the room is much more important because of the open microphones. There is an economical compromise that must be made concerning acoustic room treatment. Going to extremes for sound absorption is seldom worth the expense. A reasonable compromise can be determined rather quickly. The simplest way is to listen to the room through the microphones over monitors. The most troublesome problems will present themselves quickly. Listen for any coloration of the sound from room reverberations or unusual frequency response from standing waves. Be sure to listen to a variety of sources. Different voices have different frequency content, and sometimes, when one voice sounds great, another might not.

Visibility is also crucial to the room layout. Because radio relies on audio to convey the idea and meaning, the visual cues behind the scenes are an added way to communicate. Clear view from the host to the engineer or producer can greatly enhance the flow of the program. This often puts the guest's back toward the control room, which also helps eliminate some of the visual distractions that can interrupt the thoughts of the guest.

## Putting it all together

Producing a talk-radio program or format may not seem as cost-effective in the continuing trend of down-sizing and consolidation. A successful talk-radio program will need at least two people, but is more commonly done with three or four. Any show needs a host. The other positions are an engineer, call screener and producer. These four duties can be combined, but certainly add to the work load.

Whether the topics are broad or specific, sports or politics or just a little bit of everything, the popularity of talk radio is as strong as ever. With more nationally syndicated programs available, and even more talk-radio programs or formats showing up on FM stations, it is obvious that a good talk show can keep your listeners tuning in. The right equipment and the right people can make it all happen.

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# Audio processing


## The debate of signature sounds, loudness on the dial and clean audio goes on. Understanding how to achieve your goal helps ease the conflict.

**By Jeff Keith, CPBE**

*Audio processing.* This pair of seemingly innocent words has the power to divide a radio station, especially the programming and engineering departments. It is the one piece of technical equipment that also has the power to shape how your radio station sounds on the radio dial next to its competitors. When setting up the station's audio processor, we quickly discover that its tremendous powers have the ability to either enhance or destroy the sound of our station. Unfortunately, these two possible outcomes are usually not far from each other on the control settings, and this is where the fun begins!

The "adjuster" of the audio processing shapes the sound of the radio station just as the

Photo courtesy of Brian Chalmers.

# Audio processing

artist placing paint on the canvas creates a picture. Depending on the skills and goals of the two parties in these scenarios, the outcomes can be widely different. The seemingly random placement of paint on the canvas by a skilled artist can create an abstract work of art worth millions. However, setting up an audio processor with random control settings will probably not result in an on-air sound worth millions. The result of doing that will likely be quite the opposite.

The adjustment of on-air audio processing requires a thorough understanding of the product being adjusted. You also have to know what sound you're trying to achieve, and it doesn't hurt to have a good feel for the station's programming. You also need a detailed and recent knowledge of where the other stations are in the market. Nothing can be so frustrating as to spend several unsuccessful hours adjusting your new "Tornado 2000" processor, trying to "catch up" to your competitor across town, only to find out that they know what you're doing and are temporarily overmodulating to throw you off track. This tactic has been reported by several engineers I know. You shouldn't drive your car without using the mirrors, and it doesn't hurt to look around the corner with your modulation monitor to see if everyone else is "behaving" while you play.

## Research, research . . . research?

Radio stations spend tens of thousands of dollars annually on music and audience research. They have a good idea where they are at any point in time and what the audience needs in the way of program content. Then they spend tens or hundreds of thousands of dollars subscribing to a rating service to tell them how they're doing. We do these things to try to measure our audience to make sure it gets what it wants, but do we ever ask them about their impressions about how they think our station sounds? You could argue that many listeners might not care or be able to tell, but what about the rest? What about our target audience? Do we set our processing to please the tastes of our audience and our announcers or do we just set it to please our own egos? Depending on the political structure in your station, this can be a tough call.

The station's programming department is charged with the goal of creating ratings. The station must attract as many potential listeners as possible, but then it needs to keep them. From a technical standpoint, that usually means setting the processing so that listeners' first impression of the station is one of "power," "authority" or "bigness." You want listeners to feel that the station has the power to deliver what they want.

# Audio processing

The first impression listeners might perceive when changing stations is a change in loudness (unless, of course, you happen to be playing their favorite song). Hopefully, your station is as loud or louder than the station the listener is tuning away from, because this is likely where that first impression of power is made.

If we pursue this goal too far, we quickly get into trouble. The goal of raw loudness, without other considerations, creates many possible fatigue factors. Your listeners might find that they're mysteriously "getting tired" of listening to your station and they might not know why. The never-ending pursuit of absolute loudness is commonly blamed on the downfall of AM, and whether that judgement is right or wrong, from a technical standpoint, it is possible and we have to consider it. In any particular market there will always be that one station that is the loudest on the dial. It might be yours, then again, it might not. Does the audience care? Let sanity prevail.

When we look at how present-day audio processing products evolved from those simple boxes of 30 or 40 years ago, we see how the trend toward ever-increasing loudness levels hasn't slowed down a bit. Now, with digital signal processing, we are able to write code to do functions that would be complex or impossible in the old analog world. We still have to be able to write the proper code and that's where some problems exist in present-day digital processing products. Great progress is being made as audio processor designers are finally learning how to define their dreams in software. Some say we've finally reached the physical limits of how loud a station can be. But, since we won't know unless we try, we keep searching for that extra decibel of loudness.

Processing designers have made great strides in matching the dynamics of their gain control structures to what our ears and brain expect. Much like the bouncing ball above the words on a Karaoke machine, the processor's gain has to follow the signal dynamics in a way that please our ear. If the designers get this timing (and the height of the "ball") just right, instead of pumping and other artifacts, the music "just fills in." Much of this progress is directly attributable to advanced studies in psychoacoustics, which is the study of why and how humans hear what we do.

The designers of audio processing have also studied the various types of audio distortion created by their individual designs and have attempted to model their circuits to either minimize or mask things that the masses might find objectionable. They make trade-offs in their designs, just like we make trade-offs in the adjustments after we install their product into the equipment rack and tweak it. Let's see how we got into this loudness predicament in the first place and why it can be so technically and politically complicated.

## Audio processing: the (once simple) past

About 30 or 40 years ago, audio processors were employed in broadcasting simply as protection devices.

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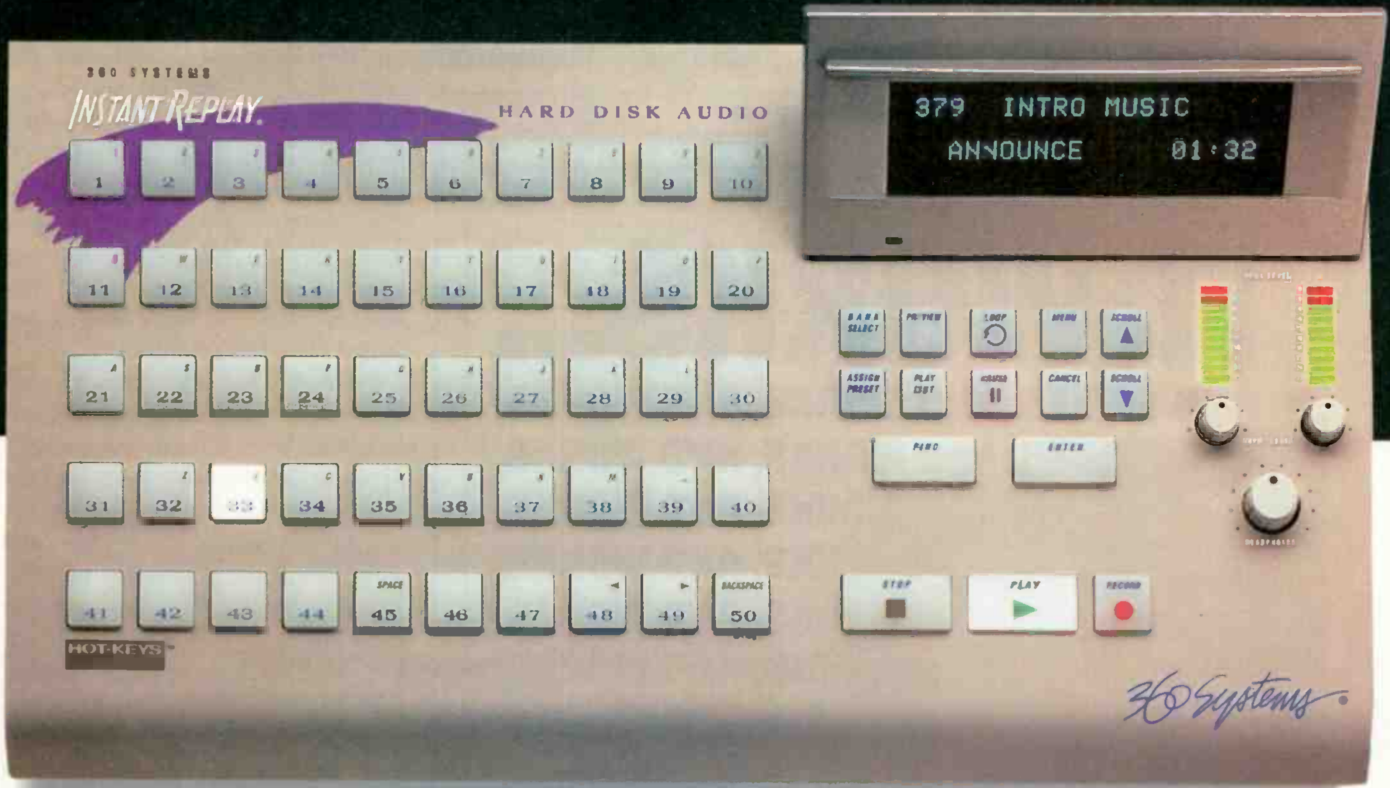


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# Audio processing

The early products were simple audio compressors and limiters, and they were used only to ensure compliance with FCC regulations by preventing overmodulation. At that point in time, audio processing was in its "purist form," and as little as possible was used so as not to disturb the artistic intent of the program material. The compressors usually had gentle characteristics and they were merely used to average out gross long-term errors in program level.

**Figure 1. A typical single-band audio processing chain.**



This also made the job easier for the limiter, preventing it from having to deal with widely varying input signal levels.

Someone then discovered that by pushing those devices just a little harder, denser modulation of the carrier resulted. In AM, the primary medium of the day, the signal was a little louder due to the resulting higher average sideband power and this resulted in small, but usable increases in coverage area. It wasn't long before compressors and limiters were being used well beyond their original intent, and for competitive reasons, this trend continues today in a seemingly endless upward escalation of signal density and audio processor complexity.

Although there were many variants before it, the most widely remembered early broadcast "processors" seem to be the CBS Audimax and Volumax pair. Not to discount the myriad of products before them, but these two boxes performed the compression/leveling and limiting functions probably better than anything before. The first Audimax (which was the model RIIZ, I believe) was a vacuum tube design. It performed intelligent leveling of the program audio and when set correctly, it virtually erased long-term program level variations coming out of the studio. The 400 series Volumax was a solid-state limiter/clipper, albeit with germanium transistors. This was probably the most popular single manufacturer audio chain that I can remember from the '60s. It was common to find these boxes teamed up with a Kahn Symmetra Peak, a passive phase scrambler. The combination worked well and was simple to set up. These boxes remained in use throughout the changes in AM rules, which went from no limit on positive AM modulation, to the later and more practical +125% limit.

## Two heads (I mean processors) are better than one

It wasn't long after the CBS products arrived that engineers began to make modifications to them to make


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them more competitive. Soon, the limitations and audible artifacts of broadband processing were realized and engineers around the country began looking for a better way. They began experimenting and borrowing some technology used in high-quality loudspeaker design. They began to gather multiple processors together and feed them with home-brew band splitting and combining boxes. The first incarnations of today's multiband processing were born, although they were crude by today's standards. I remember my first exposure to this technique in a conversation with Ed Buterbaugh, then chief engineer at CKLW. It was 1975 and I could no longer resist the temptation to call him to find out how the "Big Eight" got that great sound.

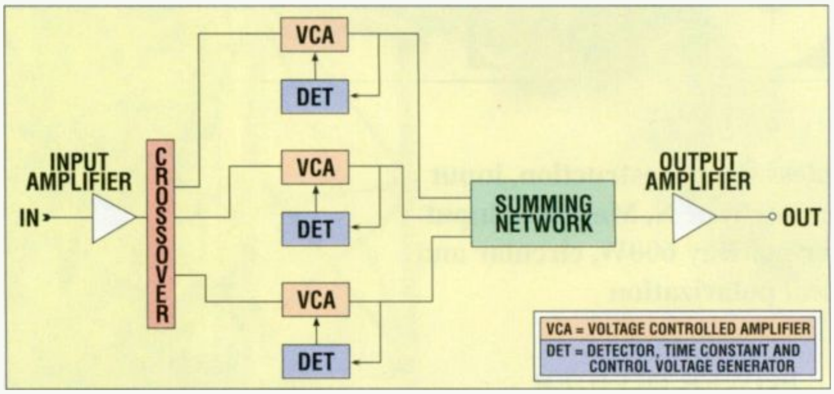

**Figure 2. A basic multiband audio processor.**

About the same time, the UREI BL-40 Modulimiter arrived. The BL-40 contained a broadband rms compressor followed by a limiter, and if you carefully study its schematic diagram, you'll notice that the BL-40 is essentially a UREI (Teletronix/United Audio) LA-3 compressor and a UREI 1176 limiter in one box, with some refinements for AM. The LA-3 is a solid-state descendent of the famous Teletronix LA-2 tube-based compressor. The popularity of the LA-2 compressor and the 1176 limiter still prevails today in recording studios worldwide, which speaks volumes about the quality and design philosophy of these two products.

Soon, Mike Dorrough marketed his DAP 310. It was the first commercially available multiband broadcast audio processor and Mike told me that it was conceived and built on his kitchen table. The DAP was a tri-band design, with gentle 6dB per octave crossovers. Each of the three bands performed limiter and expander functions. The three bands were then summed back together and passed through a rather crude but effective broadband clipper. The unit worked well and usually out-

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performed most of the home-brew attempts at multiband processing.

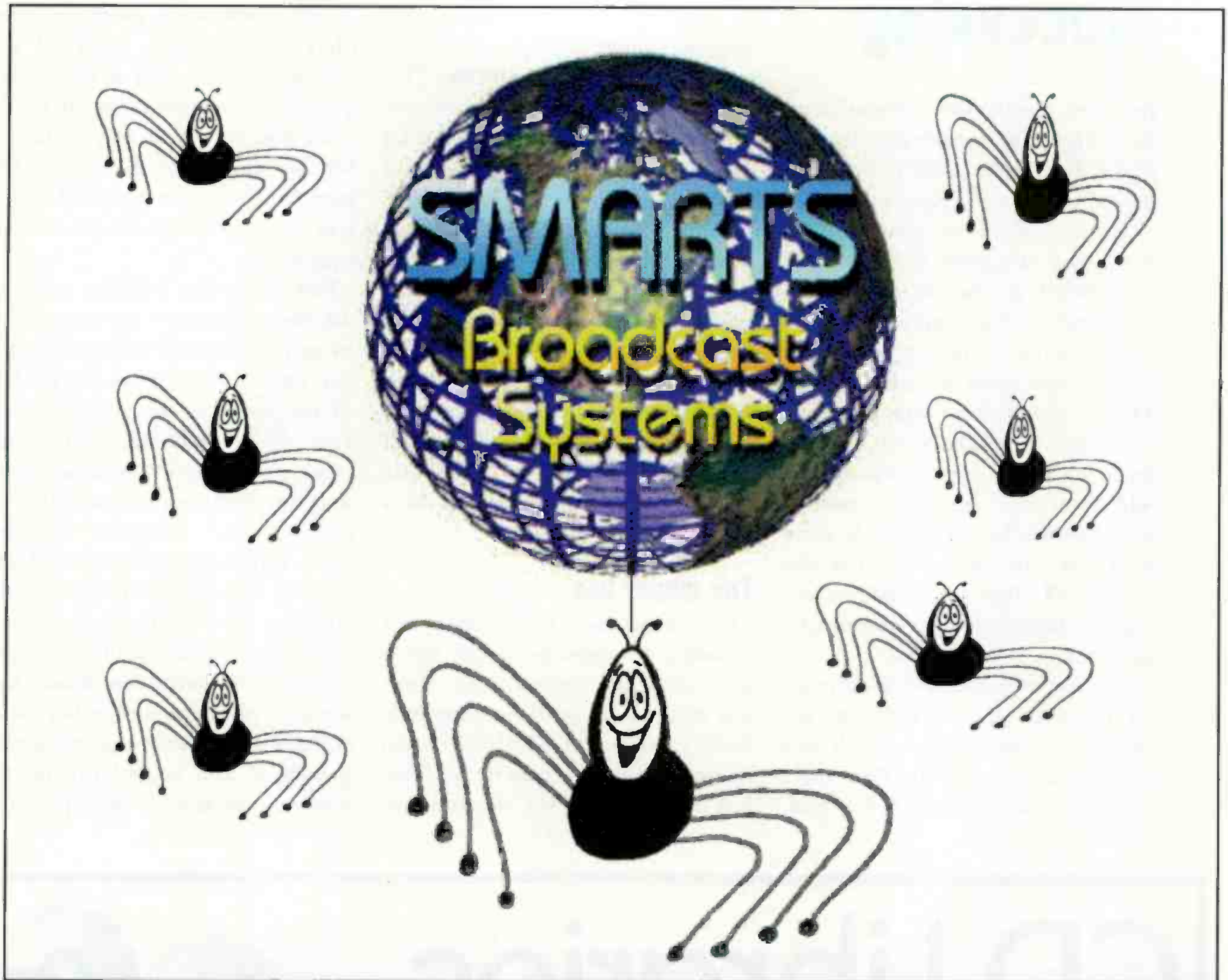
### Darn that disco bass

We've slid forward to the mid-1970s now, and there is a real flurry of processor design going on. Lots of new boxes are appearing on the market purporting to do all kinds of magical things to the audio. The Orban 9000A AM Optimod, with its six-band limiter section, is appearing in transmitter room racks around the country. Average modulation levels on AM are still climbing and processor complexity is ever increasing. The limits of AM seemingly being reached, program directors and station owners are now beginning to become more aware of that rarely touched sleeping giant in the back room, the FM transmitter.

It was really only a matter of time before highly modified AM audio processors started appearing in FM air chains in the never-ending battle for more loudness, coverage and revenue. FM, however, was a lot harder to tame. Due to the competitive modulation levels desired and that nasty 75$\mu$ FM pre-emphasis curve, engineers now had a real tiger by the tail. Strange and previously unheard processing artifacts were being generated, and modulation-robbing overshoots were preventing engineers from enjoying the same level of modulation density and accuracy that they had enjoyed with "decent" transmitters on the AM band. The problem arose of getting good loudness on the FM band; enter Bob Orban.

Orban is the first person generally recognized to discover and tackle the FM modulation overshoot problems. Through some rather astute observations, he found that even limiters with perfect output waveforms seemed to have overshoot problems when coupled to common FM stereo generators. The problem seemed to be rooted in the low-pass filters between the processor's final clipper and the stereo generator, but why was this happening?

Simply put, the audio emerging from most competitively driven clippers resembles square waves, ie., the audio can increase only up to a certain point (the 100% point) and then it abruptly stops. This "stop point" creates a flat top on the audio waveform. The resulting square waves contain energy that could interfere with the 19kHz stereo pilot signal if not removed. Therefore, complex filters are required prior to the stereo generator in order to pass program energy up to 15kHz, but protect the fragile 19kHz stereo pilot, which is injected a little more than 20dB below 100% modulation (less than 10% modulation). Unfortunately, these filters distort the time and phase relationships of signals passing through them while they perform their intended function of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# Audio processing

removing the higher harmonics of the clipped and perfectly limited signal. But, if the filter is doing its job, why do we still have a problem?

Mathematics tells us that removing the higher harmonics from a square wave (which is what the filter has to do) results in the amplitude of the square wave increasing over the level it had when it was "square." This is because the square wave's "squareness" results from it being made up of an infinite number of odd harmonics. Instead of making the amplitude higher, adding in more of those odd harmonics has the unexpected effect of actually reducing the peak value or making the square wave's top "flatter."

Likewise, removing these harmonics makes the square wave's top "not square" anymore. But, if it's not square, what shape is it? That varies a little by filter design, but overall, the waveform "grows" a sharp peak extending some distance above the rest of the top, which is still flat. This extra peak, depending on several variables, can reach well above an equivalent 150% modulation and contributes literally nothing to your station's loudness. Now, if we only have a 100% modulation limit to deal with, how do we make the station legal if these "overshoots" are occurring and reaching to 150%? You have to turn down the overall modulation until the peaks no longer exceed 100%. But, doing so obviously and significantly decreases the loudness of your station. There had to be a better way.

## The magic box

From experience, Orban knew that cascading FM processors with stereo generators and their built-in filters was riddled with unknown outcomes. Each combination of products had its own modulation penalty. So, what was his solution? He elegantly designed a new audio processor, complete with specially designed low overshoot filters and a new stereo generator. He put them all in the same box, with each stage optimized for and aligned to the others. The now famous Orban Optimod 8000A was born and the loudness wars began.

This single-box solution probably did more to improve the sound quality and loudness of FM stations than any other audio processing product of the day, and for a while it was king. Soon, multiband processing began to join the FM band, and many were installed before the 8000A to make it "more powerful." This became especially necessary and important when disco music hit the airwaves with its tremendous bass lines. Single-band processors just couldn't deal with this music well without pumping and other nasty artifacts, and there was a modulation penalty if you backed off on the processor to keep it from pumping.

# Audio processing

## Divide and conquer

When you read the early literature written by Bob Orban, you'll notice that he was strongly against all forms of multiband processing. He felt that it upset the natural spectral balance of music and he was correct. However, the realities of the market at the time were that music was coming out of recording studios "spectrally tilted" to extremes never before encountered in commercial music. Music was changing, recording techniques and equipment were improving, and that meant that bass energies were being mixed into modern music that gave current on-air audio processing of the day absolute fits.

The rescue attempts went in several different directions. Glen Clark, the founder of Texar, designed a four-band leveler that used carefully chosen crossovers and a digitally based control circuit for the gain control section. Clark used some common opto-isolators (similar to the ones used in the UREI LA-4 compressor) to create a simple and elegant variable audio attenuator. When these were coupled to his special control circuit, the unit could compress to reduce levels, expand to raise them or do nothing if the level was correct. That function alone made Clark's device unique, because it wasn't always chasing after itself like other products. The result was a significant gain in average level and frequency response consistency from song to song. Many of the artifacts and grunge that plagued prior simple compress/expand devices were absent in Clark's design. These boxes found themselves in front of all kinds of early processing, including the then popular Orban Optimod 8000A.

That took care of many of the problems coming from program material, but some felt that four bands were too many and Orban was one of them. Orban's solution was the introduction of the Optimod 8100A. The 8100A was designed to handle large amounts of bass, while leaving

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# Audio processing

the rest of the spectrum fairly intact and presentable. The unit accomplished this by splitting the audio into two bands — a master and a bass band. The master band handled energy above 200Hz and the bass band handled energy below 200Hz. Realizing that compression alone couldn't handle large amounts of

Finding your signature sound may require some testing and comparison. Photo courtesy of Brian Chalmers.

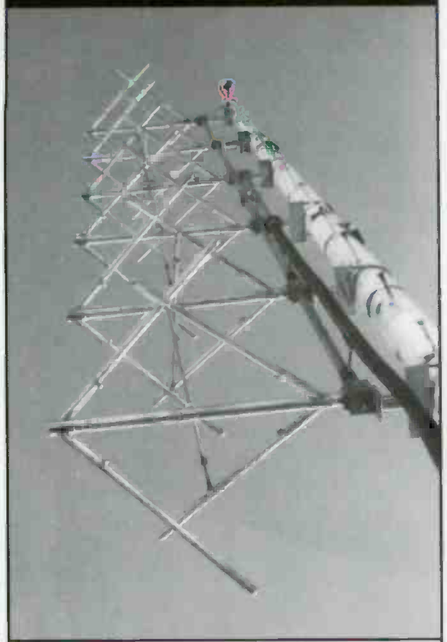
bass without pumping, Orban included a clipper in the bass circuit to take care of the momentary overshoots caused by the necessarily slow-acting bass compressor. A simple low-pass filter was included after the bass clipper to remove undesirable clipping distortion. In addition, in order to keep the bass compressor from straying too far from the compression value of the master band, Orban slaved the bass compressor to the master band so that it could follow it to a user-selectable degree, but still independently handle heavy bass when called on to do so. These circuits were so unique that Orban was awarded patents on them.

Orban still had the FM pre-emphasis curve, peak clipping, filtering and stereo generation functions to contend with. His 8100A used a new and patented approach to controlling pre-emphasized audio and could generate extreme amounts of loudness without generating objectionable distortion. For the first time, we heard terms like *distortion cancellation* when reading audio processing specification sheets. Orban's high-frequency limiter was also unique.

Rather than reduce the peak level of the program every time high-frequency material was present, his design reduced just the problematic high frequencies. This now meant that a processor could deliver competitive loudness without the usual penalties. It was a major improvement over traditional methods. However, instead of a loss of loudness during high-frequency material as in past products, this method caused a loss of brightness. Now we had a new problem — loud, but dull-sounding radio stations. Program directors wanted this fixed. They wanted more brightness, but they didn't want to lose the loudness. It was a kind of "have your cake and eat it too" scenario.

## A science lesson can be a good thing

The FM pre-emphasis curve is the enemy here. It is a standardized equalization boost curve used by all (U.S.) FM stations and it has a complementary roll-off in FM receivers. The main purpose of this pre-emphasis/de-emphasis trick is to reduce noise





# Audio processing

(hiss) in the receiver. To reduce noise, the frequency response in the receiver is purposely rolled off according to the 75μ de-emphasis response, which is down a whopping 17.07dB at 15kHz. But, in order to restore flat response to the listener's receiver, a complementary 17.07dB boost is required during transmission and therein lies the problem.

probably inappropriate for most modern program material, several new multiband limiter designs began appearing on the market. Most were stand-alone products, but surprisingly, even Orban bowed to market pressure and introduced the 8100A/XT, a new six-band limiter accessory for the Optimod 8100A. It's amazing what a little market pressure can do.

Occasionally, you would hear that other products existed that could deliver competitive loudness and good

The Orban 8200 is a multiband digital processor.

When the FM system was first proposed and put into practice, high average modulation levels were not an issue. No one thought about how difficult it could be to accurately control high-frequency modulation if the average levels were increased and audio clipping (on purpose) was nearly unheard of. No one ever thought that in 50 years, we'd be trying to shove a tornado through a keyhole. And, it should be obvious by now that we can't just simply stop using this problematic pre-emphasis on the transmitter to alleviate the problem. You can only imagine how many receivers you'd need to modify in order to make that idea work.

## Invent, invent, invent

In the mid-'80s, we were surrounded by the limitations of the laws of physics. It is said that necessity is the mother of invention, and it certainly was in this case. Realizing that a single high-frequency limiter was

high-frequency power on the air. There were Aphex Dominators, the CRL products, the Gentner Lazers and many others. And, there were also modified versions of all of those products on the air in some market somewhere. Every now and then, you would even hear of the elusive and exclusive Gregg Ogonowski boxes and what they could do. But we engineers were quietly wishing that we could somehow be saved from this modulation mess we'd gotten ourselves into. We can't just turn it all down and start over. Besides, who's going to volunteer to be first? I'll tell you what, I'll be a gentleman and let you be first, OK?

## The promise of digital

We would hear about the new emerging digital technology and all the promises that could be realized by the resulting digital hardware. For a few years, it seemed like all we did was read and dream about them. We read the promises, and we shared

the hopes and dreams of the manufacturers who were feverishly working in their labs to make these new products a reality. We hoped that some magical new process would be discovered in digital signal processing that could give, as one promise I read said, "digital audio processing will provide an additional 3dB of loudness." We engineers were skeptical of this promise and rightfully so. Anyone who understands audio processing knows that 1dB is a good increase in loudness when we're talking about highly processed audio. But 3dB? Wow. We could only wish. Shhhh . . . don't tell the PD.

Luckily, for audio processor manufacturers, two of the most difficult to do functions contained in on-air audio processors are now able to be done nearly perfectly in the digital domain. These are stereo pilot protection filtering and stereo generation. Both processes are difficult to do well in the analog domain because of physical obstacles like variations in parts tolerances and the like. Even the layout of the circuit board frequently becomes the limiting factor that prevents an analog designer from getting his creation to work to its theoretical limits.

Digital signal processing (DSP), on the other hand, is good at math. If the code is written properly, it does the same thing, the same way, every time. If the equations can be reduced to DSP code, nearly anything is possible and therein lies the challenge. How do we define those nebulous and mystical analog processes that we have refined over the last 50 years? The manufacturers of DSP-based audio processors are well on their way to doing this, and current-day products demonstrate just how close we can get to perfection if we spend enough time and money working toward it.

## Digital processors galore

While earlier products touted the word "digital," they were in reality, still basically analog signal paths under the control of a microproces-





# Audio processing

sor. In most cases, the microprocessor's job was simply to set and remember digital representations of what would be analog knob settings. This aided repeatability to a large degree, but the signal paths were still analog. Not that analog is bad, just that this digital technology wasn't truly "all digital" as some expected. The Cutting Edge Unity 2000i, the CRL Audio Signature and even the Texar Audio Prism had analog signal paths under some form of digital control. Again, not a bad idea, but not truly digital either.

Orban was the first to introduce a truly digital audio processor, the Optimod 8200. The most popular version, the U3S, combines a broadband AGC, five bands of limiting and a digital implementation of the famous and patented Orban multiband distortion canceled clippers. The stereo generator is a hybrid digital/analog design using digital switching techniques. This is surprising, because stereo generation, being a mathematical process, would seem to lend itself to be a DSP process of relative simplicity. But, the 8200 is a powerful product and is easy to use. There have been several small and incremental changes since its introduction, such as remote-control software, AES/EBU interfacing and some subtle internal refinements that have improved its sound over the early versions. There is also a budget version of the 8200 available that closely emulates the previous analog 8100A. Recently, the 9200 processor was introduced for AM stations wanting to go digital.

Now, the Orban 8200 digital processor has competition from two new, purely digital processing products that have emerged from Cutting Edge Technologies and CRL. Both products contain state-of-the-art, high-speed DSP chips, as well as current-generation A/D and D/A integrated circuits. There is also a trend in the newer products toward giving the end user more controls with which to tailor his station's sound. This act alone addresses one major complaint most users had about the earlier products — not enough knobs.

The Cutting Edge Omnia's internal structure is a broadband AGC, phase rotator, a four-band compressor and a four-band limiter, a final, distortion canceled limiter/clipper and a digital stereo generator. It contains a built-in composite clipper and an SCA protection filter. The crossover frequencies are also adjustable and there is a unique "make-up gain" function to smooth out abrupt changes in program level. The unit has a built-in remote software interface that supports most web browsers and two PCMCIA card slots are available for networking and software upgrade capability. Also, a selectable bass enhancement circuit allows stations striving for a clean and powerful bottom end "kick" to attain it. This feature fills a need not available from prior products. Cutting Edge is quick to warn though, that enough low-frequency power is available from the Omnia to possibly unlock an exciter's AFC loop. For a modern exciter, that takes a lot of bass.


**The CRL DP-100 features a touchscreen control panel.**

The CRL DP-100 is also all digital. Its signal path begins with a broadband AGC, a selectable phase rotator and five bands of compression. Following this are five bands of distortion-canceled limiting/clipping, a final clip-

# Audio processing

per and digital stereo generator. The action of the five-band compressor sections can be modified by an adjustable "dynamic gate," which analyzes the incoming program and adjusts the compressor's dynamics accordingly. Controls are available on the touchscreen interface for setting the desired equalization curve, as well as certain characteristics of the five-band compressors. AES/EBU is standard, as is a TOSLINK interface. Remote-control software has not yet been released for this product, but is under development. DSP code upgrades are made through a serial connection to a host PC running Windows 95.

Even in the face of all the digital processors, one company has taken a different approach. The Aphex model 2020 FM Pro is the processor for those who want advanced technology, but are still skeptical about the DSP-based digital audio processors, or for some reason, just don t like the sound they produce. We all tend to cling to analog technology because it is familiar and easy to understand. And, it does have infinite resolution, something digital will never have, no matter how many bits we throw at it.

The FM Pro is a combination of several patented or patent-pending circuits, which in the usual Aphex way, are unique circuits far superior to anything typically found in analog processing. Just as the Aphex Compellor and Dominator were unique and transparent sounding, so is the 2020. It starts out with a smart leveler to get control of audio levels from the studio. Following this are four bands of compression, with all kinds of adjustable parameters like crossover frequencies, drive and mix levels and release times.

After processing by the four compressors, audio flows to the optional split-band clipper and pre-emphasis limiter for final processing. The limiter stage features adjustments to allow the sound and feel of the bass frequencies to be set according to taste, as well as overall drive levels and control over the distributed pre-emphasis filter. The final limiter can then feed the exclusive (and optional) Aphex PPDM (parallel path digital modulation) stereo generator. The FM 2020 provides analog inputs and outputs, AES/EBU digital inputs and outputs (optional), automatic switchover to analog if the digital input fails and a realm of adjustments and features not commonly found on analog processors. There is even processing preset/daypart switching built in and a remote-control interface that supports both the Windows 3.x and 95 operating systems.

## Keep your goals in sight

In today's radio markets, not only are there many more stations than ever before, but sometimes more than one station is vying for the same audience. If two or more stations in the market are playing the same music, the same commercials and who knows what else, what delineates them from each other? The air talent is an important ingredient in defining the sound and feel of the station. But beyond those resources you have to rely on the technical tools at your fingertips to help shape the rest of the station's image.

Audio processing can help do just that. Given a processor with enough flexibility and power, nearly any on-air image can be created and that's why most engineers felt that early processors didn't have enough controls. If you can stray only a little from the settings the guy across town is using, how can you set those two stations apart? (We have to ask ourselves if the listener really notices these subtle things, but hey, that's a whole argument in itself.) The bottom line is that there are many tools now at our fingertips with which to help shape the sound of our stations. With other digital audio delivery systems for the consumers just around the corner, we must make certain that our listeners, and even the potential ones, are drawn to our station and captivated by what they find there. Creativity in not only the on-air product, but also the technical aspects of the station can set your station above the rest. Extreme attention to detail in your technical facility is also required in order to maintain audio quality, which is comparable to those new sources of programming.

But, whatever processor you decide to use or buy, before you tweak it, watch, listen and learn about the market and your competitors. Look over your shoulder a few times at the modulation monitor while you and the PD are defining the air sound. Don't get duped by the guy across town who knows you're trying out a new demo of the "Tornado 2000" and just wants to make sure that you send it back. Remember, we might all be in radio, but it's a war out there. There can be only one winner!

*Jeff Keith, CPBE, has been a broadcast engineer for more than 25 years and is the chief engineer of WMJI, Cleveland. He also holds a First Class NARTE certification with a Master RF endorsement.*

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# Coming in the November/December issue of BE Radio

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# Field Report

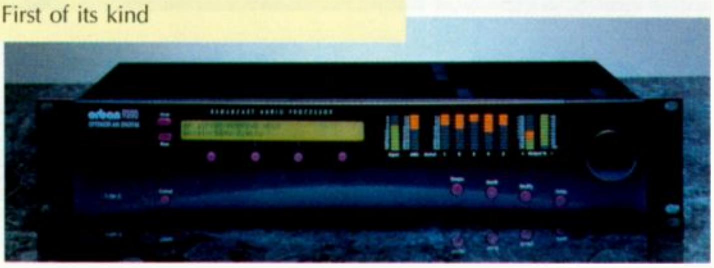
# Orban Optimod 9200

**By Paul Marks**

Orban revolutionized broadcast audio processing more than 20 years ago when it introduced the original Optimod. Now, the company has another first: the Optimod 9200, a DSP-based AM broadcast processor. Why add a digital AM processor to the product line? Perhaps to offer AM broadcasters many of the features that have been available to users of digital FM processors for the last several years. The significant features include simplified setup through factory preset settings and a LESS-MORE control; the flexibility of having many adjustable parameters for those seeking more control; the ability to store and recall user preset settings; and off-site adjustment with a PC and long-term stability without the thermal and component aging problems associated with analog circuitry.

## *Performance at a glance:*

- Designed for AM
- DSP-based processor
- Simplified adjustments
- High degree of flexibility
- Optional AES3 I/O
- First of its kind



All processing functions are accomplished with DSP technology. Its functions include adjustable high- and low-pass filters for the input audio and a broadband automatic gain control that provides a consistent audio level to the five-band limiter. A three-band receiver equalizer is included to improve the sound delivered to listeners by compensating for the typically limited bandwidth of AM receivers and also allows the station to customize its audio tonal balance. The five-band limiter and its distortion-canceling clippers make the station sound more consistent, increase density and provide peak control. An adjustable low-pass filter is provided ahead of the output to meet occupied bandwidth requirements, selectable for the country of operation. The transmitter equalizer section can be used to compensate for imperfections in transmitters and antenna systems that can limit the station's apparent loudness and coverage, with settings for day and night operation. One thing that the 9200 doesn't do is stereo. It is strictly a monaural processor.

The 9200 comes standard with an analog input and two analog outputs that use 18-bit A/D and D/A converters. Each of the analog outputs has its own level control and transmitter equalizer. The optional AES/EBU digital input and output have 20-bit resolution and support sample rates of 32kHz, 44.1kHz or 48kHz. The user can select the left channel, the right channel or a mono sum of the AES/EBU input signal as the source. Separate output level and transmitter equalizer controls are also provided for the digital output.

With the exception of the level controls for the analog outputs, adjustments to the 9200 are accomplished using soft keys and a control knob. An LCD screen display with large, easy-to-read characters provides labels for the soft keys and other information. Ten-segment LED displays provide metering of the input level, the gain reduction of the broadband AGC and for each band of the multiband limiter. Separate output displays are provided for positive and negative modulation.

### The back panel

In addition to the XLR input and output connectors, the rear panel has a ground lift switch to float the circuit ground from the chassis for connection to an unbalanced input. The DB25 connector allows remote control of up to eight different functions, including preset recall, day and night mode and analog or digital input selection. An RS-232 serial interface permits control using the included Windows (3.1 or 95) software directly or through a modem for off-site control.

Removing the top cover reveals the main circuit board with its eight Motorola DSP56004 chips. Surface-mount devices are used extensively and, consequently, many of the ICs are not socketed. A jumper on the main circuit board permits the user to change the input impedance from 3.6kΩ (as shipped) to 600Ω. Another set of jumpers changes the input sensitivity.

Some measurements were made at the transmitter site test bench with the 9200 in the bypass mode. Frequency response and distortion measurements were within their specs. Noise was measured at -74.8dB with a 22Hz to 22kHz measurement bandwidth. The published spec is -75dB from 50Hz to 20kHz. The RFI filtering of the input and outputs appeared to work perfectly since no sign of RF ingress was detected. Using tone bursts, the throughput delay of the 9200 in the operate mode was measured to be about 5.5ms.

## Going on the air

Setting up the 9200 is a straightforward task. When following the basic system setup section in the manual, the user begins to get a sense of the flexibility that the 9200 offers. Independently adjustable high-pass and low-pass filters for day and night modes set the system bandwidth. User presets can be used to reduce the system bandwidth if desired. The manual discusses the benefits of operating with reduced bandwidth under certain conditions. Adjusting the transmitter equalizer controls (low-frequency gain and break point and high-frequency shelf and delay) requires observing an RF sample with an oscilloscope while the transmitter is fed by the 9200 with its internal square wave generator active. It is possible to defeat the 9200's internal AGC when an outboard AGC is used. A positive peak control adjusts the level of positive peaks for asymmetrical modulation.

General-purpose medium, general-purpose heavy, music medium, music heavy, news, news plus noise reduction, sports and fine arts comprise the eight factory preset settings that are included to provide a starting point for finding the desired sound for the station. The general-purpose settings are intended to complement voice and music and the fine arts setting is optimized for classical or jazz formats. News plus noise reduction uses the dynamic noise-reduction feature of the 9200 to help quiet noisy sources.

Once a preset has been selected, it may be altered by using the LESS-MORE control. This allows adjustment of the 9200 with one control changing many parameters simultaneously. Advancing the LESS-MORE control in the MORE direction results in a louder, denser sound with increasingly more of the negative side effects (distortion, loss of highs, etc.) associated with turning the processing "up." Conversely, by moving in the LESS direction, the sound becomes softer and less dense with fewer side effects.

Individual parameter control is possible with the full control menu. Available parameters include AGC drive and release time, gate threshold, bass coupling, multiband drive, multiband release, multiband clipping, noise-reduction downward expansion threshold and high-frequency clipping. In expert control, the user can adjust final clipping and individual drive levels to and output levels from each of the bands of the multiband limiter. Eight user presets are available for storing customized settings.

Two levels of equalization adjustment are available. Basic equalization includes gain adjustment for the low- and mid-frequency bands. Also included is a versatile high-frequency section that gives the choice of using the fixed NRSC curve or selecting other curves with steeper slopes that can be adjusted to provide greater amounts of boost. Parametric controls for the low- and mid-range bands, a DJ bass boost feature and a mid-bass control are added in the expert equalization menu.

The KVTO Berkeley facility presents a challenge to an audio processor. Being a Class C station limited to 1kW of power without night-time protection, KVTO is processed aggressively to cover as much of the San Francisco Bay Area as possible. To make matters more interesting, KVTO is a time-brokered station programmed by many different clients that provide audio with varying quality.

The 9200 has to be pushed hard to keep up with the KVTO Optimod 9100 that has been modified for increased loudness capability. Subjectively, I find the sound of the 9200 to be brighter and contain more high-frequency detail than that of our modified 9100. It also has a more "airy" sound quality than the 9100 on musical cuts and voices. I was able to get a more satisfying bottom end out of the 9200 with more dynamic feel and punch. In critical listening with some voices, the 9100 has a warmth and fullness in its mid-range that is not conveyed by the

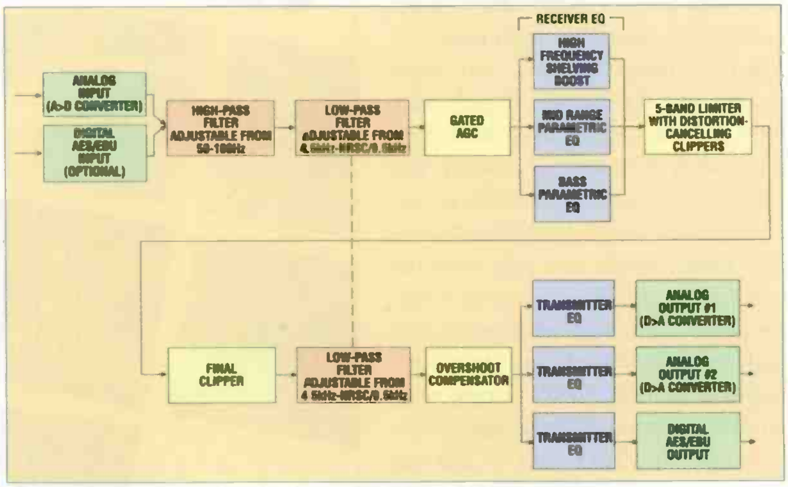


Figure 1. Functional block diagram of the 9200.

9200, which by comparison can sound a bit lifeless.

The multiband limiter does a good job of making the varying quality of KVTO's program material sound more spectrally consistent. This is not an easy task. It can bring the audio from a dull cassette recording up out of the mud and make it sound brighter and more intelligible. The noise and distortion of poor source material are sometimes also brought up, but the result is still a positive one creating a more listenable sound.

Turning on the dynamic noise reduction and adjusting the downward expander threshold almost completely removed a low-frequency room rumble from one segment of a recorded program, but resulted in some noise pumping on a hissy segment that followed. The manual cautions discretion when using the noise reduction, and that better results may be obtained by turning it off if extremely noisy audio is expected.

After five months of operation at KVTO, the 9200 has been completely reliable in the Orban tradition without as much as a single glitch. The Optimod 9200 offers an extensive feature list, a high degree of flexibility and rock solid stability. It can be an easy processor to set up using the LESS-MORE control for those seeking simplicity or a complex processor using the full control and expert adjustments for those wishing to be power users. All of these things are available with the 9200 for less money than you would pay for the Orban analog counterpart with fewer features and less flexibility.

*Paul Marks is chief engineer of KBLX-FM/KVTO-AM, San Francisco.*

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# New Products

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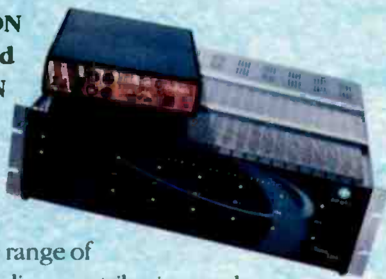
**Intraplex**



▶ **IntraLink ISDN multiplexer and OutBack ISDN codec:** these ISDN products allow flexible use of lower-cost ISDN services for a wide range of applications, including contribution and distribution of program audio and interfacility intercom communication, as well as data transmission for call screening, automation system control and LAN connection; IntraLink is a space-efficient unit that supports connection and management of up to six individual BRIs in a three-rack unit while allowing personnel to monitor and manage multiple channels and codecs from a single point via a PC; the OutBack is a compact, easy-to-use ISDN codec that supports MPEG Audio Layer II and G.722 coding; it is ideal for remote program feeds.

508-692-9000; fax 508-692-2200; sales@intraplex.com. www.intraplex.com
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## Fixed vacuum capacitors
**Comet North America**

▶ **MC Mini-Caps:** a family of high-quality factor, high-current capacitors designed for use in demanding RF applications; the MC Mini-Cap capacitors are a reliable and cost-effective alternative to using mica, ceramic "doorknob" or banks of ceramic "chip" capacitors that can have limited current-carrying capability and tend to show considerable thermal instability; the MC Mini-Cap features a compact design with a conservatively rated hold-off voltage of 15kV peak test; the unique design results in minimal internal inductance and features high-quality RF electrical connections that allow the capacitors to perform reliably at frequencies up to 100MHz and higher.

203-852-1231; fax 203-838-3827
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## Analog oscilloscope
**Iwatsu America Inc.**

• **Model SS-7811:** this oscilloscope is an eight-trace, multifeature precision instrument that incorporates high-speed automatic setup, full TV triggering with field and line selection, a fast sweep to 2ns/div. and a five-digit frequency counter with +/-0.01% accuracy; some of the high-performance features include DC to 100MHz bandwidth, 2mV/div. sensitivity and save/recall for 32 panel settings; it also includes easy-to-use cursor functions and offers a responsive multifunction control for hold off, delay, trace separation and TV line.

201-935-8486; fax 201-935-8533; www.iwatsu.com
Circle (162) on Free Info Card

## Ultracardiod dynamic microphone
**Electro-Voice**

• **N/D957:** an ultracardiod dynamic microphone that features a "clean sheet" design; this true-solution-oriented hand-held vocal microphone was specifically developed as an ultrahigh gain before feedback design; according to Electro-Voice, it is without rival in high SPL performance venues; it features a unique truncated profile, ultracardiod pattern and peak-free response to provide a high-level performance-oriented microphone; the N/D957 also includes the first hygienic windscreen assembly that is easy to clean; a new design concept is incorporated to provide both near- and far-field voicing and the integral personality module can be configured for "up close and in your face" or back-up optimization.

800-234-6831 or 616-695-6831; fax 616-695-1304
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## Product catalog available on CD-ROM, web site and fax on demand
**Andrew Corporation**


▶ **CATALOG 37:** a complete reference document of all products and services offered by Andrew; it is updated on a quarterly basis and the 784-page full-color catalog includes new sections on Andrew Institute technology training, wireless accessory products, wireless RF active products and base station antennas; the catalog is available on the company web site (www.andrew.com) or from the Andrew fax-on-demand service (800-861-1700 or 708-873-3614).

800-255-1479 ( extension 301); www.andrew.com
Circle (164) on Free Info Card

## Product applications guide
**Sabine**


▶ **The FBX Advantage:** tips for controlling feedback, pointers on using the FBX Feedback Exterminator and definitions of often-misunderstood terms can be found in this free booklet; using easy-to-understand language, the guide explains what causes feedback and discusses how to control and avoid it; it also provides clear explanations for terms, such as comb filters, constant-Q filters and net gain before feedback.

904-418-2000; fax 904-418-2001
Circle (165) on Free Info Card

### Review audio with ease

**Prophet Systems Inc.**

• **Crystal Ball:** this product allows you to easily review audio by typing the date and time of the audio to be reviewed; retrieval is automatically pulled from the hard disks of up to four computers or off DAT for instant replay without interrupting any program being recorded; Crystal Ball records onto compact storage digital DAT tapes.

800-658-4403; fax 308-284-4181; sales@prophetsys.com
Circle (166) on Free Info Card

### Audio tubes product guide

**Svetlana**

▶ **1997 Audio Tube Product Guide:** this catalog provides detailed technical information, including performance curves, for each Svetlana audio tube, from pentode, to beam tetrode to power triode; the guide provides information on the full Svetlana audio tube line and is especially helpful for users buying new tubes for amplifiers and OEM designers.

205-882-1344; fax 205-880-8077
Circle (167) on Free Info Card

### DAB receiver in the works

**Philips Electronics' Advanced Systems & Applications Laboratory Eindhoven**

• **DAB 752:** development has begun for this DAB receiver that will be the successor of the Philips DAB 452, also known as the DAB "reference" receiver; applications for the DAB 752 will include monitoring and synchronizing of DAB transmitters, field tests, lab measurements and mobile data reception in combination with a PC; the DAB 752 will be designed specifically for the professional user and will feature a 19-inch 2U high cabinet; other features include Band III and L-band reception, half-rate audio support, analog and digital audio output and PC-interface (RS-232) for information logging and control; a prototype should be available in November with the first products available in the first half of 1998.

+31 40 2732080; +31 40 2737353; www.philips.com/sv/newtech/
Circle (168) on Free Info Card

### AES router

**Leitch**

▶ **ASR-16x16:** this AES router includes the new synchronous quiet switch (SQS) processor that guarantees quiet switching of AES digital audio; the SQS provides two processing steps: 1) synchronous switching that maintains AES framing during and after crosspoint switches so that downstream equipment does not lose lock during a switch, and 2) user-selectable crossfade that guarantees a quiet switch between any two sources by fading the two signals near the switch point; the fade may be enabled or disabled, and the fade duration is user-adjustable.

800-231-9673; fax 757-548-4088
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# New Products

### PDP series of tapes expanded
Sony Electronics' Recording Media and Energy Group

• **Pro DAT Plus series of tapes:** the PDP series of tapes has been expanded to offer extended lengths of 15, 35, 50, 65, 95 and 125 minutes; the tapes now come with a shatter-proof jewel case, providing greater durability in the field and better storage for archiving; an anti-static lid and a high-precision, temperature-resistant shell with a sliding hub cover virtually eliminate dust and dirt.

201-599-3445; fax on demand 800-SONY-022; www.sony.com
Circle (170) on Free Info Card

### Digital network audio transceiver/recorder/editor
Nagra

▶ **Nagra C-PP:** a digital network audio transceiver/recorder/editor that incorporates many of the performance features of the ARES-C in a lower-priced studio unit; like the ARES-C, C-PP is designed around PC card FLASH memory technology; instead of tape or disc storage units, the credit-card-sized PC cards provide random access to recorded data, are reusable, require less power and are weather and vibration resistant; more than two hours of continuous high-quality mono recording time is provided by a single 64MB PC card and the system has no internal moving parts so it is virtually maintenance-free; a built-in ISDN codec provides digital transmission virtually anywhere in the world through the ISDN network.

+41 21 7320101; fax +41 21 7320100; mail@nagra.com
Circle (171) on Free Info Card

### Pivoting panel-mount racks
Datatel

• **PPM racks:** these 19-inch pivoting panel mount racks are ruggedly constructed of 0.060-inch steel and each unit is equipped with 0.125-inch thick multirails tapped with 10-32 threads on one side and 12-24 on the other; a pivoting hinge system is equipped with a positive stop that permits the entire rail assembly to be tilted down away from the chassis for easy punchdown; the racks allow easy rear access and six models are available: the PPM-2, the PPM-4, PPM-6, PPM-6-12, PPM-8-12 and the PPM-8-18.

201-616-7078; fax 201-616-7179
Circle (172) on Free Info Card

### New additions to VM-3 series
Kramer Electronics

• **VM-3M:** a 1:3 microphone amplifier that splits a single input source into three identical outputs without any discernible signal degradation; like all products in the VM-3 series, the VM-3M offers high-quality audio distribution in compact housings that can easily fit in a toolbox or pocket; it uses low-noise circuitry.

• **VM-3AB:** this product can be used as a 1:3 stereo audio distribution amplifier, as well as a 1:3 balanced mono DA with one channel being positive and the other negative; it also splits a single input source into three identical outputs with no discernible signal degradation.

888-303-5600; kramerel@netvision.net.il
Circle (173) on Free Info Card

# Contract Engineering

*Continued from page 12*

standardize installation practices, the Electronics Industries of America (EIA) and Telecommunications Industries of America (TIA) have collaborated to create EIA/TIA 568. This guideline covers all aspects of cabling methods.

## Hubs, bridges and routers

The network hub acts as a common terminus for networks configured in a star arrangement. Depending on network size, hubs range from a small box with a few ports to a large rack-mounted system with thousands of ports. Hubs can be passive or active. A passive hub is a resistive combining network for LANs using coaxial-based media and is almost non-existent.

Active hubs electronically regenerate the signals on its ports. Currently available hubs support data rates up to 100Mb/s and allow mixing and matching of data rates, protocols and media types. If you require data rates in excess of 100Mb/s, *switching hubs* are designed to complement the higher-performance protocols.

It's occasionally desirable to combine two or more LANs within a company. A *bridge* is a relay that reads, buffers and sends information between LANs operating on different protocols and makes the two datalinks appear as one to the higher-level applications. A *router* is a network device that relays network traffic between a LAN and other network devices. *Simple network management protocol* (SNMP) was developed to simplify the task of managing hubs, bridges, routers and network interfaces. Using a special program, network managers can view, configure and troubleshoot all devices on the network using SNMP.

*Kevin McNamara, CNE,* BE Radio's *consultant on computer technology, is president of Exegesis Technologies, a consulting firm in New Market, MD.*

# Managing Technology

*Continued from page 16*

scales, from on-premise paging, to local and regional paging services with multiple transmitters, to national and global services linked by satellite. Paging companies now offer two-way capability where the user can reply to messages. Now cellular and PCS services are in the act, blending paging into their lists of features.

Datacasting has one key advantage over the wired world — *portability.* FM datacasting could have an advantage over TV datacasting with *mobility.* The future of FM subcarrier datacasting will hinge on its ability to offer valuable information to users who are in motion. It is interesting to note that the three proponents of high-speed subcarrier technologies have highly mobile applications: highway and traffic information, consumer news and information and paging. New radio datacasting standards, if any are established, will have to take advantage of the mobility that broadcast signals offer.

Somewhere between the wireless world of PCS/cellular communications and the wired world of the Internet, radio datacasting has a niche of its own. Yet, it does not look like it will become a big deal for broadcasters. Today's estimated $40 million in national annual subcarrier lease revenue is dwarfed by the roughly $8 billion in advertising revenue of FM broadcasters. Broadcasters like the additional padding put on the bottom line by a subcarrier tenant. And there is clearly room for growth as new users emerge. The business of datacasting, however, seems to belong to another industry, unless we broadcasters come up with a better way to use our auxiliary bandwidth.

*David Maxson is vice president of Charles River Broadcasting Company, which owns WCRB Boston and WFCC Cape Cod, and is also a principal of Broadcast Signal Lab, New England's signal monitoring company.*

***Editor's note:** Some data and the charts for this article were obtained from the NAB publication* FM Subcarrier Report/Technology Guide. *This report is available from the Station Services Department of the NAB by calling 800-368-5644.*

# Business/People

## BUSINESS

**Fidelipac**, Philadelphia, agreed to market Grass Valley, CA-based Graham-Patten's DMIX-41, a four-channel no-frills mixer, as the Fidelipac DMR-41. The unit will serve in the radio market as both a front end to the Dynamax Digital MX/D radio console and as a router for automation of radio programming.

**Broadcast Electronics**, Quincy, IL, was awarded a contract in excess of $3 million by the United States Information Agency, parent organization of the *Voice of America*, in conjunction with the government of the Hellenic Republic of Greece. Under terms of the agreement, a total of 84 transmitters will be supplied over a 18-month period.

**360 Systems**, Westlake Village, CA, sold a Shortcut personal audio editor to the IMS Radio Network, Indianapolis. IMS is owned and operated by the Indianapolis Motor Speedway and has originated the radio broadcast of the Indianapolis 500 for 45 years.

Additionally, 360 Systems announced that its products, including a Shortcut, an Instant Replay audio player and a DigiCart II recorder, were installed throughout the TV and radio facilities of the Southern Command Network (SCN) in Panama City, Panama. The SCN is part of the Armed Forces Radio and TV Service Network.

**LBA Technology, Inc.**, Greenville, NC, was selected by the International Broadcasting Bureau to provide critical components for the modernization and retrofit of the Voice of America medium-wave AM station in Bangkok, Thailand. Supplied under the contract were RF inductors, phase sampling equipment, coaxial isolation coils and other high-power accessories.

**Panasonic/Ramsa**, Cypress, CA, announced that Merging Technologies' Pyramix Virtual Studio will be used by the Montreux Jazz Festival to record several of the evening concerts directly to hard disk. It will be the first time that a computer-based recording system will be employed by the festival.

**Digital Courier International**, Vancouver, announced the availability of collect sending for radio stations on its two-way North American network. With the company's *DCI Collect* service, member stations can reverse charges on delivery of commercials, voice tracks, music and other short-form audio received over the network.

**ABG, Audio Broadcast Group** opened a sales office in the Tacoma, WA, area. The office, staffed by Pat and Bernice Medved, can be contacted at: toll free 888-565-9960; telephone 253-565-9360; fax 253-565-9359; E-mail: medved@abg.com. The office's address is 6824 19th St. West, Ste. 185, University Place, WA 98466.

## PEOPLE

**Frederick L. Godard** was promoted to president and chief operating officer for Leitch Technology Corporation, Chesapeake, VA. **Bob Lehtonen** will retain his position as chief executive officer, but is resigning as president.

**Barry Margerum** was announced as CEO of Euphonix, Palo Alto, CA. **James Dobbie**, who has held both the chairman and CEO positions since 1991, will continue as chairman and will support the company on a part-time basis.

**Mark Eagle** was promoted to senior vice president/director for Columbine JDS, Golden, CO.

**Andrew Greenebaum** was appointed executive vice president and chief financial officer for CD Radio, Inc., Washington, DC.

### WEB SITES

| | |
|---|---|
| Kart-A-Bag | www.kart-a-bag.com |
| Mediatron | www.mediatron.com |
| PMA Marketing | www.amfmtv.com |

## DAB forum highlights the 103rd AES convention

A forum on digital audio broadcasting (DAB) is one of the highlights that will be featured at the 103rd AES convention, Sept. 26-29, in The Jacob K. Javits Convention Center in New York City. The broadcast forum on digital audio broadcasting will take place Sept. 26, at 2:00 p.m. in the Special Events Hall. A panel of experts will address the critical issues related to DAB throughout the world and will include representatives from NAB; EIA; Multimedia Communications Research Laboratory, Bell Labs; Lucent Technologies; USA Digital Radio (Gannett); Westinghouse Wireless (USADR); CRC (Canada); CD Radio and CBS, Inc. Attendees can register in advance at www.aes.org.

## NAB call for presentations

The National Association of Broadcasters (NAB) has made a call for presentations requesting proposals for the NAB Broadcast Engineering Conference, NAB MultiMedia World and NAB Communications and Connectivity '98. The deadline for the one-page proposal and a brief presenter biography is Oct. 31, 1997, and full manuscripts are due by Jan. 30, 1998. If the proposal is accepted, the author will receive an NAB '98 complimentary registration. Papers that are accepted will also be up for consideration for publication in the 1998 Broadcast Engineering Conference Proceedings or Interactive Insights '98: A MultiMedia Compendium. For more information, contact John Marino, vice president, Science and Technology, NAB at 202-429-5391 or E-mail to jmarino@nab.org.

## NANBA supports worldwide standard for HF digital audio broadcasting

The North American National Broadcasters Association (NANBA) is urging all major worldwide broadcasting unions to adopt a single worldwide standard for high-frequency digital audio broadcasting. According to NANBA secretary general Bill Roberts, a single global standard would allow manufacturers, broadcasters and consumers to have superior access to programs without the extra burden of having to choose between incompatible technologies.

High-frequency digital audio broadcasting could replace analog high-frequency broadcasting, currently the international standard for broadcast short-wave bands. The growing demand for higher-quality broadcasts combined with the increasing band congestion is driving the standard and advanced HF digital development projects are already under way in France, Germany and the United States.

## @Home Network expands services

The @Home Network (www.home.net) has teamed up with Internet music partners NetRadio Network (www.netradio.net) and TheDJ Network (www.thedj.com) to offer high fidelity music, news and sports content.

In addition to receiving diverse programming, @Home subscribers can receive a new broadband audio service, dubbed "TuneIn" that delivers high-quality audio streams directly to a user's desktop at 80kb/s. The @Home's high-bandwidth service allows users to tune in even while browsing on the web and it is always on. @Home subscribers can use the beta release of the TuneIn feature located on the home page or the pop/arts page where they can choose from eight different music channels. The TuneIn feature also offers interactivity so that users can find out the title of the song that is playing, get more information about the artist or purchase a CD.

TuneIn will feature live music events, technology news and sports talk/news through partnerships with N2K's Music Boulevard, CNET Radio and CBS SportsLine. In order to begin listening, subscribers need Progressive Networks' Realplayer, which is included in the latest version of @Home's customized Netscape browser.




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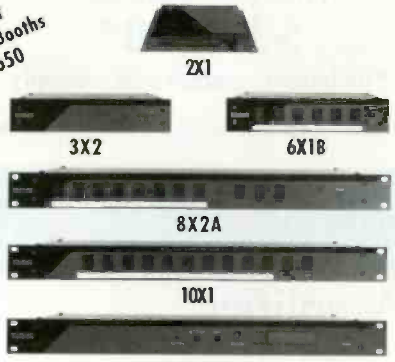


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| Digigram USA | 31 | 12 | 703-875-9100 |
| DRS Ahead Technology | 52 | 35 | 408-226-9991 |
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| Energy-Onix | 16 | 7 | 518-758-1690 |
| Equi=Tech Corporation | 66 | 41 | 514-597-4448 |
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| ESE | 86 | 79 | 310-322-2136 |
| Euphonix | 59 | 55 | 415-855-0400 |
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| Gentner Communications | 29 | 24 | 800-945-7730 |
| Gepco | 74 | 75 | 312-733-9555 |
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|---|---|---|---|
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| Murphy Studio Furniture | 30 | 25 | 619-698-4658 |
| Neumann (USA) | 17 | 8 | |
| Neutrik Canada | 38 | 29 | 514-344-5220 |
| NPR Satellite Services | 33 | 14 | 202-414-2613 |
| NSN Network Services | 48 | 31 | 800-345-VSAT |
| OMB America | 53 | 36 | 305-477-0974 |
| Orban | 11 | 18 | 510-297-2774 |
| Allen Osborne Assoc. | 66 | 42 | 805-495-8420 |
| Penta Labs | 89 | 88 | 818-882-3872 |
| Phasetek Inc. | 78 | 65 | 215-536-6648 |
| Prism Media Products | 12 | 19 | 201-299-7790 |
| Pristine Systems Inc. | 86 | 78 | 864-292-0300 |
| Prophet Systems Inc. | 21 | 11 | 308-284-3517 |
| QEI Corporation | 70-71 | 71 | 800-334-9154 |
| Radio Computing Services | 23 | 21 | 914-723-8567 |
| Radio Programming & Mgmt. | 58 | 54 | 800-521-2537 |
| RadioSoft | 40 | 44 | 904-426-2521 |
| Radio Systems | 61 | 58 | 609-467-8000 |
| Roscom | 91 | 93 | 770-992-2230 |
| Satellite Systems | 56 | 52 | 719-634-6319 |
| SCEC '97 | 67 | 60 | 800-288-8606 |
| Scott Studios Corp. | 35 | 26 | 800-330-3004 |
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| Silicon Valley Pwr Amplifier | 84 | 69 | 408-986-9700 |
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# The Last Byte

## Satellite DAB update

**By Skip Pizzi, editor in chief**

There's been plenty of action lately in Satellite DAB (S-DAB) for the United States. In April, the FCC auctioned two licenses for S-band service (2,320-2,345MHz) to the highest bidders: CD Radio and American Mobile Radio Corporation (AMRC). Since then, both companies have been moving forward in their plans to launch service, scheduled to begin in late 1999.

The two companies eventually expect to offer between 60 and 100 channels of pay-radio programming, with subscriptions costing approximately $10 a month or less. CD Radio has been more specific about its programming plans, which include 30 CD-quality music channels and 20 news/sports/talk channels from its twin satellite system. (Both birds are being built by Loral Aerospace, and each will beam the same signal in a transmit-diversity scheme.)

The antenna required is a silver-dollar-size disk that must have line-of-sight to one of the satellites, thus ruling out most in-building listening. In fact, CD Radio calls its service satellite-to-car radio. But even car-mounted antennas won't see a satellite in densely constructed "urban canyons." Satellite DAB, therefore, may experience its worst reception in the areas of highest population.

This could be solved with terrestrial "gap-filler" repeaters, but neither company is committing to this. Their plans seem to center on smaller markets and rural areas first, where less choice in radio programming currently exists. Nevertheless, it's likely that COFDM-style transmission formats will be adopted, allowing on-channel terrestrial repeaters.

### Receiver ideas

Like DBS-TV, it's expected that each of the two services will require a separate receiver. Like cable radio, the receivers will likely display artist and title data in a text window.

In addition to seeding the market with three-band (AM/FM/S-DAB) radios in new cars and trucks, CD Radio is developing an adapter and wireless antenna for easy aftermarket conversion of existing car audio systems by consumers. The receiver will be fashioned to plug into a cassette or CD slot and internally couple with the player's audio pickup. A faceplate on the adapter will remain exposed for tuning control and display. The battery-powered antenna will stick to the inside of the vehicle's rear window and retransmit its wideband pickup across the passenger compartment to the receiver.

### Market questions

As with any DBS system, start-up costs for S-DAB operations are high. AMRC paid $89 million for its license, and CD Radio paid $83 million. (Congress had hoped to raise more, while CD Radio had hoped to pay less with a Pioneer's Preference, which was denied.) Add another $300+ million for satellite construction and launch, plus the cost of operations, and each company is facing expenditures on the order of $500 million before the first customers sign up. Further revenue drain may occur if the S-DAB companies have to subsidize receiver manufacturers (like DBS-TV services do today).

On the other hand, if S-DAB is a hit, these two operators will have the market to themselves, and their license fees could seem like bargains. Analysts cite a figure of approximately one million subscribers each for the services to break even, and best-case estimates forecast that the two services could have a combined listenership of 15 million by 2003.

Until then, however, plenty of patient capital will be required. AMRC is blessed with the deep pockets of its parent companies, among them AT&T and GM/Hughes. CD Radio is less well-endowed, relying on private investors and venture capital. It's been successful in raising enough to keep its 10-person operation going for now, but still has a steep funding hill to climb.

S-DAB's ultimate success may rest simply on whether listeners will pay for radio service. Will the choice, content and nationwide uniformity of S-DAB service be sufficiently compelling to get people to subscribe? Will the quality of service and coverage be good enough to keep subscribers around?

Finally, if the subscription model is unsuccessful, will these services convert to tried-and-true commercial radio operations? Nothing in their licenses precludes this, and some pundits have observed that it may be the S-DAB services' ultimate plan, with the subscription idea serving simply to deflect some of the NAB's opposition until licenses were granted.

S-DAB supporters claim that most who will listen to S-DAB service are already lost to local radio by their preference for cassette and CD players or they are at least already dissatisfied with existing choices. If so, today's trends among radio stations to eliminate local content and move toward further homogeneity of format play right into S-DAB's hand. Keeping a strong, vibrant, unique and locally rooted service is the best defense for radio stations against the coming competition from the skies.


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103