Radio World®

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Welcome International Readers

NEWS WATCH

ITC to Intro New Cart Machine

Bloomington, Illinois USA Just one month after purchasing International Tapetronics Corp. (ITC) from 3M, new owner Donald Carle announced that the company planned to introduce a new series of cart machines.

The Series 1, in development by 3M for the past year, will offer high performance, LED meters and easy maintenance and care in a new cast aluminum nickel-plated deck, according to ITC.

Features on the record/playback and companion playback-only units include mono/stereo capability, high-speed cue, DC servo motor, three standard cue tones, 1 kHz add and defeat and a new cart hold-down system that uses plastic parts to lock carts in smoothly.

Carle, a successful electronics entrepreneur based in West Vancouver, British Columbia, Canada, agreed to purchase ITC last December. In the deal, he acquired not only the company's plant and laboratory in Bloomington, Illinois, 55 employees and the right to manufacture cart machines and carts under the ITC name (3M will furnish the tape), but also 50 patents.

Part of ITC's marketing efforts for the Series 1 will be focused in Europe. Carle spent the first six weeks as owner of the company strengthening ties in nine European countries.

"It's very key to me that we continue to build up our outside distribution," he said. "I want to attune to the needs of the market, not some engineer's dream of what it should be. To insure that, I want to get around and see as many customers as possible."

For more information on the Series 1, circle Reader Service 94.

Eight FM Stations Plan Combiner Sharing

Shoreview, Minnesota USA A group of eight FMs in the Minneapolis area are joining together to share a new combiner and antenna that should be in operation by June.

LDL Communications will supply and install an Alan Dick combiner and 12-bay circularly polarized FM antenna for the Shoreview FM Antenna Group. The antenna system has a 10-station capacity.

LDL Sales and Marketing VP Ray Tattershall said antenna combiner systems are an opportunity for stations to have a state-of-the-art system they might not be able to afford individually.

The company has installed similar systems in Jacksonville, Florida, Atlanta, New Orleans and in various Canadian cities.

For more information from LDL, circle Reader Service 77.

(continued on page 2)

The Berlin Wall Barrier Broken

by Dee McVicker

West Berlin WEST GERMANY Christian Eckhardt of West Berlin's SFB never actually said he felt like he was talking to a brick wall. But, of course he was. That brick wall, felled by political pressure in November, 1989, is what left his radio show on the deaf ears of East Berliners.

Not that they weren't listening; they were. SFB, a community of West German radio and television stations operating under the auspices of West German government, never had a problem penetrating The Wall into East Berlin.

Given its close proximity to the Iron Curtain, SFB's West Berlin broadcasts were virtually immune to this political barricade. Other forms of communication, however, were not as successful in breaking the Berlin Wall's concrete barrier.

As Eckhardt explained, "If you know (that by calling) somebody in East Berlin that (you've caused him) real prob-

(continued on page 10)

A sentry of international flags decorates the front of SFB radio-TV in West Berlin.

FM Rates Belar

by Russ Mundschenk, CE
WEAZ-FM

Philadelphia, Pennsylvania USA RF real estate has become a very valuable commodity. As more and more stations avail themselves of power upgrade opportunities, the "people next door" will be that much closer to our doorstep.

WEAZ-FM has been continually committed to close maintenance of modulation and center frequency to uphold the "good neighbor policy."

Since we are short spaced, with co-channel stations in New York City and Washington, DC, we have a significant area of signal overlap midway between Philadelphia and these cities. Tests have shown that minimization of the amplitude beat frequency (and subsequent multipath reduction) in these areas can be achieved by keeping as close as possible to our center frequency.

Repeat customer

EZ-101 has always been impressed with the quality, accuracy and cost effectiveness of Belar monitors, having purchased two FMM-1/FMS-1 monitors (one of them serial number 1) and one FMM-2/FMS-2 monitor. All three units are still in continuous use and none (not even the old guy) have given us a bit of trouble.

When Belar told us they would be producing a digital monitor, the FMM-4A, that would continuously monitor our main carrier, pilot and subcarrier frequencies, we requested one of the first units.

We installed the monitor at the studio and bridged the on frequency output of the RF amplifier to the unit's RF input. We then connected the pilot output of our Belar FMS-2 stereo monitor to the frequency monitor's pilot input. In this configuration, the FMM-4A can be programmed to automatically alternate between the carrier and pilot frequencies.

The unit also will measure the frequency shift of any subcarrier when connected to the output of a Belar SCM-2 sub-subcarrier monitor. The FMM-4A is frequency agile, and will measure the deviation from any 100 kHz center frequency

(continued on page 24)

Radio World En Español see page 16

(continued from page 1)

Government Seeks Input on Spectrum

Washington, DC USA US broadcasters now have an opportunity to tell the federal government their views on how the entire radio spectrum could be better managed—an opportunity that may not come again for 20 years, according to National Telecommunications and Information Administration (NTIA) chief Janice Obuchowski.

NTIA launched a procedure last December that "will establish the spectrum policy foundations to lead the US into the 21st century," Obuchowski said.

She encouraged comment from radio equipment manufacturers, firms providing services using the spectrum, individual spectrum users, developers of new technologies and federal agencies that use the spectrum.

NEWS WATCH

Harris' Swanson Wins Award

Washington, DC USA

Harris Broadcast Senior Staff Scientist Hilmer I. Swanson received the National Association of Broadcasters (NAB) Engineering Achievement Award for 1990 at the association's annual convention in Atlanta, held 31 March to 3 April.

Swanson is credited with development of much new technology used in AM transmitters. NAB recognized him for giving AM broadcasts the potential to compete with other high fidelity media.

Swanson, who joined Harris (formerly Gates Radio) in 1965, made significant contributions to the development of pulse modulation techniques and, more recently, digital modulation for AM.

Ultimate Radio Stalls

Washington, DC USA For more than a year, the National Association of Broadcasters (NAB) has touted its "ultimate" radio at just about every meeting or function its representatives attend.

But despite the continued promise of the high quality receiver, which is to feature continuous tuning, AM stereo/FM stereo and FMX (a noise reduction system designed to improve FM stereo coverage), a date for the radio's arrival to consumers remains uncertain without an agreement to manufacture it.

Although NAB has not been able to secure a manufacturer, Radio Board Chairman William Sanders said the ultimate radio remains a "high priority." So far, according to Sanders, NAB has spent approximately $50,000 on the project.

Government Blamed for Slow AM Stereo Growth

Washington, DC USA The manufacturer of the dominant AM stereo transmission system in the US has cited the government's decision not to select a single AM stereo standard as the major reason for the slow growth of AM stereo acceptance in the US.

Currently, only about 14% of US AM stations broadcast in stereo.

"The 1982 Federal Communications Commission many-systems decision, coupled with US antitrust laws, has been causitive to slow broadcasters' conversion (to AM stereo) and non-automotive receiver availability," said Motorola's Frank Hilbert.

"Broadcasters continue to cite the lack of receivers and lack of a standard as reasons for not converting. In turn, foreign receiver manufacturers (the primary supplier of receivers in the US) cite the lack of stereo broadcasts and lack of a standard as reasons for not participating," he added.

Motorola suggested the government choose a single standard to facilitate the acceptance of AM stereo.

Motorola's C-QUAM system is already considered the industry's de facto standard by many, with more than 700 stations broadcasting with the system worldwide.

Japan Wary on DAT

by Bryan Harrell

Tokyo JAPAN Sensitive to further criticism from the US, Japanese manufacturers continue to be extremely cautious on questions about DAT, even when the questions pertain to professional broadcast equipment.

An Aiwa representative acknowledged his company's deal in the spring of 1988 to supply 1000 units of a special broadcast- version DAT machine to Harris Corp. of Melbourne, Florida Since then, however, Aiwa has not exported any machines, though some—made their way through grey market channels to America and Europe, alongside those of other manufacturers—Sony and Technics in particular.

The rep indicated, however, that Aiwa has a strong interest in supplying broadcast-use DAT equipment, though there are no definite plans at present.

Japanese manufacturers continue to be extremely cautious on questions about DAT, even when the questions pertain to professional broadcast equipment.

This fall, Aiwa will make strong efforts to market its new portable DAT recorder. The main unit is extremely compact (3.7"×1.5"×5.75", 21.5 ounces), and boasts a price tag to match: 77,000 yen (US$550 at the current rate of US $1=140 yen). For recording from analog sources, the snap-on A/D converter (22,000 yen/US$157) is necessary.

Another major feature is still video recording with the connection of a snap-on adaptor (53,00 yen/US$379), which sandwiches some 3600 images alongside a digital audio soundtrack on a single 120-minute cassette. Though resolution is not exceptional, future versions of this technology for broadcast use could be used in such applications as news programs.

Other firms were less talkative on professional DAT.

A representative of Matsushita (Panasonic/Technics) said his company was interested in future development and currently working on several different prototypes but he had nothing to announce at present. Representatives of two other firms currently producing broadcast equipment, JVC and Teac, declined to answer questions on DAT.

Sony's public relations office in Tokyo, however, has a lot to say about the future of DAT. "Professional DAT recorders are not affected by consumer considerations," spokesman Paul Campbell said.

Sony has exported to the US and other countries for the past two years, he said. The Sony professional DAT line now includes a portable recorder, a desk-top recorder and a duplicator and Campbell indicated the line will be developed further in the future.

"Broadcasters are currently using a variety of different systems, both analog and digital," Campbell explained, "and we expect that DAT will be a very attractive proposition to them."

He added that Sony's current range should be extended to suit the needs of broadcasters. "We will be talking to them about this in order to make DAT a broadcasting standard," he said.

■ ■ ■

Bryan Harrell is a Tokyo-based contributor to **Radio World**.

INDEX

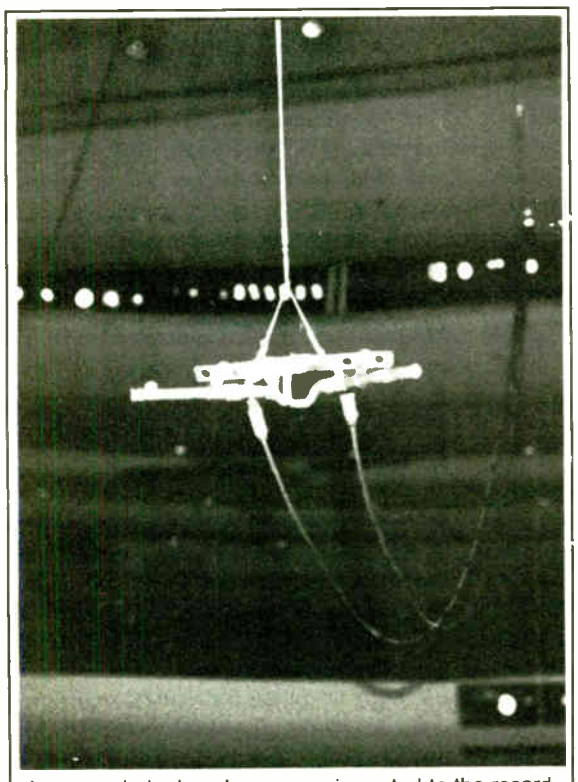
Capturing LA's Philharmonic

by Frank Beacham

Los Angeles, California USA In a cramped, dimly lit sound booth, atop tight rows of tiny knobs on the audio console lay an orchestral score for Jean Sibelius' *Symphony No. 5 in E flat, Op. 82*. Inches away, intensely scrutinizing the sound through headphones, is Joseph Magee, part musician, part engineer, part producer and "ear" extraordinaire.

Behind the scenes

A hundred feet below, in a wooden stage shell in the Dorothy Chandler Pavilion, 100 members of the Los Angeles Philharmonic majestically perform the Finnish composer's "war symphony." It's dress rehearsal and the quest for perfection under guest conductor Vladimir Ashkenazy is intense.



A suspended microphone array is central to the recording process.

Magee, preparing to record the performance for national radio broadcast, has little time left for microphone adjustments. He directs audio engineer Fred Vogler, high atop a ladder in the nearly empty 3200 seat auditorium, to move the microphone array less than an inch.

In the minimalist world of microphone placement, a few inches can mean the difference between a great recorded performance and a disaster.

In the minimalist world of microphone placement, Magee says, a few inches can mean the difference between a great recorded performance and a disaster.

Joseph Magee, 36, is one of a handful of American audio engineer/producers responsible for the broadcast sound identity of a major philharmonic orchestra. A former professional musician who specializes in recording classical and acoustical jazz, he demands that everyone on his engineering team have musical, as well as technical, experience.

A seven-year veteran with the Los Angeles Philharmonic, Magee is in the midst of recording a 13-week concert series with the Los Angeles Philharmonic that will be broadcast on the National Public Radio (NPR) network next summer. The series is being co-produced by the LA Philharmonic, NPR and Los Angeles public radio station KCRW.

For Magee, every performance is different and each calls for a recording technique to suit the circumstances. But the general rule of thumb is the fewer microphones the better. Thus, he refers to his miking philosophy as "minimalist."

"Acoustical instruments in a classical setting don't sound quite right when you get up on them too close," Magee said.

Miking the classics

"The overall balance of the ensemble doesn't come together until you get away from it. Same thing with a large chorus. It is really up to the conductor to create the balance in the ensemble, not me. If I can come up with a stereo system that accurately represents what's occurring—as best as I can, because we know this is an interpretation—then the conductor and the ensemble can do the rest."

For this performance of the works of Sibelius and Schoenberg, Magee chose a spaced omni/near coincident microphone technique.

"It's a four-microphone system," he explained. "In the center, there's a spaced near-coincident cardioid B&K (Bruel & Kjaer) 4011 system with about a 110° spread. There are approximately 12 inches from capsule to capsule. Hard left and hard right are Sennheiser MKH 20 omni's."

The general rule of thumb is the fewer microphones the better.

The four microphones hang in the same plane on a single ceiling-suspended harness about 11′ above and 5′ in front of the orchestra. There is a distance of about 30′ between the two Sennheisers, with the B&K array in the center.

Magee chose this miking technique in part because of the 70′ high unsealed wooden shell enveloping the orchestra. "The shell in this house has gaps everywhere. Information is escaping out the back. And the shell doesn't reinforce the double bass instruments. They barely make it into the house," he said.

"By using this combination of spaced-omni and near-coincident techniques we can get in fairly close to the ensemble. So we are not out in the hall far enough to have to rely on that shell to project too far," Magee said.

Because of the shell problem, Magee added a Shoeps cardioid as a spot microphone to touch up the principal bass player. "It's the only spot mic out there and that's to get the bow on the string . . . the impact and definition. The spot mic is very low in the mix."

Knowing the setting

From the microphones Magee insisted on using Monster Cable's Pro-Link Series 1 and M1000 cable. "I know. When you start to get into the differences in sound in mic cables there are hard core engineers who are against this, many of them my best friends," Magee said, laughing. "And then other of my best friends who work hard at this and know cables say they are getting to be like microphones and preamps in the way they sound.

Magee rides gain as the band plays on

"I say the mic cable makes a lot of difference. The way I've proved it to other folks is to have a set-up with an orchestra like this and change only the cables . . . nothing else. There was a big difference. That's how I justified getting this cable in here."

Next in the chain are Jensen Twin

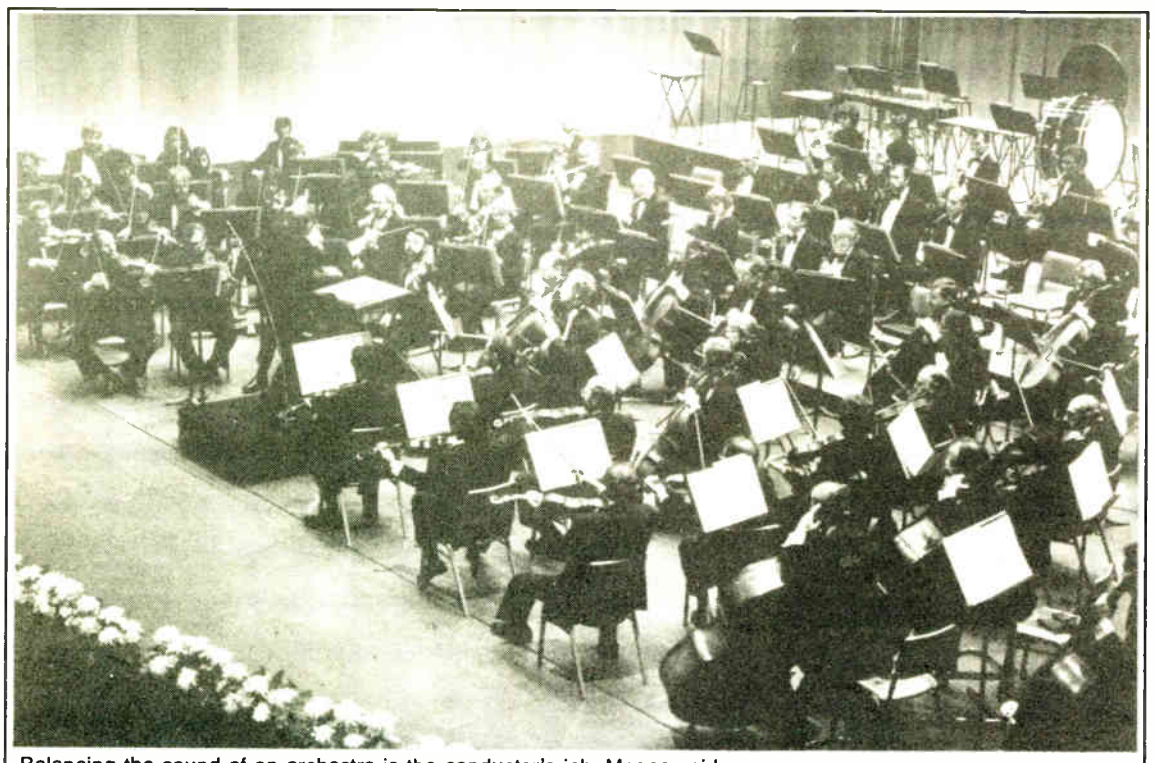
(continued on page 4)

Subtleties of Symphonic Miking

(continued from page 3)

Servo microphone preamps, located in the theatre rafters at the top of the microphone hang. "The Jensen has an incredibly beefy power supply. You can hit it with a wallop. That's an important thing to have when it's up in the middle of nowhere," Magee said.

The engineer's console of choice is the original prototype of the Ramsa WRS-852. "Great simplicity. No automation. Great grounding scheme. Quieter and has better cross-talk rejection than any of the studio consoles I've sat behind. Great for straight-to-two-track stereo recording," Magee exclaimed.



Balancing the sound of an orchestra is the conductor's job, Magee said.

The only processing in the mixing chain is a Lexicon 480 digital reverb unit. "This thing is a lifesaver because when you are doing live recording for broadcast you generally can't utilize ambient microphones due to the noise," Magee said.

The only quality concession to skimpy public radio budgets is the recording medium. Magee prefers high-quality analog but called it cost prohibitive in this situation. "In order to do really great analog, it's going to cost you about three times as much as digital. It's a sad thing we can't get our everyday digital to come up to that level," he said.

For the radio broadcasts, two digital recording systems are used simultaneously. The primary recorder is a Panasonic SV-3500 professional DAT recorder with custom-built analog electronics on the front and rear end. A dbx 700 Digital Audio Processor with a Panasonic VHS video recorder also is used to record the concert.

"The first thing I see in all DATs is the analog electronics are very, very poor, no matter how good the digital electronics are," Magee said. "So the first thing we do is get rid of them and put in good analog electronics. This machine sounds very nice but is still a little hard in the high end. A little brittle up there. Otherwise it sounds very nice. I would say most people would be happy with it."

Magee will record two performances by the orchestra of the same material. The best of both concerts will be combined with feature material produced by host Gail Eichenthal for each two-hour broadcast. Post production, which was to begin in early 1990, will be performed on a yet-to-be-determined digital random access editing system.

Regardless of the high sonic quality of the original production, Magee is uncomfortable with the broadcast chain that will follow. "We used to have concert feeds which would originate from our master to the satellite at a precise time and stations would take the downlink and go directly to air. But that's not done much anymore. Most of the feeds are now taped by individual stations," Magee said.

"If I'm lucky there's someone at a station with equipment that's fairly respectable ... maybe he has Dolby SR, 15 ips; maybe he has a good-sounding or modified DAT; maybe he has a tweaked-to-death F1 ... something respectable. Even if a guy is using Dolby A, 15 ips, with good tape on Studer machines, I'm still happy. But when we get involved with those facilities using 406, 206 and 207 with no noise reduction, even on a good machine ... Come on, we've got problems.

"In order to do really great analog, it's going to cost you about three times as much as digital."

"This is classical music ... the dynamic range is excessive, signal to noise is going to be imperative and the quiet passages are going to end up in the noise floor. This is something I don't really want to think about. There are some stations out there who really care a lot."

READERS FORUM

If you have comments for *Radio World*, send a letter to International Readers' Forum, Radio World, Box 1214, Falls Church VA 22041 USA. All letters received become the property of Radio World, to be used at our discretion and as space permits.

Long-wave AM solution

Dear RW:

Most of the solutions proposed to "save" AM broadcasting appear to be dead-ends which will fail to retain and attract audiences. The two most significant technical problems facing AM are, quite simply, noise and interference. Until these two problems are eliminated, audience erosion will continue.

Because spectrum space is extremely valuable, I strongly advocate a radical national policy which:

Promotes US government support of a program assigning the frequency range of 150 kHz to 280 kHz to broadcast.

Requires conversion of existing AM radio stations to FM transmission in the same band.

Requires the manufacture (or importation) and sale of appropriate receivers to be available for new transmission standards.

Let's look at how AM broadcasters will benefit:

1. The "long-wave" band noted above has long been assigned to broadcasting in Europe, Africa and the Middle East. This band is little used here except for navigational purposes. I propose using the band to develop the first "new-band" FM broadcast stations. Use of the "long-wave" band for development (and eventually) full broadcast service also provides the lead time to legislate, produce and market suitable receivers.
2. Once the "long-wave" band using FM is developed and receivers available, today's AM broadcasters would be required to convert to frequency modulation, on the existing "medium-wave" band, perhaps over a period of one year.

   I suggest that this is also an appropriate time to strive for "international" standards in long and medium wave FM broadcasting, such as 9 kHz channel spacing to match the majority of world broadcasters. It also provides additional channels within the existing medium-wave band.
3. The inherent noise-suppression and capture-ratio effects of FM would help eliminate the problems which will continue to plague AM.

I believe radio broadcasters should admit to themselves that standard AM will never sound the same ("as good") as VHF frequency modulation. It cannot simply because of bandwidth requirements. What I propose gives today's AM broadcasters a chance to eliminate or reduce the two most significant problems which drive away listeners and advertiser revenue. Other solutions already proposed are merely bandages on a wounded and dying public service.

Donald M. Sites
Springfield, Virginia USA

In support of long-wave

Dear RW:

Even though I am now involved in shortwave broadcasting I still like to keep up with current happenings in commercial AM/FM radio. I read with interest the "long-wave AM solution" letter from Donald M. Sites. I have also considered the long-wave radio idea. It might be the best place for true regional clear-channel stations.

I was particularly pleased to read of his mention of "FMing" the AM band. A couple of years ago, George Yazell, a retired professional engineer in Florida, came up with not only the idea but the method for a complete FM signal, stereo and subcarriers included, on the AM band. The system—NFR, for Noise Free Radio—works, but does not have support of the National Association of Broadcasters (NAB). I have noted with awe how fast the NAB has been able to work toward AM improvement.

Mr. Sites is right. "Other solutions already proposed are merely bandages on a wounded and dying public service." Perhaps a long-wave band would be a good way to implement the FM method.

Tim Coucke, Staff Engineer
Shortwave Station KNLS
Anchor Point, Alaska USA

From The Publisher

Welcome to the first issue of **Radio World's** new monthly international edition. To properly introduce ourselves, let me first tell you a little about **Radio World's** USA edition.

Founded in 1977, **Radio World** newspaper has grown to become the leading technical radio industry publication in the USA. Published twice a month, and averaging 60 tabloid pages per issue, **Radio World** has a loyal following of over 20,000 readers throughout the USA. Each issue is packed with news, feature articles, equipment reports, new product introductions and classified ads — much like you will find here.

The international radio market has always interested us. That is partly because many of **Radio World's** advertisers are either American companies that export equipment, or foreign-based companies with USA offices. We also feel that no other publication serving the worldwide broadcast industry meets the specific needs of radio broadcasters. As you are probably already aware, all of these publications are dominated by television-related editorial and advertising.

Radio World, on the other hand, is dedicated to covering the technology of radio broadcasting. And we are all radio, all the time.

Most of the articles you will find here have appeared in **Radio World's** USA edition. In the future, as our international edition grows, we will incorporate more coverage of radio broadcasting in other countries. To that end, we invite you all to participate by sending news releases and articles about your own station's technical innovations (see box below). We would like to hear from you.

We hope you enjoy this first international edition of **Radio World**, and that you will look forward to receiving it each month. To ensure your continued free subscription, please fill out and return the subscription card enclosed in this issue. Or, if you prefer to receive your copy more quickly for a nominal fee, fill out and return the air mail subscription form found on page 19.

In the meantime, let us know what you think about Radio World, and how we can make it better for you. We are listening!

Blow Your Own Horn

Write for Radio World

The world's only publication devoted entirely to radio station operations is seeking articles on interesting and unusual applications of technology. Tell a worldwide audience about technical innovations at your station. We're interested in news of new product applications, maintenance techniques, tech tips, etc.

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If we publish your article, you will receive a special American Express Gift Check, valued at US$50.

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A Basic Primer for Digital Interface Use

by Mel Lambert

Studio City, California USA It is often all too easy to overlook the fact that digital technology, *per se*, doesn't just mean elegant boxes that do such a stunning job of digitizing, storing, processing and replaying stories, music, drama and other essential radio material.

Before there were affordable 16-bit PCM recorders, digital meant dedicated microprocessors and general-purpose computers for controlling automation systems, signal routing, billing and other chores around the station.

Understanding digital recorders

Remember that, in reality, a digital recorder or processor is nothing more than a very fast, custom designed minicomputer. Its sole task in life, rather than process complex spreadsheets, is to convert audio into a string of 16-bit numbers, store them on a magnetic/optical medium and then replay them for our delight and edification. The real magic came in reducing the cost of such systems to a point where we could afford to use them outside of a well-funded R&D lab!

It occurs to me that our current appetite for affordable recorders and editing systems for air and production studios means more than cultivating a knowledge of general digital interface schemes such AES-EBU, S/P DIF, SDIF-2 and the other lexicons of our age. Most CD players, DAT machines, editing controllers and other recent components of the all-digital studio feature serial and/or parallel control ports, whose origins date from the early days of computer connectivity and networking.

With just a modicum of engineering talent and ingenuity it is possible to fabricate a customized system that can streamline some of the more mundane button-pushing and stop/start functions we encounter every day around the station.

Most devices now sport some form of remote control. Parallel, switch-closure type remote ports, while useful, are usually pretty restrictive in the kinds of functions we can effect from a distance. (Not to mention the fact that we need a single connection per function; a cable trunk the diameter of your wrist is probably not the way to go if you need to access every front-panel button and keypad command!)

A better and more compact approach is offered by serial interfaces of the type we see on a growing number of CD, DAT, reel-to-reel, NAB cartridge and other devices. Using simple commands, running at moderately fast bit rates, we can implement a host of useful functions from a distance.

The majority of equipment we are likely to encounter in the studio features RS-232 or RS-422-type serial interfaces. Designed originally to enable computers to interconnect with a range of peripheral equipment, including terminals, printers, modems, plotters, scanners and other input/output devices, we can now use the same hardware and communications protocols to stop/start and otherwise manipulate audio hardware.

Standard RS-232/RS-422 connections can be run via simple two/three/four-wire circuits (422 is, in essence, an electronically balanced version of 232, designed for use in more "hostile" RFI environments, and over longer distances) using simple, two/three/four-byte commands.

Two scenarios immediately present themselves. As a first-stage implementation, consider fabricating dedicated boxes that can issue commands in response to pre-designated sequences of front-panel button pushes from a mixing console and/or which can direct the appropriate commands to the required CD or DAT machine located in a remotely located rack system.

Subsequently, such a configuration can be easily upgraded to accommodate a central computer which, with suitable software, will extend the programmable capabilities within a radio facility.

Applications

Typical applications might include the setting up of an automated replay sequence using a collection of CD players or pre-recorded DAT tapes; controlling various CD, DAT and conventional tape machines in a full-on production suite from one remote control panel, rather than cluttering the console with a bunch of different remote boxes; or maybe automating routine machine alignment tasks around the station, utilizing a microprocessor-controlled test station and an off-the-shelf general-purpose PC.

DIGITAL DOMAIN

A simple configuration might comprise a controller located at the sound engineer's position that could control, via a serial interface, the primary transport and front-panel functions of a CD or tape machine. A medium-complexity configuration might involve a master controller that could set up to control transport, record/replay and other functions for various "families" or subsets of machines.

Master controller scenario

Moving on to an even more complex scenario, we might need to establish a master controller that would be able to select a specific CD or DAT machine, for example, from among a family of dozens of transports that can be selected for on-line use, and then maintain full remote control of its various functions.

If such a system reminds you of an IBM PC or Apple Mac connected to a couple of printers, with keyboard selection of routing the current output to either a dot matrix, letter-quality or laser printer, that is hardly surprising. The same type of serial interface we currently use between these various units can also be used to provide on-line control of studio hardware.

The one possible fly in the production ointment is that suitable software to implement remote control of serial-equipped CD players, DATs and other hardware isn't exactly available off-the-shelf at your local computer store. But, having dabbled recently in the development of a remote control system for a well known brand of DAT transport, I can tell you from personal experience that some powerful tools do exist to simplify your life, at least partially.

Sure you can write the software in BASIC, FORTRAN, PASCAL, C or one of the other higher-level programming languages. (I doubt that you will need to resort to machine or assembly code, simply because the speeds at which we need to run such serial-control software are relatively modest.)

What often happens, however, is that writing the program section that extracts switch closures from the keyboard, compares them with a look-up table, and then executes the appropriate commands, can take maybe five minutes, while the development of a suitable screen display can often be extremely time consuming.

HyperCard solution

One way around the problem of writing various menu displays that ask questions of the operator and then display the results is to use one of the newer dedicated software languages being developed for hypermedia applications.

One excellent example of the genre is Apple Computer's HyperCard software, which utilizes a very intuitive language known (not unsurprisingly, perhaps) as HyperTalk. Based on simple on-screen button icons, keyboard clicks and/or inputs to one of the Macintosh's pair of serial ports, it is possible to very quickly develop a series of sophisticated control programs.

In essence, each of the various screen panels, or "Cards," can contain a variety of text display windows and buttons. A series of "scripts" written in the easy-to-learn HyperTalk programming language let you write sequences of commands that implement complex input/output routines for controlling virtually any unit equipped with a serial port that communicates via RS-232/422 interfaces.

The information flow can also be bi-directional. Certain audio and video decks, for example, output information that can be used to display current motion status and shuttle speeds on the Mac's display screen.

And, with the addition of a simple serial-to-MIDI interface, we open up an entire world of remote control of MIDI-equipped signal processors, mixing consoles and effects units.

With such technology you could develop a system that automatically inserts pre-set EQ profiles into the signal path for different DJs or voice-over talent. Or what about being able to switch-select various compression slopes, attack and release times via remote control?

And your future might even include a multi-channel mixing and processing workstation, whose myriad MIDI-and serial-based functions can be controlled entirely from a CRT using a mouse and a few keyboard instructions.

I realize that not everybody has the time and/or inclination to develop sophisticated software for controlling such hardware around the station. But if you are looking for a way to implement a flexible, easy-to-program, relatively inexpensive system for remote-controlling a variety of recording and production devices, a Macintosh running customized HyperCard serial-control software might bear some additional investigation.

■ ■ ■

Mel Lambert has been intimately involved with the production and broadcast industries on both sides of the Atlantic for the past dozen years. To comment on this article, write him at Radio World International, P.O. Box 1214, Falls Church, VA 22041 USA.

Space: The Final Audio Frontier

by Ty Ford

Baltimore Maryland USA Production Rat shifted his weight under the electric blanket. He was not quite awake yet. It was in this semi-consciousness Alpha state that many great ideas came to him. He sometimes wished he could arrange to spend the whole day this way.

The clock radio popped on and yanked him into full alert. The station was in a spot break. Even though the audio was playing through a cheap two-inch speaker, Production Rat (PR to his friends) analyzed every element; agency spot, hired voice, retail copy ... not bad. Too much reverb! When would people realize that too much reverb made a spot sound really retro and cheesy!

The BASE processor, from Bedini

It wasn't that he didn't like effects processing. In fact PR often used multiple processing during different stages of production. Applying a little reverb on the voice track followed by just the right radical early reflections later in the mix created strikingly different ambiances. The secret was in how to combine several different effects rather than a lot of just one.

After years of practicing his craft, Production Rat had gained an appreciation for this kind of subtlety. It not only made his work sound better, it drove the competition crazy. Because he never overused any one effect, it was almost impossible to tell what made his work sound the way it did. It stood out.

Even if he used no music at all, voice tracks he recorded had a certain sound to them that had nothing to do with intonation, projection, articulation or enunciation. He had learned by experience how to vary the effects slightly so each project had its own sound.

Voicing complaints

When it came to doing his job, one of PR's biggest complaints was production music that was arranged so that certain instruments ate up space that should have been left open for a voice track.

The worst was when a producer would put a wailing saxophone solo or killer snare drum dead center in a mix. The only way you could get a good voice-over was to drop the level of the whole bed so the instruments didn't get in the way. When you did that the music wasn't strong enough.

Production Rat had tried notch EQing the music track to make room for the voice track. With a parametric equalizer he experimented with reducing the gain of the music track between 200 Hz and 4 kHz—where most of the voice frequencies were—to make room for the voice. It worked ... sometimes.

What he really needed was a box that would let him get the center channel, or mono signal, under control. And because each piece of music was different, the control had to be variable.

The BASE experience

Did such a box exist? It did! It's name? The Bedini Audio Spacial Environment, or BASE. Bedini, as it turns out, has been somewhat of a pioneer in the "three dimensional" audio movement. If you're an avid reader of movie credits, you've probably seen the box credited on a number of major films.

Much of the Bedini box is the result of work based on the theory that, "more information exists on audio source material than the brain can assimilate through current playback equipment. The BASE processor allows you, the listener, to hear more of the ambient acoustics that were present during the actual recording."

Inputs and outputs on the unit are unbalanced TS jacks or XLRs (pin 3 hot, 1 and 2 ground). The BASE unit also has a mono side-chain loop accessible by ¼" jacks on the back panel which allows independent processing of the extracted mono audio. Stated input impedance is 47 kilohms, output impedance is 47 kilohms into 600 ohms. THD is .0025%, 20 Hz to 20 kHz. Maximum input before clipping is 2.5 volts RMS. This is a single-ended device, which means it doesn't require a second decoding box.

A close look at the BASE processor reveals some simple yet sophisticated circuitry. The BASE processor separates the mono information from an incoming stereo source. It then allows you to vary the gain of the mono signal as well as pan it across the stereo outputs.

The phase of the stereo signal is reversed before it is fed to the "stereo space" control. As you increase the "stereo space" control the phase reversal causes the stereo image to widen.

Unlike reverb and delay, the BASE widening effect disappears when the signal is combined to mono. Also, there is a slight decrease in what was center channel information when processed stereo is combined to mono. This means that the levels you set while recording in processed stereo still work if heard in mono.

PRODUCER'S FILE

As might be expected, the effect is more noticeable when listened to with headphones. When the "stereo space" of a stereo music track increases, strange things start to happen. The perception is that the music is being pushed toward the ends of the stereo spectrum, leaving a space in the middle.

As you max out the "stereo space" control, it sounds as though the music is pushed up the walls of an imaginary container. It's as if the music has a finite spatial volume which acts like a physical mass. In visual terms, it's like the effect they used when Moses parted the Red Sea in the movie *The Ten Commandments*.

Hole-y music track, BASEman

So now you have this extra wide stereo music bed with a hole in the middle. You can fill the hole back in with music by increasing the "mono gain" control, or by mixing in other audio sources ... like a voice track. If you choose to fill in the gap with music using the mono gain control, you also have the option of panning the mono anywhere on the stereo spectrum.

The amount of separation depends a lot on the nature of the stereo signal you're using. In one particular case, a Paula Abdul CD was fed to the BASE processor. The output was then fed to stereo line inputs on a mixer and played through a set of studio monitors about 15 feet apart. A/B comparisons made it obvious that separation had increased. So much so that a moderate amount of "mono gain" had to be used to fill in the gap.

Another noticeable effect was that the intricate percussion overdubs seemed more defined. The increased definition was partially due to a perceived increase in the 5 kHz to 10 kHz range, perhaps a by-product of the phase twisting. While this improved the ability to hear the leading-edge transients of those instruments, it also brought Paula's vocal closer to the sibilance threshold.

The increased separation seemed to spread out instruments in the mix. Percussion instruments that were almost on top of each other in the unprocessed mix began to move apart as the effect was increased.

Next stop was the voice-over studio. Stereo CD production music was fed directly into the BASE processor, which was then fed to stereo console inputs. A mic was brought up on another channel. When the processing was kicked in, the music track moved outward in both directions from the center channel, leaving a nice space for the voice.

When processing was removed, the music became so loud that it interfered with the voice track and had to be lowered.

This phase of the experiment had been done under headphones; however, we had been rolling tape on the mix. We

(continued on page 10)

Coming Events

April 10-18

Telekinoradiotekhnik-90
Expocentr, Moscow. Contact: Telekinoradiotekhnik-90, Expocentr, Sokolnicheski val, 1-a Moscow 107113, USSR.

May 17-20

Asian Broadcast '90
Singapore. Contact: IMS Exhibitions & Conferences (Singapore) Pte Ltd, 190 Middle Road, 18-06A Fortune Centre, Singapore 0718. Phone: 3364611. Fax: 3366762.

May 29-June 1

Broadcast Asia '90
Singapore World Trade Centre. Contact: Overseas Exhibition Services Ltd, 11 Manchester Square, London W1M 5AB. Phone: 01-486 1951. Fax: 01-935 5637.

June 6-8

APRS '90
Olympia 2, London. Contact: Association of Professional Recording Studios Ltd, 163a High Street, Rickmansworth, WD3 1AY. Phone: 0923 772 907. Fax: 0923 773 079.

July 11-13

Pro Audio Asia '90
Wanchai, Hong Kong. Contact: Business and Industrial Trade Fairs Ltd, 28/F Harbour Centre, 25 Harbour Road, Wanchai, Hong Kong. Phone: 5-756 333. Fax: 5-8 341 171

To have items included in this section, send information to: Radio World International Edition, Calendar of Events, P.O. Box 1214, Falls Church, Virginia, 22041 USA.

An Introduction to Sampling Devices

by Bruce Bartlett

Elkhart, Indiana USA Buh-Buh-Buh-Buh-Believe me, the stuttering sample effect defines the contemporary sound of many remote DJs and production-music composers. A sample is a memory-chip digital recording of a single note or sound. You can capture any sound you hear and make it part of a DJ production, or part of your station's musical spots. Here's how:

LINE OUT

Using a microphone, you pick up a single note of an instrument, voice, or sound effect—a violin, electric guitar, drum, dog bark, thunder and so on.

The microphone signal goes into a sampling keyboard, which is an electronic musical instrument with a piano keyboard and computer memory circuits. Inside the sampling device, the signal is converted into digital data and is stored (recorded) in computer memory.

To play the sample, you press a key on the keyboard. You'll hear a note of whatever you recorded. The higher the key you press, the higher the pitch of the reproduced sample.

How sampling works

Let's get into the details of how samples are made. First we need to define the term "computer memory." This is a group of integrated-circuit chips, each containing thousands of solid state switches. Information is stored in binary format (1=switch ON; 0=switch OFF).

Each 1 and 0 is called a bit, which stands for binary digit. Memory stores bits of information. Memory space is limited and is measured in bytes, where 1 byte equals 8 bits.

Suppose you want to sample a live sound, such as a cymbal crash or train whistle. You plug a microphone into a sampler or sampling keyboard. It records the sample as described below (and shown in Figure 1).

The signal from the microphone passes through an analog-to-digital (A/D) converter. This converter measures the voltage of the audio waveform several thousand times a second. Each time the waveform is measured, a binary number (made of ls and 0s) is generated that represents the voltage of the waveform at the instant it is measured (Figure 2). These binary numbers are stored in RAM (Random Access Memory), making a digital recording.

Later, the sampled sound is played back by pressing keys on the piano-style keyboard (Figure 1). The keypress triggers the sample. Which key you press determines the reproduced pitch of the sample. That is, different keys cause the sample recording to play back at different rates, shifting the pitch of the sample.

However, too much of this pitch shifting can cause an unnatural sound. That's because the pitch-shifted notes have the same harmonic structure for low and high notes, while notes from a real instrument have different harmonic structures for low and high notes.

Sample sources and hardware

Instead of having the entire keyboard control the pitch of one sample, it's better to record several samples at different frequencies—say one octave apart—and control the pitch of each of these samples within a smaller range. This procedure, called multisampling, is explained in your sampling keyboard instruction manual.

There are many sources of samples. They can be recorded live with a microphone, or can be recorded off records, compact discs, tapes, etc. Recordings of notes can be made first on an analog or digital recorder, then transferred to a sampler or sampling keyboard.

Compact discs are available that contain notes of various instruments to sample. Pre-recorded samples on magnetic disk and plug-in ROM (Read Only Memory) cards are offered for many sampling keyboards.

An alternative to a sampling keyboard is an external sampler. It is triggered by a separate keyboard or sequencer. Another alternative is a computer running a sampling program.

Sample-playing keyboards play prerecorded samples, but do not record them. This is a practical option because most keyboard users rely on factory or third-party samples and do little sampling of their own.

In general, sample-playing keyboards are more useful than synthesizers. It's easy to make a sample sound like a synthesizer, but it's difficult to make a synthesized note sound like a sample. Some instruments contain samples that are digitally resynthesized so that you can modify them.

Permanent storage of samples

You can store your samples permanently on a RAM cartridge that plugs into a sampling keyboard or on a computer magnetic disk. Once you've sampled many sounds and have built up a library of them, any sample can be loaded into your keyboard and played.

Some keyboards and all drum machines have factory supplied samples stored in permanent memory (ROM or Read Only Memory); these are digital recordings of real instruments stored in memory chips.

Three parameters to consider in sampling are quantization, sampling rate and memory constraints.

Let's explain quantization first. As stated earlier, the audio signal is measured several thousand times a second to generate a string of binary numbers. The longer each binary number is (the more bits it has), the greater the accuracy of the measurement.

In other words, short binary numbers provide poor resolution of the waveform's amplitude or voltage; long binary numbers provide good resolution.

The quantization of a sampler is its amplitude resolution, measured in bits. The higher the quantization, the less the distortion and the greater the dynamic range. Commercial samplers range from 8-bit to 16-bit quantization. Quantization of eight bits is good, 12-bit is very good and 16-bit is excellent.

The rate at which the waveform is measured is called the sampling rate, measured in samples/sec. At a sampling rate of 40 kHz, 40,000 measurements are generated for each second of sound.

The higher the sampling rate, the wider the frequency response of the recording. The upper frequency limit is slightly less than half the sampling rate. If the sampling rate is, say, 20 kHz, the sound you sampled will be reproduced up to about 9 kHz.

High-frequency sounds (cymbals) need a high sampling rate for fidelity (about 40 kHz); low-frequency sounds (bass, kick drum) can be recorded adequately with a low sampling rate (about 12 kHz).

While most sampling machines have

(continued on page 13)

Figure 1.

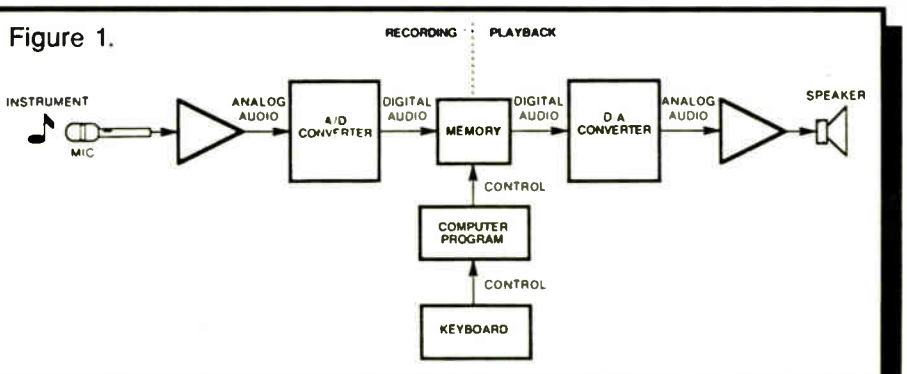
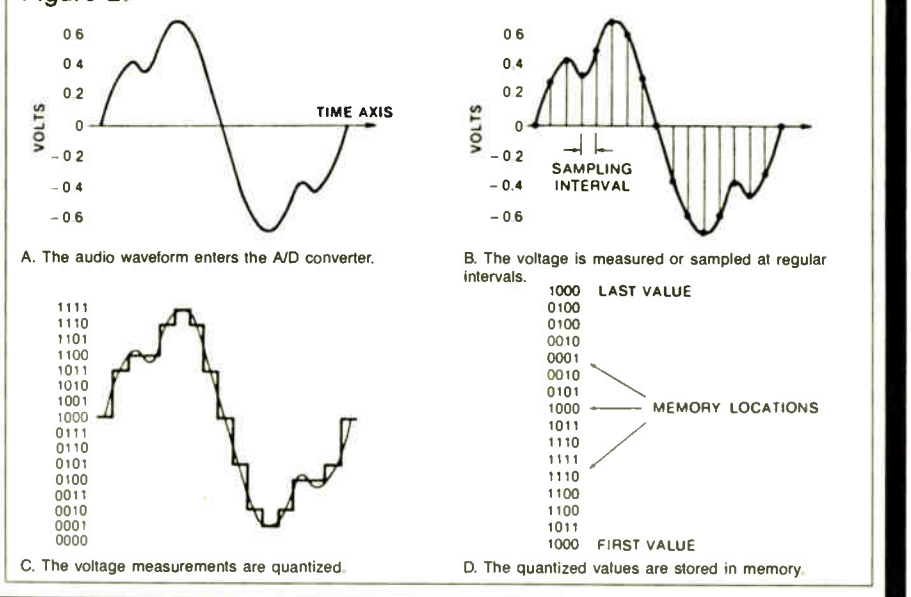


Figure 2.



A. The audio waveform enters the A/D converter.

B. The voltage is measured or sampled at regular intervals.

C. The voltage measurements are quantized.

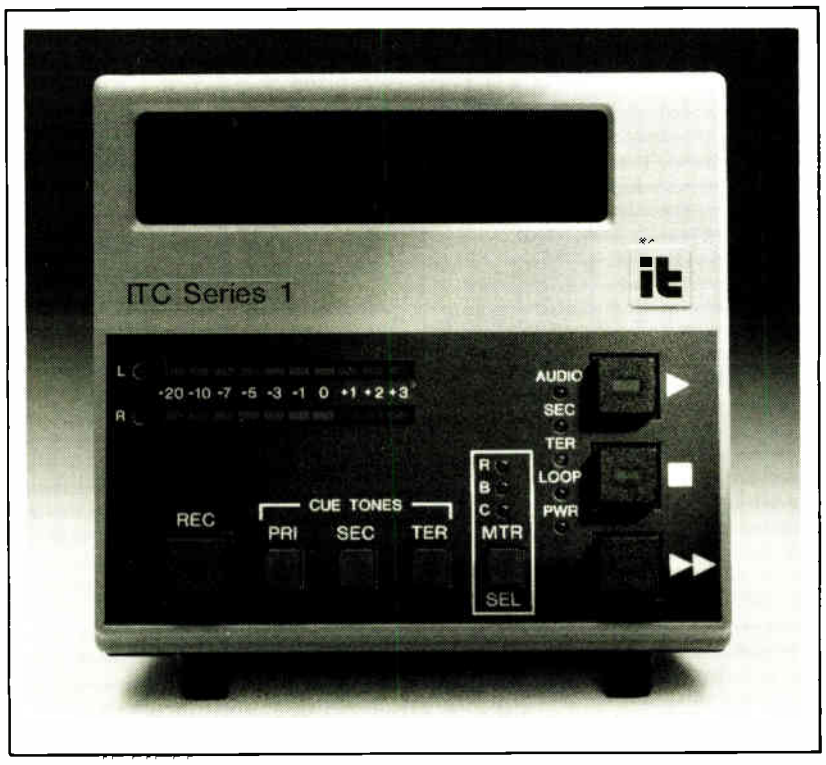
D. The quantized values are stored in memory.

Figure 3.

$$B = Q \times SR \times ST$$

where B = Bytes of memory filled by a sample
Q = Quantization in bytes/sample or bytes/measurement
SR = Sampling rate in samples/second or measurements/second
ST = Sample time in seconds

If Your Only Excuse For Not Buying An ITC Cart Machine Has Been Money, You Just Ran Out Of Excuses.



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Circle 32 On Reader Service Card

Airwaves Penetrate the Wall

(continued from page 1)

lems; if you know things like this, you cut the communications."

But now, today, Eckhardt is being inundated by letters from his fans on the other side of The Wall. A communication that, up until recently, would have raised the ire of East Berlin officials.

Exchange of programming

Robert Barett was a full-time engineer at WTXF-TV and a part-time disc jockey at WCZN-AM in Chester, Pennsylvania until recent health problems brought him out of the radio arena. He has raised the ire of some of these statehood officials.

(Left) Robert Barett at WCZN radio.
(Below) Joint German/American radio in West Berlin.

Barett and Eckhardt, both country music enthusiasts, met some years ago while Barett was visiting SFB. The two began exchanging weekly radio programs, a novelty that has stretched across the Berlin Wall on more than one occasion.

Barett's show, aired by SFB in exchange for Eckhardt's show which was aired by WCZN on Sundays, had developed a loyal listenership in Germany—some of it behind the Iron Curtain. "I was always getting mail from East Berlin," said Barett, "long before The Wall came down."

On the show's second anniversary in June, Barett flew to West Germany to do the show live. During his broadcast, Barett greeted an East German country music fan, addressing him by name over the air. Not until later, when Barett decided to pay a visit to East Berlin, did he learn of the problems his broadcast had caused his East German country music fan.

Arrested for possession of Pepsi

After crossing The Wall into East Berlin, and being arrested for possessing two cans of Pepsi and then questioned about his radio show on SFB, Barett was told that the East German country music fan had been arrested. "He was a member of the Communist Party," said Barett, "and Party members are not supposed to talk or fraternize with the people from the West."

Although the East German listener and Barett were eventually released, the experience was, as Barett put it, "harrowing." And, there were other run-ins because of the Berlin Wall, he said.

During his visit to West Germany last February Barett applied for permission to enter Brandenburg, East Germany. "I had met three people from a country music club who were living in Brandenburg," he explained. "I wanted to visit them there, but they (the East German government) wouldn't give me permission to go there." The unofficial reason, claimed Barett, was that the East German government did not want the media entering the city.

"Apparently," he said, "they felt something was wrong with having the media come into Brandenburg, which is not a tourist city. Media people just didn't go there. They were suspicious of that." After a political ballet with East German officials that lasted months, Barett finally was able to meet his fans in East Berlin.

Barett's experiences, said Eckhardt, are typical of the communication barrier that once existed between East and West Germany. Even Barett's enthusiasm for country music was frowned on by East Germany's government, which had barricaded all Western forms of art including country.

Prior to the dismantling of the Berlin Wall on 9 November 1989, according to Eckhardt, "there was absolutely no communication between stations in West Berlin and stations in East Berlin."

German Schlager

To understand the significance of what has just taken place, Eckhardt introduced a German word: "Schlager," a form of music that is similar in lyrics to American country music—with similar ties to culture and heritage.

When the wall was erected after World War II, East Berlin was shut off from the Western world, and later from its Schlager music. Described by Eckhardt as "laugh music," Schlager was eventually reintroduced to East German listeners, but its style had changed drastically.

Reported Eckhardt, "They were not allowed to play swing music, or Glen Miller, things like that. Later on they didn't want to have Schlager either. And they tried to do other things (with music), but nothing worked out very well. So in the '60s and '70s, they did their own style of Schlager."

Schlager is now an open curiosity being explored by Germans. But it is not the only curiosity of two differing dominions. For radio, there are the matters of production, recording and in general how broadcasting is done on both sides.

'Today," said Eckhardt with the awe of someone who hardly believes his own words, "I had a visitor from an East Berlin radio station and he came to my boss to say, 'let's sit together and let's talk about how you do your programming and things like that.'"

Just a few months prior, this open line of communication was unthinkable.

World in motion

In full motion today are the letters—thousands of them—that SFB is receiving from East Germany. Listeners that were once cloaked behind the Iron Curtain are now becoming a part of West Berlin radio. And Western music that was once hands-off for East Berliners is now entering the Eastern block through the radio stations that they once listened to, but never dared talk about.

And sometimes, said Eckhardt, "for hours—or a day—we have the same programs." That, exclaimed Eckhardt, is also "amazing." It's an indication that while he might literallly have been talking to a brick wall all these years, the listeners behind The Wall were, in fact, tuned in.

(Above) A view of SFB taken from a tower in West Berlin.
(Below) An East Berlin tower.

Nothing Quite Like BASE

(continued from page 7)

were curious to hear if the effects were as audible on studio monitors as they had been in our cans. Even though the separation was not quite as apparent in the monitors, the notch left for the voice track was very obvious. When the processing was removed, the music crept up on the voice track.

Because the unit allows separate control of mono and stereo information—and because most vocals are mono and most music is stereo—you can change the relationship between the voice and music levels on an already mixed stereo master!

It's worth mentioning that the BASE does its processing without noticeable noise increases.

Jim Harmon of Soundwave, Inc. in Washington, DC has been using the BASE processor in his studios for the last year. He commented that it was especially useful in creating a larger sound when the processed audio was to be played back in an acoustically confined area. He also found that stereo sound effects became more apparent when processed by the unit.

Listing at US$3000, the BASE processor is more expensive than the average piece of procesing gear. For the moment, however, there is nothing else quite like it on the market.

■ ■ ■

Ty Ford is an audio production consultant and voice talent. To comment on this article, write him at Radio World International, P.O. Box 1214, Falls Church, Virginia 22041 USA. For more information on the BASE processor, circle Reader Service 30.

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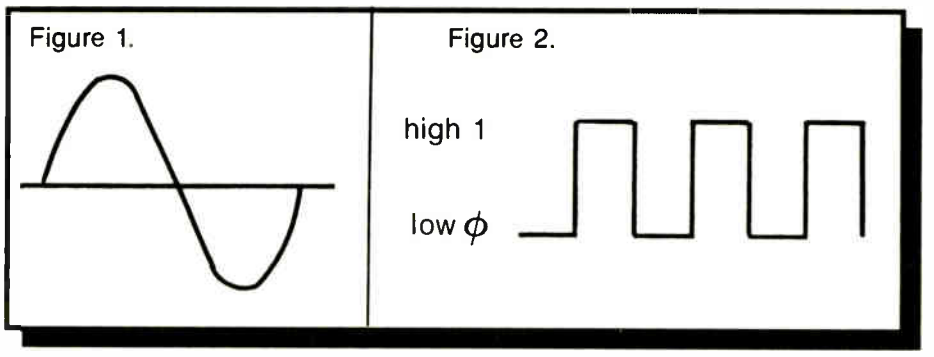
Circle 61 On Reader Service Card

Origins of Today's Digital Technology

This month, we begin a 12-part series titled, An Introduction to Digital Electronics. Parts I and II are presented here. Look for the continuation of the series in next month's issue. Ed Montgomery currently is an electronics teacher at Thomas A. Edison High School in Virginia. To comment on this series write him at Radio World International, P.O. Box 1214, Falls Church VA 22041 USA



It is now important that all individuals working in electronics have an understanding of digital circuitry.

by Ed Montgomery

Part I of XII

Annandale, Virginia USA The latter portion of the twentieth century has brought great changes to the world of electronics. The field has changed from one concerning itself with linear operation of active devices to a controlled nonlinear operation identified as digital electronics.

Fundamentally, digital systems are electronic devices that operate in two discrete states. This principle combined with the development of the integrated circuit is the driving force behind pocket calculators, clocks, personal computers, video and audio discs, and state-of-the art transmitters.

It is now important that all individuals working in electronics have an understanding of digital circuitry.

Analog versus digital

For years, linear or analog operation was the standard in electronics circuit design and operation. Figure 1 is an illustration of one cycle of a sine wave. An analog circuit will take this signal and attempt to reproduce its electrical signal variations at every instant throughout its cycle. The voltage and current within an analog circuit can be said to vary continuously.

A digital circuit operates at defined electrical levels. Figure 2 is an example of what a digital circuit might look like. Digital circuits are designed to handle signals only classified as "high" or "low" as illustrated in Figure 2. These levels are often referred to as "0" for low and "1" for high.

This rather simple form of circuit design has resulted in circuits with a high degree of reliability and accuracy in very small packages. The development of this technology has brought about programmable watches, robotics and synthesized speech. Test instruments relying on digital technology include multimeters, counters and oscilloscopes.

In communications one must remember the digital technology employed in Pulse Duration Modulation transmitters. This technique has improved transmitter audio at high power reducing harmonic distortion along with the amount of power required to run the transmitter.

Much of the digital technology associated with communications has its roots deep in the history of electronics. One of the first circuits employing electronic circuitry surrounding the concept of operating at defined levels is the television sync generator.

Birth of the sync generator

The sync generator was developed in the 1930s by RCA when it invented the totally electronic scanning system. The original generator required several racks of equipment. Most of its functions can now be performed on a small chip.

In the early 1960s, Gates developed the first direct FM broadcast transmitter. This system employed "serrisoid modulation" converting analog signals into a series of pulses resulting in a frequency modulated signal.

This technology employing sawtooth waves and triggers is very similar to PDM technology in present day AM transmitters and FM systems going all the way back to Edwin Armstrong's research. The method of encoding an analog signal into frequency deviations contains much of the technology surrounding the digital audio in use today.

How to Test for a Digital Signal

by Ed Montgomery

Part II of XII

Annandale, Virginia USA Once there was a time when an electronic circuit could be diagnosed with a multimeter. An oscilloscope was handy to have, but the multimeter could give a technician most of the information needed to solve a problem.

Digital electronics is a different matter. Circuits are always operating in one of two states. Figure 1 is an illustration of what the two conditions are.

The circuit is considered to be high (1) or low (0). Low is usually in the range of 0 volts or the ground potential of the circuit and high is usually in the range of 3 to 5 volts. Anything in between these levels will be considered to be undefined. Thus, if for some reason a system is incapable of reading in the "high" voltage level range, the circuit will not function.

"Baud rate"

The high and low pulses are sent in sequences. This is often referred to as the "Baud rate."

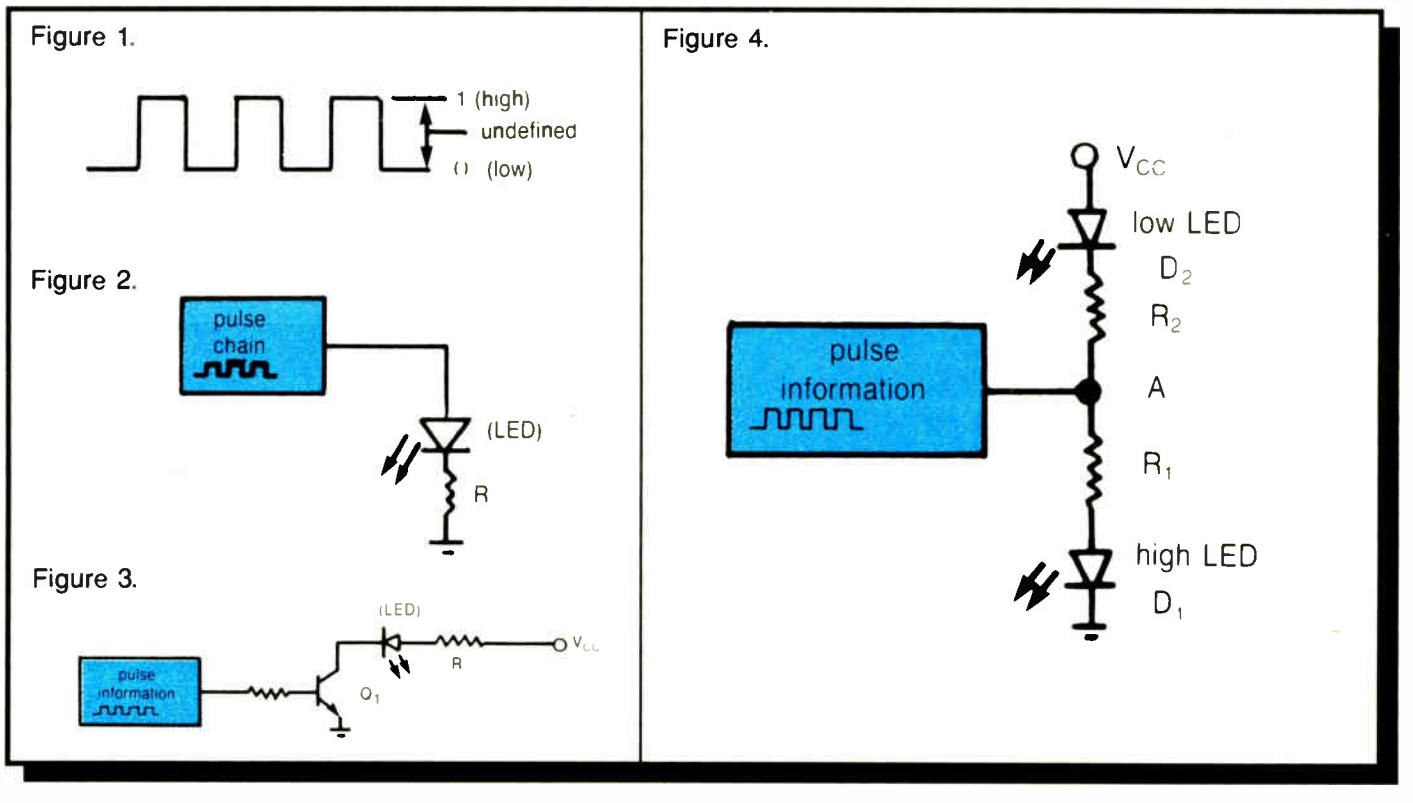
The term "Baud rate" originated from the Baudot-encoded teletypewriter that encoded signals for radio-teletype transmission. The system consisted of five information pulses (levels) encoding the alphabet, numerals and symbols.

The speed at which this information was sent was defined as its rate and expressed in words per minute or characters per second. Because the information is transmitted at such a high speed, a multimeter normally will not respond accurately to the level indications.

One simple indicator of a digital signal level is the light-emitting diode. The LED will respond to a high level and be illuminated (illustrated in Figure 2). The resistor in the circuit is a current limiter to protect the LED from damage.

A transistor can be used to drive the LED (illustrated in Figure 3). The purpose behind using the transistor rather than just operating the LED as demon-

(continued on next page)



Logic Probe A Must

(continued from previous page)
strated in Figure 2 is to reduce the current drain on the circuit. LEDs tend to draw much more current than the average digital circuit is capable of delivering.

Determining signal level

It is often convenient to determine whether a circuit is exhibiting a high or a low level signal. This can be achieved by using two light-emitting diodes. Using just one LED will work, but there will be doubt as to when a low level indication is present—in other words, is it really a low level or due to loss of power or component failure? For this reason, the illustration in Figure 4 will produce more accurate results.

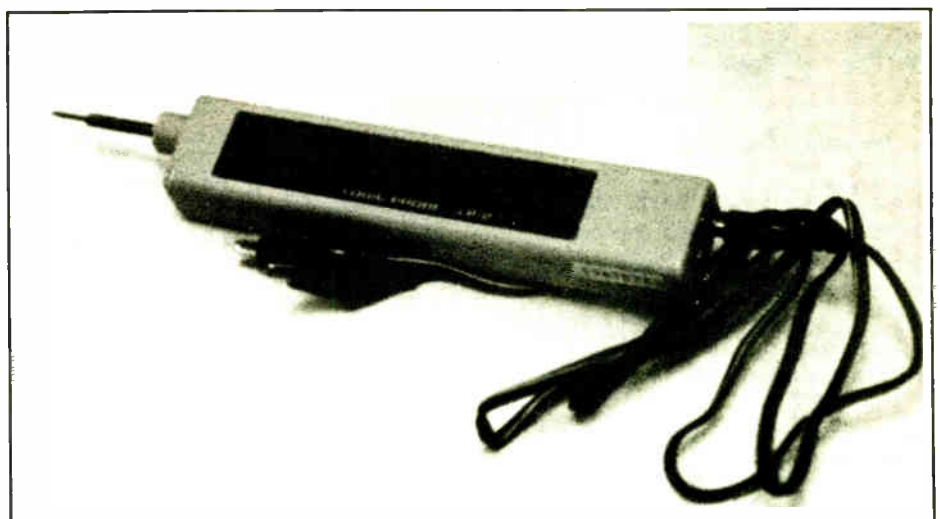
In this circuit, when the input is high, LED D_1 will illuminate and LED D_2 will be off. If the voltage at point A enters the undefined region between high and low, both LEDs will be illuminated. When the voltage output is at the low level, only D_1 will come on. Both high logic level and V_{CC} voltage are equal in this circuit.

Digital electronics equipment is often filled with light emitting diodes giving information regarding the status of the circuitry.

The test instrument that is indispensable in testing digital circuits is the logic probe. This device derives its power from the circuit being tested.

The logic probe has LEDs that indicate high and low levels, and pulse rates. The probe is capable of measuring voltages of TTL logic and CMOS logic. CMOS (complementary metal oxide semiconductor) can handle levels as high as 15 volts.

The logic probe has LEDs that indicate high and low levels, and pulse rates



A logic probe is used to test digital circuits.

Sampling

(continued from page 8)
adjustable sample rates, they have fixed quantization.

As the A/D converter generates binary numbers, they are stored in memory. Each number goes to a separate memory location. Unfortunately, memory space is limited. Once it is filled, part of the recorded note is cut off. This puts constraints on the sample time, sampling rate and quantization. Figure 3 provides an equation that shows how these four factors are related.

If you have a sampler with 8-bit (1-byte) quantization, and you set the sampling rate to 40 kHz and record a 2-second sample, you use up 1×40,000×2 or 80 kilobytes of memory. Since memory is limited, you try not to fill it up, because that can cut off the end of a sample.

You can economize on memory space either by keeping the sample time short or the sampling rate low. The higher the sampling rate, the more memory is used, because a high sampling rate generates more binary numbers than a low rate.

The longer the sample time, the more memory is used, because a long sample time generates more binary numbers than a short sample time. Stereo samples use twice as much memory as mono samples.

A one-second sample is enough for the notes of many instruments, but cymbal crashes may require three seconds or more.

■ ■ ■

Bruce Bartlett is a microphone project engineer and technical writer with Crown International. To comment on this article, write him at Radio World International, P.O. Box 1214, Falls Church, VA 22041 USA.

A Guide to Digital Cable Radio

by Douglas Conn

New York, New York USA The digitizing of consumer entertainment product is clearly the wave of the future. Both the audio and video media are undergoing a transition from analog to digital as new generations of technologies and distribution networks are introduced.

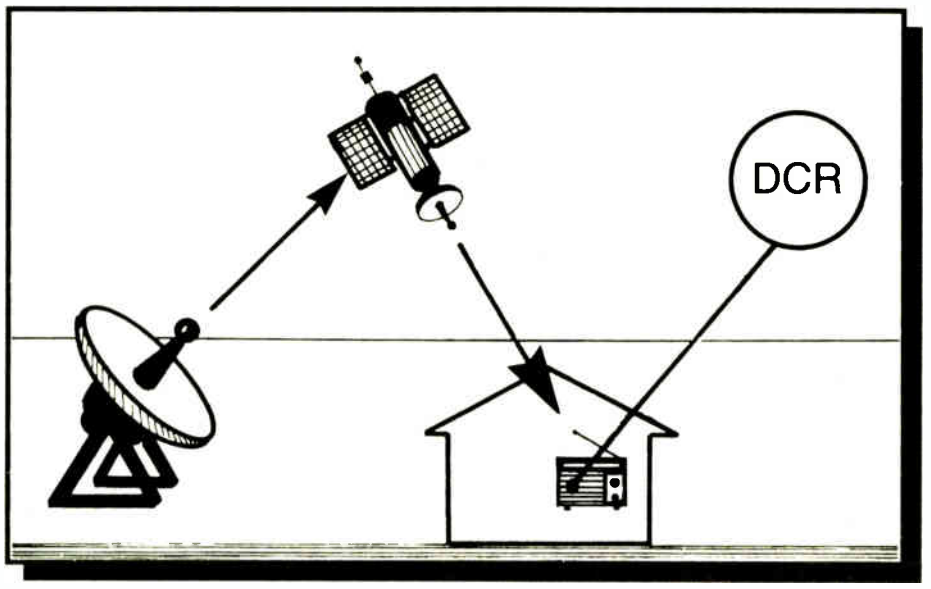
Remarkably, little attention has been paid to a new cable service, digital cable radio (DCR). DCR's potential lies in its ability to bring digital audio signals to a mass audience.

While it does not appear to be a revolutionary new application, nor will it soon overtake the popularity of broadcast analog radio, this new form of home audio delivery may have far reaching implications for at least three major industries: radio broadcasting, cable television and recording.

Digital inroads

As of late 1988, digital music signals were in about 12% of US homes in the form of CDs. Digital cable radio is a digital audio signal which is transmitted via satellite to local cable system headends and then piped into subscribers' homes where it is transformed to analog by a cable converter box and played through an ordinary home receiver.

DCR programming will consist of new packaged channels, digitally retransmitted radio broadcast stations, and digital simulcasts of program channels. The quality is said to be comparable to a CD because the digital signal travels locally over shielded coaxial cable lines, minimizing electrical interference.



But that is not the only dissimilarity between DCR and broadcast radio. The new DCR programming has virtually no disc jockeys and it is basically commercial free, leaving DCR programmers to collect their revenues directly from subscribers rather than advertisers.

Moreover, DCR channels will offer packaged music in specific sub-formats (ie. classic jazz, dixieland jazz, big band sound, new age, etc.). And DCR can be packaged on a pay-per-listen basic, similar to cable's pay-per-view, only here the subscriber pays to receive an album, live concert or pre-recorded music special at a specific time of the day.

Thus, DCR, with its many permutations of program offerings, is distinctly different from analog broadcast radio and may one day become a potent competitor.

Cable not new to radio

Cable radio is not by any means a new idea. In Switzerland, private wire radio was introduced in the 1930s to bring radio to mountainous Swiss terrain. Today, a typical Swiss cable system provides about a dozen television channels and up to 18 stereo radio channels.

In the US, non-digital cable radio has been available for some time and as an industry has just under 10 million subscribers, or 11% of total TV homes and 14% of total cable TV homes passed.

However, DCR's distinction lies in the superior digital quality of its signals. But, as the record industry has always known, consumers also want the ability to listen to pre-recorded music on their own schedule. The introduction of digital audio tape (DAT) will enhance this prospect.

As DAT players gain acceptance in the US, home taping via DCR will make it doubly convenient to acquire digital quality music. With this in mind, it is entirely possible that DCR will ride on the crest of DAT's popularity or stumble in the wake of its failure.

Concerns have been raised that, in some areas, cable radio stations, like cable video channels, will begin to flourish and their numbers will eventually keep broadcast radio stations off of cable systems or relegate them to less desirable slots on the dial.

This would be especially worrisome if listeners who subscribe to cable (60% of US TV homes) set up their home receivers so that they only have convenient access to radio through DCR.

These concerns though, are probably unfounded. Portability and mobility outside the home will always be important for radio listeners, insuring the longevity of broadcast radio.

Additionally, some digital radio receivers, such as Jerrold's, are extensible to 98 channels, leaving slots for the retransmission of broadcast stations. Also, an A/B switch could be employed to allow the listener to easily switch from cable to broadcast radio.

The cable television industry has much to gain from DCR. The additional fixed cost of new headend technology is minimal and the cable operator utilizes excess capacity to generate new revenues.

DCR may also enhance the profile of cable service offerings and, through digital audio simulcasts, may act to increase the value of current video services; however, the value added of digitizing the audio portion of premium or basic video channels, such as music channels, is unknown.

In the least, by digitizing audio signals, cable operators are helping to boost their image, both as purveyors of new technologies and as innovative service providers.

No copycode disputes

Finally, the recording industry should be taking a long, hard look at DCR, since its basic distribution methods are potentially greatly affected.

Pay-per-listen by itself may be successful but perhaps not until DAT is widely accepted and consumers can shift the times they choose to listen to programming through home recording.

Pay-per-listen and its eventual sophisticated step-brother, listen-on-demand/CD home jukebox, will allow consumers to listen or record digital music in their home conveniently while at the same time allowing the recording industry to be compensated on a consumer by consumer basis.

Arguably, the age-old dilemma of persons recording from others will not go away, no matter what the technology. But, the recent decision by DAT manufacturers to employ technology which prohibits the making of copies of copies will help to discourage rampant copyright violation.

DCR may even help bridge the gap between those who want to purchase music and those who are apt to record. The artist, record company and publisher will each be compensated at the point of purchase, accurately and quickly, and the number of real consumer outlets to purchase recordings (ie. record stores plus DCR jukeboxes) will increase.

It will take some time for DCR to be accepted. First, the technology must be installed by the cable television industry. Then, the programming component of the industry must itself begin to take form and diversify; whether from entirely new entrants or established radio programmers, who perhaps may develop national superstations.

Finally and most importantly, consumers must recognize the improvement in sound quality and new radio formats and adapt to the notion of digital tape recording via cable television lines.

Digital cable radio may be the technology which accelerates the rate at which digital audio signals reach a mass audience.

■ ■ ■

Douglas Conn is Associate Director at the Center for Telecommunications and Information Studies at Columbia Business School. To comment on this article, write him at Radio World International, P.O. Box 1214, Falls Church, VA 22041 USA.

A Pressurized Line Saves Down-Time

by Don Henderson

Pendleton, Oregon USA As engineers, we all have our troubles during the winter weather. However, a recent event during the month of February really brought to mind the age-old expression, "An ounce of prevention is worth a pound of cure."

I received a call from a station which complained that the transmitter was kicking off for no apparent reason. The unit looked fine and all of the meter readings were normal.

Also, none of the tally lights were on, so the technical staff would bring it back up on high power and watch it for a while. The transmitter would run for a day or two and then—"Bang!"—off it would go.

TECH TIP

This continued for several days but they were always able to bring it up on high power and keep it there.

Finally, after doing this for a week or two it started not wanting to stay on high power when they brought it up. They were able, however, to get it to run on low power.

After inspecting the transmitter site I still couldn't get a solid handle on the problem, until I received a call one night following a drastic temperature drop.

Fighting the cold

When we arrived at the transmitter site it was four degrees below zero outside. Inside the building it was 32° and this caused the exciter to fail while we were there.

After getting new heaters in the building and warming things up to a toasty 65°, I thought we might have the problem licked.

Of course things are not always as they seem. Two days later the station called again and said the signal ws so weak it couldn't be heard in their city of license!

This time I knew we had a serious problem, but what? Antenna heaters? Ice on the antenna? All kinds of questions went through my mind.

As soon as it warmed up a little, I decided to climb the tower and check the heaters as well as the antenna. On my way up the tower, we found the problem.

The answer hit me when I was on the tower and the fellow on the ground heard something strange as I moved the transmission line. You guessed it—arcing in the line!

This particular installation had not had any pressure in the transmission line for quite some time. I drilled a small hole into the outer Hyperlon jacket and found water.

I don't mean a small amount—we removed about a pint or more from the outer jacket.

We got all of the water out that we could, but the transmitter was still unable to put out enough power to cover the city.

Listening to the cable as we moved it back and forth, I found the area where the line was arcing. I knew that we would have to cut that section out and ordered a splice kit to repair the line.

When I received the splice kit we spent most of a day getting the rig back on the air. I found a three foot section of the line had arced; this had melted the inside Teflon spacer, allowing the inner conductor to come into contact with the outer conductor.

This arc was caused by the water that had gotten into the line. Where did the water come from? Normal condensation. This line had not been pressurized for some time; air had gotten in and condensed with the heating and cooling of the line.

After getting the splice in I did some calculations and figured that transmission line was absorbing approximately 3400 watts of power!

Needless to say, this whole affair was rather expensive. Besides the lost air time, the station had the expense of the splice kit, next day air service to get it and the time that it took to find the problem and repair it.

All of this could have been avoided by following what most, if not all, of the transmission line manufacturers suggest—*pressurize the line*. Yes, simply keeping a tank of nitrogen or using a dehydrated air pump could have prevented a lot of frustration and expense.

Do it now

If you haven't checked your "pressure on the line" for a while, now is a good time to do it. Also, if you have leaks in the line due to an open bleeder valve or whatever, fix them so that the line will hold a good three pounds at all times.

There will always be a small amount of pressure that will leak out, but if you keep the pressurization system in good working order you might avoid the expense of a repair job and lost air time.

■ ■ ■

Don Henderson is president of Henderson Electronics. To comment on this article, write him at Radio World International, P.O. Box 1214, Falls Church, VA 22041 USA.

Radio World®

En Español

Cuba Observa la TV Martí

La Habana CUBA Representantes del gobierno cubano recibieron con cortesía a los miembros de una delegación formada por radiodifusores y funcionarios del gobierno estadounidense durante la reciente visita de dicha delegación al país isla.

Sin embargo, los funcionarios cubanos manifestaron a sus huéspedes que la interferencia contra las estaciones AM de los Estados Unidos aumentará si la TV Martí comienza la transmisión a Cuba de sus programas de noticias e información.

De acuerdo con la información brindada por los participantes en el viaje de investigación, que se llevó a cabo del 19 al 21 de diciembre, los funcionarios cubanos hablaron francamente de las acciones que pondrán en práctica si la TV Martí se lanza al aire.

Opciones en estudio

Wayne Vriesman, Presidente del Clear Channe1 Broadcasting Service, indicó que los funcionarios cubanos nunca utilizaron la palabra amenaza, prefiriendo referirse a las "opciones" disponibles en caso de que la TV Martí iniciara transmisiones.

El Clear Channel Broadcasting Service representa a 59 estaciones AM de onda exclusiva en los Estados Unidos y el Sr. Vriesman es, además, Vicepresidente para Operaciones de Radio de la WGN en Chicago.

La delegación estadounidense visitó La Habana a invitación del gobierno de Cuba para hablar de asuntos relacionados con la Radio Martí, la TV Martí y la interferencia, dirigida a las estaciones AM de los Estados Unidos, que Cuba utiliza oomo represalia.

La delegación fué presidida por el Representante Al Swift (Demócrata de West Virginia), quien es miembro de la Subcomisión para Asuntos de Telecomunicaciones y Finanzas de la Cámara de Representantes e incluía a representantes de la industria de radio y televisión.

Vriesman manifestó que Cuba está considerando un aumento en la interferencia a las estaciones AM e interferencia directamente dirigida a la TV Martí.

La interferencia a las estaciones AM de los Estados Unidos no es asunto nuevo ya que ésta se inció en 1958 cuando Radio Martí inició operaciones.

Las operaciones de la TV Martí, estación compañera de la Radio Martí, fueron aprobadas recientemente por el Congreso. Una programación similar a la de la radiodifusora será transmitida utilizando un aeróstato o pequeño dirigible a miles de pies de altura sobre el territorio del sur de la Florida. Las primeras pruebas comenzarán este mes.

La interferencia cubana hasta el momento ha estado dirigida a las estaciones de canal libre o de onda exclusiva que operan en cuatro frecuencias: 710, 830, 1040 y 1160 kHz. Las transmisiones consisten en información turística de Cuba.

Un dolor de cabeza de un millón de watts

Vriesman calcula que Cuba posee una capacidad de transmisión de un millón de watts que puede ser dirigida a las estaciones que operan en las frecuencias antes mencionadas. Como resultado, las estaciones estadounidenses frecuentemente sufren el impacto durante las horas de la noche.

De acuerdo con las quejas recibidas por las estaciones, los oyentes tienen que

(continua en la pagina 18)

Procesamiento de Micrófono

John "Q" Shepler

Rockford, Illinois USA Todo el mundo busca el sonido que presenta una VOZ PODEROSA. Uds. saben a que me refiero: La voz del amo que resuena y surge de los altavoces haciendo que los cuadro se caigan de las paredes. Y aún aquéllos que no sufren de delirios de grandeza quieren, por lo menos, que su voz se sobreponga a la música. No hay nada que dé una impresión de mayor debilidad que una voz totalmente opacada por las primeras notas de la próxima selección musical.

Prestarle ayuda a la naturaleza no es cosa nueva. La mayoría de los procesadores de primera línea tienen sistemas complejos que crean la ilusión de un audio más fuerte que la realidad. Esta ilusión es de capital importancia, ya que un audio "desnudo" es aburrido. Aun las estaciones de música clásica introducen unos 20 dB para un sonido 'normal'.

Algunas veces el procesador principal y un micrófono apropiado son suficientes para satisfacer las necesidades de la música y de la voz. Sin embargo, si desea que al superimponer la voz ésta haga más impacto o si desea sencillamente mejorar la situación, el procesamiento suplementario para micrófono le brindará la solución.

La cadena básica para micrófono

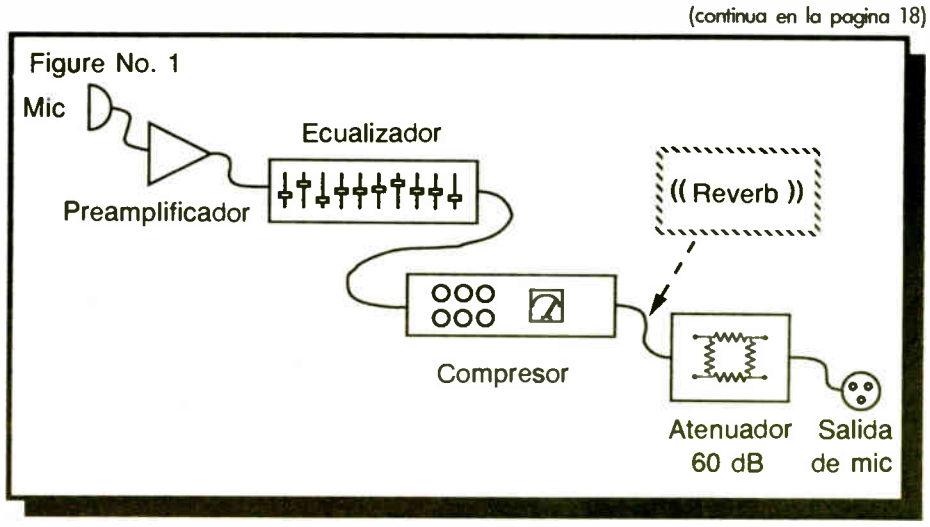
La cadena de procesamiento para micrófono puede ser tan compleja como su procesamiento principal y puede ser igualmente costosa.

No olvide que la música ya ha sido procesada en el estudio para grabaciones. El sonido producido por el disco compacto pasó por equipo complicadísimo antes de ser transformado en bitios digitales. La señal de la voz sale directamente de un micrófono cardiode colocado en un ambiente de acústica imperfecta.

La cadena de amplificación para micrófono debe constar por lo menos de un micrófono de buena calidad, ex-

(continua en la pagina 18)

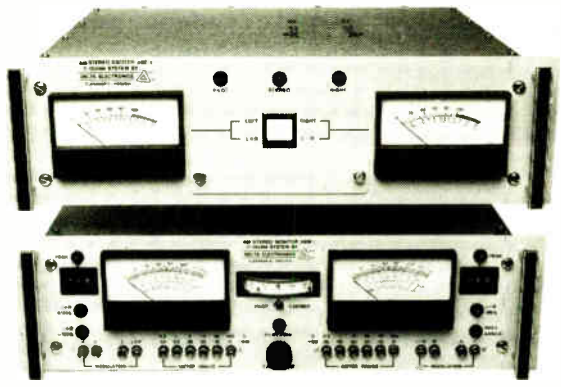
Figure No. 1



Adhere to Broadcast Standards.

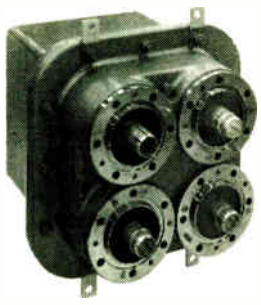
Every day, all over the world, Delta's full line of products make sure the most important broadcasting standard of all is met—your total satisfaction.

AM Splatter Monitor—Spectrum analyzer performance at a significantly reduced price! An inexpensive means of verifying FCC and NRSC spectral compliance. This frequency agile instrument tunes from 1700 kHz down to 450 kHz, with 9 or 10 kHz channel spacing. The monitor also measures incidental phase modulation (IPM). Designed to be rack-mounted or operated from a vehicle's 12 volt supply using an optional antenna.

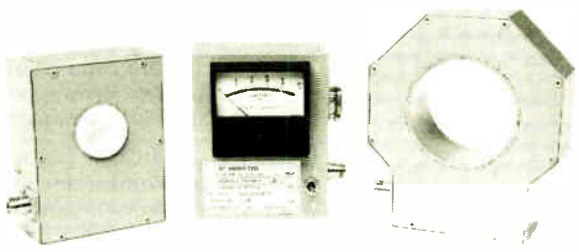


C-QUAM® AM Stereo—The Above Standard Industry Standard is easy to install and maintain with its modular design and construction. Offers standard features other manufacturers charge as options. A sound value, built to last.

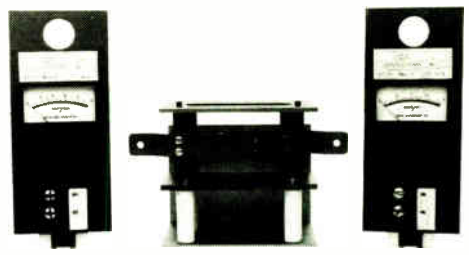
High Power Pulse Reflectometer—Strong interfering fields that would destroy time domain reflectometers are virtually ignored by the PRH-1. This instrument can handle up to 1,000 watts of induced power on an intermittent basis as it locates faults on transmission lines. Provides a visual representation of the transmission or sample line, STL coax, or antenna, using your oscilloscope.



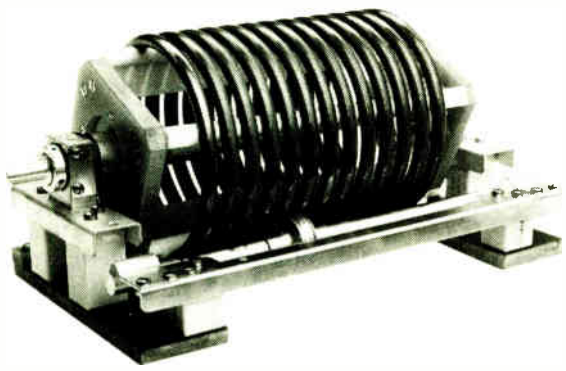
Coaxial Transfer Switches—These 1⅝" and 3⅛" motorized four port switches are designed to switch between antennas, transmitters, or dummy loads both quickly and efficiently. The switches can also be operated manually and are fully interlocked.



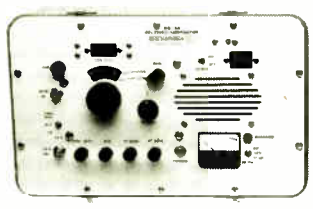
RF Ammeters and Sampling Toroids—Precision toroidal current transformers (TCTs) provide stable antenna monitor sampling while eliminating the problems associated with loops. TCTs also work well in supplying additional modulation monitor or test sample RF outputs. The transformer coupled ammeter (TCA) offers stable base or common point current readings, independent of modulation. The dual and single scale meters also provide remote DC outputs.



Low Power RF Ammeters—When every milliamp of current counts, depend on the accuracy of the TCA-Jr. This portable RF ammeter is designed to plug into either a Delta MJ-50 Meter Jack (pictured above), or a standard J-plug jack. Two current ranges are available: 0.2 to 1.0 Ampere, or 0.4 to 2.0 Amperes.



Rotary Variable Inductor—Where long life and high reliability are required, specify the RVI. Designed to provide long life, even under continuous rotation, the RVI is available in either 12 μH or 10 μH versions (maximum inductance). Other values by special order.



RF Receiver/Generator—A rugged, high output (2 watts) generator and correlation detector receiver virtually eliminate false nulls caused by interfering signals. The RG-3A operates from 0.5 to 1.65 MHz, and the expanded range of the RG-4 generates signals from 100 kHz to 30 MHz.

Transmitter Power Controller—Your insurance against over- and under-power citations. Continuously monitors transmitter power levels, compensating for AC power line sag by adjusting the transmitter to 100% power.

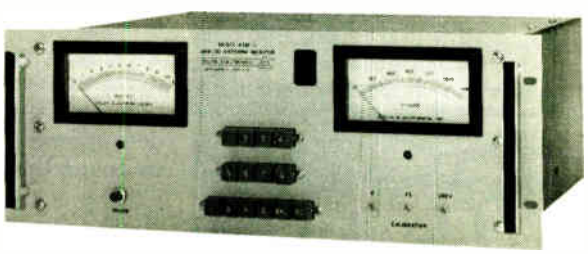
Digital Controlled Processor—This inexpensive, stereo tri-band processor boasts user-friendly controls and an aggressive sound. Mono stations can take a step toward AM Stereo, at a price that won't break the budget.

Impedance Bridges—At last, a means of measuring your impedance under full power. Both portable and in-line bridges are available, with a variety of features, for both AM broadcast and HF applications. The in-line Common Point Bridge can be supplied with a TCA RF Ammeter to permit precise current and impedance measurements.

The Above Standard Industry Standards.

DELTA ELECTRONICS

Delta Electronics, Inc. • 5730 General Washington Dr.
P.O. Box 11268 • Alexandria, VA 22312 • U.S.A.
Telephone: 703-354-3350
FAX: 703-354-0216 • Telex: 90-1963



AM Antenna Monitors—These are true ratio monitors which deliver a ratio reading without the need to continually reset the reference tower to 1.000. This simple operation reduces errors by non-technical personnel and makes tuning an array easier.

Circle 34 On Reader Service Card

Procesamiento de Micrófono

(continua de la pagina 16)
citando un preamplificador de poco ruido y alta capacidad. Casi todos los preamplificadores de tablero son aceptables. Ud. puede hacer una prueba conectando un oscilador a la entrada de micrófono a −55 dBm. No toque el potenciómetro y accione el preamplificador hasta que éste comience a limitar o recortar crestas. Con suerte, obtendrá Ud. sonido de 40 dB o más.

Ud. necesita esta capacidad de sobrecarga para poder controlar la voz de los gritones. No olvide que el micrófono conecta directamente con el preamplificador y que el potenciómetro, que está después del preamplificador, no evitará el recortamiento de crestas.

Escoja un micrófono a su gusto pero es conveniente que éste ofrezca un aumento gradual de sensibilidad en las frecuencias superiores para producir un sonido nítido y que tenga capacidad de sobrecarga. En los micrófonos electrostáticos de años atrás el sonido carece de nitidez y los condensadores "mueren" talvez con demasiada fuerza. A Ud. le corresponde decidir.

Nos permitirmos hacer una sugerencia: Si Ud. no piensa procesar la voz de cada uno de los locutores individualmente, trate de que todos los micrófonos de la estación sean iguales. En esta forma logrará Ud. una acción más uniforme del aire a las noticias y a la producción.

Punto de partida

El procesamiento para micrófono se inicia generalmente con un compressor o un ajustador de ganancia. La compresión al micrófono se lleva a cabo con el mismo objetivo por el cual se hace la compresión de la línea principal, es decir, aumentar la potencia de la señal. En este caso se desea aumentar la potencia de las voces para que puedan competir con la música que ya ha sido procesada.

Las voces no tienen gran potencia inherente. Si se observa la senal producida por la voz en el medidor, se notarán muchas crestas y un promedio bastante bajo. Su objetivo debe ser obtener un nivel promedio más alto sin destruir las crestas. La limitación de crestas en el micrófono ejerce un efecto negativo en la habilidad del locutor para lograr altas crestas, cuando éstas son necesarias para énfasis o efectos especiales. Si las crestas del micrófono son recortadas o limitadas, las crestas de la música se sobrepondrán al micrófono. En esta situación es preferible no procesar el micrófono y aumentar el volumen para que las crestas de la voz puedan surgir y ser escuchadas por encima de la música.

Lo deseado es un efecto de compresión que haga la voz más fuerte. Esto se logra con tiempos de desprendimiento y de ataque moderadamente rápidos. Si el desprendimiento es muy lento, no habrá aumento de potencia. Si el ataque es muy rápido, se producirá la limitación de crestas. Un compresor con constantes de tiempo ajustables para desprendimiento y ataque es excelente para lograr los pequenos cambios y ajustes necesarios. En las situaciones en que no se pueden hacer estos ajustes es necesario cambiar los resistores y capacitores hasta alcanzar el efecto deseado.

Es necesario acostumbrarse a la compresión de recuperación rápida y generalmente se oyen comentarios de que el micrófono se "chupa" al locutor. Esto sucede especialmente cuando la reducción de ganancia continúa recuperando. Los ruidos del estudio aumentan en magnitud y el casco de auriculares se convierte en un dispositivo de retroalimentación.

La solución es controlar el paso de impulsos o de senales en la compresión para que cuando el locutor deje de hablar la ganancia o incremento de potencia cese. También vale la pena tratar de limitar la reducción de ganancia o de incremento. Cuarenta dB de compresión presentan una situación peor que 20 dB.

La cadena para potencia en el micrófono

El dibujo No. 1 presenta una cadena para potencia en el micrófono con un compresor y algunos otros dispositivos.

Escoja un micrófono a su gusto pero es conveniente que éste ofrezca un aumento gradual de sensibilidad en las frecuencias superiores para producir un sonido nítido y que tenga capacidad de sobrecarga.

Un compensador o ecualizador ha sido agregado después del preamplificador y puede ser un ecualizador gráfico o paramétrico. Puede ser accionado por un preamplificador exterior o se puede insertar el equipo procesador antes de la salida del preamplificador.

El ecualizador debe ser ajustado (tuned) cuidadosamente para cada voz. Es como si se hiciera un micrófono a la medida para cada locutor. Si los locutores están trabajando cerca a micrófonos direccionales y esta proximidad produce un sonido turbio, esto puede ser eliminado de 200 a 400 Hz acentuando o reforzando el ecualizador en el campo de 50 a 100 Hz para mantener el extremo inferior ajustado. Probablemente sea necesario agregar alguna presencia en la banda de 3 a 5 kHz.

A continuación les indicamos una manera fácil de ajustar el ecualizador sin que los locutores tengan que leer o anunciar mientras Ud. practica. Utilice el audio de una prueba en el aire del micrófono. Debe ser el audio virgen del preamplificador del microfóno y no del aire. Ahora conecte el ecualizador con la salida de la cinta y compare el sonido con el audio virgen. Ud. ha logrado su objetivo cuando el sonido ecualizado sea el mejor.

El compresor cambiará los requerimientos de ecualización. Conforme las constantes de tiempo son más rápidas, el extremo inferior se vuelve más denso y algunas áreas del extremo superior tienen una fragilidad extrema. En este caso es necesario que Ud. compense.

Un dispositivo opcional es el reverberador electrónico. Hace muchos años este dispositivo consistía en una caja de resortes diseñada para los órganos electónicos. Boing . . . boing . . . boing . . .

Hoy en día los efectos deseados se logran con microprocesadores y disopositivos acoplados y se puede ejercer mayor control. La reverberación o repercución proporciona cierta "vida" al audio virgen, especialmente en un cuarto con un ambiente muy seco. Sin embargo, resista la tentación de añadir demasiada reverberación para que su transmisión no parezca originarse en las profunidades de una caverna.

Toques finales

Casi todo el equipo procesador es equipo en línea para entrada y salida del sonido. La primera caja se alimenta de un preamplificador exterior. Es posible que los preamplificadores de tablero no tengan suficiente ganancia y en este caso debe Ud. utilizar un intensificador o reforzador de fabricación facil. Si Ud. ha eliminado la salida del preamplificador, entonces el último dispositivo es menos atenuado o compensado para impulsar la barra mezcladora.

Conserve la capacidad para amplificación posterior estableciendo los niveles con los valores previos. Conserve, además, las impedancias.

El procesamiento para micrófono puede ser muy divertido y puede Ud. escoger los componentes o utilizar las combinaciones a la venta en el mercado. Sin embargo, recuerde que el procesamiento debe usarse de a poquitos para que el sonido sea mejorado pero sin perder su calidad realista.

■ ■ ■

John Shepler es ingeniero, administrador, consultor para asuntos de radiodifusión y escritor. Si desea hacer comentarios relacionados con este artículo, dirija la correspondencia a Radio World International, P.O. Box 1214, Falls Church, Virginia, 22041, USA

Cuba y la TV Martí

(continua de la pagina 16)
escuchar a un "DJ" en español, desde el atardecer hasta la madrugada, en vez de la programación favorita en las estaciones locales.

El Asesor Jurídico de la Subcomisión para Asuntos de Telecomunicación y Finanzas, Terry Haines, ha indicado que la intensa interferencia radial es represalia debido a que las autoridades cubanas consideran que la Radio Martí es un "ataque a la soberanía cubana."

Los funcionarios cubanos manifestaron a los miembros de la delegación que la TV Martí abriría las puertas a mayor interferencia contra los Estados Unidos, según las palabras de Haines.

La delegación observó directamente la importancia que Cuba da a la TV Martí, hizo saber Vriesman.

Durante una gira por una instalación para transmisiones, la delegación presenció una demostración de interferencia o perturbación intencional al video y al audio de una señal de televisión en el canal 13.

A pesar de las advertencias de interferencia de parte de Cuba contra las estaciones de los Estados Unidos y la TV Martí, los funcionarios cubanos permitieron la visita a Radio La Habana y a las instalaciones de satelite para TV. Sin embargo, indicó Vriesman, negaron autorización para visitar las instalaciones para transmisiones AM.

Los miembros de la delegación manifestaron enfáticamente que no habían viajado a Cuba con fines de negociar un tratado o acuerdo si no a obtener información sobre el tema de radio y teledifusión de interés para ambos países.

El SNG-1, una Ayuda para las Pruebas

Por Bud Aiello
Director de Ingenieria
EZ Communications, Inc.

Fairfax, Virginia USA En la primavera de 1989 Delta Electronics introdujo al mercado el generador para ruido estéreo SNG-l. El SNG-l ha sido diseñado para la industria de radiodifusión estéreo como una fuente de señal de alta calidad para pruebas, que permite una evaluación rápida del audio y del equipo de transmisión, utilizando técnicas comprobadas para la medición de ruido.

Previamente, muchos de nosotros hemos usado analizadores de tiempo real (RTAs) y sus fuentes internas de ruido como la señal de prueba. Ninguna de estas unidades ha proporcionado la señal de salida ideal para satisfacer las demandas de una instalación moderna de radiodifusión.

El SNG-1, con su excelente construcción y funcionamiento sencillo, sigue la tradición establecida por los otros productos Delta.

El SNG-l proporciona una excelente fuente de ruido estéreo (o mono) con una baja impedancia de salida para impulsar casi cualquier carga con capacidad para salida de alto nivel.

Tradición de excelencia

El SNG-l, con su excelente construcción y funcionamiento sencillo, sigue la tradición establecida por los otros productos Delta. El panel frontal tiene los controles para seleccionar "Tipo de Ruido" (rosa, blanco o USASI-Instituto de Normas de los Estados Unidos), "Modalidad de Ruido" (op, compuerta ext o impulso NRSC), "Nivel de Salida," "jacks" equilibrados para salida de izquierda y derecha (¼" TRS) y el "Selector de Salida."

El selector de salida combinado con las salidas equilibradas de izquierda y derecha hacen del SGN-l una valiosa herramienta de trabajo.

El selector de salida puede producir ruido de izquierda solamente, de derecha solamente, mono (ruido idéntico, no existe I=−D), estéreo (ruido de izquierda y de derecha independiente), NRSC para pruebas de AM estéreo (los canales dererecho e izquierdo están parcialmente combinados de manera que el subcanal está a 3 dB más bajo que el canal principa1), I=−D (el derecho e izquierdo se derivan de una fuente única, de manera que no existe D+I) y finalmente una posición recortada (clipped).

Las crestas (picos) positivas del canal izquierdo y las crestas negativas del canal derecho son recortadas o limitadas. Esto es útil para verificar el enfasamiento y la polaridad en todo el sistema estéreo.

Finalmente, tiene un diodo emisor de luz (LED) que indica cuando el aparato está prendido. El panel posterior tiene "jacks" de salida para izquierda y derecha que se desconectan al usar los "jacks" de ¼" del frente.

El SNG-1 de Delta complementa cualquiera combinación de equipo para pruebas.

Un conector BNC se usa para las aplicaciones de una señal exterior de activación cíclica, del fusible de la línea de energía y del conector de energía. Esta unidad no tiene interruptor de energía.

Uso con un RTA (analizador de tiempo real)

El SNG-l, ccmbinado con un RTA y un osciloscopio, permite la evaluación rápida de la respuesta de frecuencia y el mantenimiento del equipo para grabación de cintas. Una vez que se han alineado correctamente las secciones reproductoras de la grabadora, usando cintas de alineamiento regular, las secciones grabadoras pueden ser ajustadas en cuestión de segundos usando el SNG-1, el RTA y el osciloscopio.

Un procedimiento útil es alinear cuidadosamente su cartucho grabador principal o maestro y luego generar una serie de cintas con alineación acimut utilizando ruido rosa. Las cintas se utilizan para hacer un alineamiento de fase de todas las reproductoras del estudio con la grabadora maestra.

El diseño del SNG-l presta atención cuidadosa al seguimiento de nivel de las salidas derecha e izquierda. Esto proporciona una señal extremadamente exacta para el alineamiento del equipo multibanda para procesamiento de audio. El SNG-l y el RTA pueden ser usados para dar respuestas de frecuencia rápidas y acertadas y hacer ajustes al seguimiento de nivel a sistemas multibanda para procesamiento de audio.

Activando el SNG-l cíclicamente con una onda cuadrada con frecuencia de repetición muy baja, Ud. podrá ajustar los circuitos de activación instantánea de su procesador para umbrales idénticos.

El SNG-l y el RTA son importantes para los procesadores de micrófonos. Todo ingeniero que emplee varios procesadores de micrófonos en el estudio ha gastado muchas horas tratando de duplicar condiciones iguales.

Una vez se han logrado las condiciones deseadas en un procesador, sus características pueden ser duplicadas fácilmente con el RTA. La compresión, expansión y umbrales para activación cíclica o intermitente pueden ser duplicadas fácilmente con la función "Activación Externa" y la fijación de nivel de salida del SNG-l.

En el caso de la evaluación de un sistema de transmisión FM estéreo, la separación dínamica estéreo del sistema completo puede ser medida con facilidad. Unicamente tiene que aplicar las señales sólo de la derecha o de la izquiera en la entrada del sistema y le será posible leer la separación en el monitor de modulación. Esta es la prueba de fuego para las actividades de procesamiento de audio y STL y para el sistema generador estereo, transmisor y de antenas.

Nuestras medidas indican que un sistema correctamente alineado puede rendir 50 dB de se paración. Pero, no olvide que esta es una medida dínamica con una señal de prueba de banda ancha. Sus resultados pueden ser menos admirables si los componentes no están debidamente alineados.

En el banco de pruebas es fácil evaluar rápidamente los efectos de dispositivos compuestos para procesamiento en la separación estéreo. Mida la separación del generador estéreo directamente en el monitor de modulación. Inserte, luego, el dispositivo compuesto en serie con el generador y el monitor de modulación. Es muy fácil reducir la separación estéreo 20 dB o más si se utiliza demasiado procesamiento compuesto.

Satisface las pruebas NRSC

El SNG-l satisface los requerimientos del NRSC-l y NRSC-2 para pruebas pertinentes a instalaciones transmisoras AM. La salida pulsatoria de ruido de USASI puede ser usada para medir la respuesta de audio enviada al transmisor y el espectro de RF de la salida del transmisor.

El SNG-l es sumamente útil como parte del equipo para pruebas que utilizan los técnicos encargados del mantenimiento de una instalación para transmisión estéreo. En el mundo moderno es necesario utilizar técnicas rápidas y exactas para medir si queremos mantenernos al día en el campo tecnológico. El SNG-l es un dispositivo de precio razonable que por el tiempo que ahorra y por las nuevas avenidas que abre constituye dinero muy bien gastado.

■ ■ ■

Para obtener más información referente al SNG-l, entre en contacto con su representante local de Delta Electonics o marque 'Reader Service' 66.

MARKETPLACE

Send us your new product info. Be sure to include a black and white photo. Send all submissions to Radio World International Marketplace, P.O. Box 1214, Falls Church, VA 22041 USA.

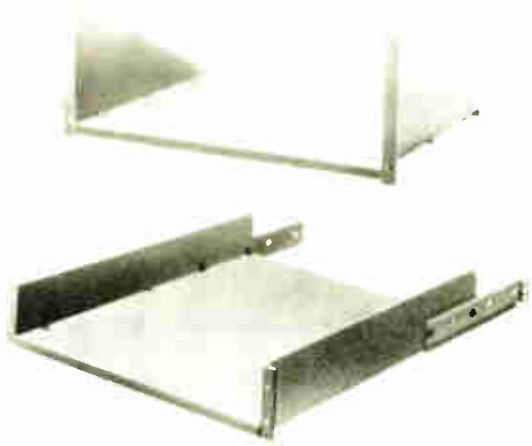
Capstan drive motors

Shown above are three capstan drive motors manufactured by the Beau Motors division of Manger Engineering.

Beau is the original manufacturer of many of the capstan drive motors currently used in tape and cartridge machines.

Beau can repair or replace most capstan drive motors, both foreign and domestic.

For information circle **Reader Service 67.**



Rack mount shelves

Two new types of rack mounted shelves by the Winsted Corporation—Universal Rack Shelves and the Extra Deep/Heavy Duty Shelf—are designed to fit all 19" EIA racks, and are available in four models.

For information circle **Reader Service 78.**

Portable DAT

The Panasonic SV-255 portable DAT recorder features newly-designed mic preamps offering low distortion, high stability and 128 dB (EIN) signal-to-noise ratio.

Another new feature is the dual-channel mono recording mode. In this mode, the right channel input is recorded at full level on the right channel and 15 dB lower on the left channel.

For information circle **Reader Service 84.**

Digital recorder

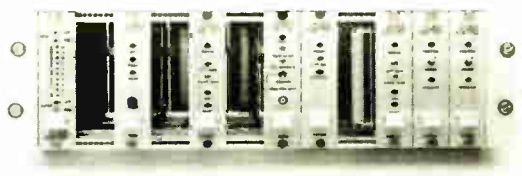
Alpha Audio's new DR-2 digital hard disk recorder can record up to 60 minutes of 16-bit stereo audio with time code at 44.1 kHz or 48 kHz sample rates.

It connects directly to audio and video editing systems via dual RS-422 serial ports.

Options include 30 and 60 minute versions.

Additional options soon to be available include digital I/O and remote for standalone use.

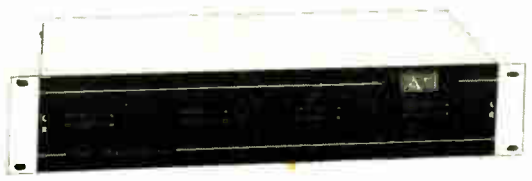
For information circle **Reader Service 58.**



Modular housekeeping system

The Model 8000 Modular System from J.N.S. Electronics, also known as The Frame, consists of 16 modules: stereo/mono DA, mic-to-line amp, audio monitor amp, audio EQ, stereo synthesizer, stereo audio limiter, stereo-mono-stereo switch, audio failure detector, program changeover, stereo audio switcher, AM RF detector, stereo validity generator and detector, program circuit loss alarm, audio test oscillator and video DA.

For information circle **Reader Service 93.**



Studio metering

ATI's new Micro-Meter Studio Monitoring System allows visual monitoring of many audio lines simultaneously with expandable ATI micro-meters that display one, two, three or four stereo signal pairs (up to eight channels) on high resolution, three color, 16 segment LED bar graphs with simultaneous VU and PPM peak display.

Balanced, bridging inputs prevent line loading and are individually switchable for OVU indication at −10, +4 or +8 dBu.

For information circle **Reader Service 76.**

Tape degausser

The TD-5 metal partical tape degausser by Audiolab will erase and neutralize both high coercivity tape cartidges (i.e. Beta SP, M-II, DAT, D1-2 and 8 mm) reels up to 16 inches in diameter and two inches in width.

For information circle **Reader Service 85.**

Rate reducer

The DRR-1500 digital rate reducer is one of Intraplex's Intralink series of T1 multiplex and DACs communications equipment.

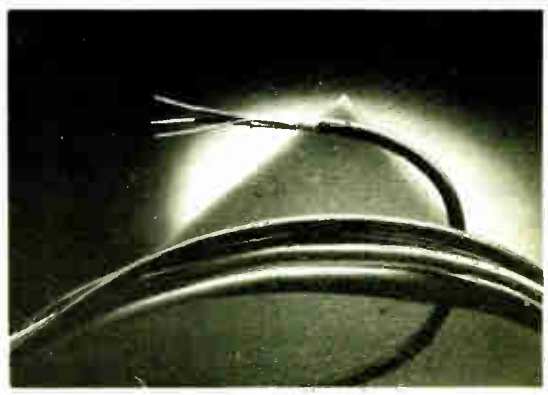
The unit links only those channels within the T1 signal needed by the user for transmission over terrestrial and satellite networks.

The DRR-1500 interconnects T1 equipment such as multiplexers, PBXs and switchers at rate less than the T1 aggregate.

It also provides end-to-end T1 frame transparency.

DRR-1500 is available in both simplex and duplex configurations.

For information circle **Reader Service 80.**



Interconnect cable

Belden Wire and Cable's 1266A is a NEC CM rated miniature twisted pair audio interconnect cable.

Features include: a 22 AWG (7 ×30) twisted pair tinned copper conductor, polypropelene insulation and electrostatic reinforced matallic foil shielding.

For information circle **Reader Service 73.**

Radio World

BUYERS GUIDE

Test & Monitoring Equipment

Dorrough Pleases WQXR

by Herb Squire, CE
WQXR-FM

New York, New York USA The Dorrough Stereo Test Set Model 1200 is perhaps the best thing to happen to a "gain set" since the Daven 1OB Gain-Set filled the racks of radio master control rooms 50 years ago. The Model 1200 not only monitors and tests static signal conditions, it will do the job dynamically as well.

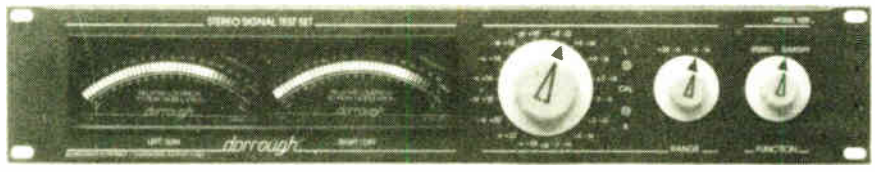
USER REPORT

I saw the Model 1200 for the first time at the 1989 NAB Convention in Las Vegas. I needed a gain set-type device with analog meters to finish up the test position in the new WQXR master control room. The Model 1200 did exactly what I needed, and I got more than I bargained for.

A variety of uses

Besides being a basic gain set, the Model 1200 checks polarity and phase compatibility. But wait—that's not all! It also measures system headroom, noise floor, crosstalk, multitrack azimuth and the effectiveness of dynamic signal processing. It doesn't slice or dice, but it sure will help light up your control room in a festive holiday mood with eighty flashing light-emitting diodes!

The Model 1200 is a relatively small package, taking up 3½" of rack space.



The Dorrough Model 1200 wins kudos from WQXR.

The unit uses two Model 12-B relative loudness to peak modulation meters.

These meters are combined with a pair of input amplifiers, a precision 30-step range attenuator, a high/low range selector and a function control for left-right stereo monitoring or L+R and L–R (sum and difference) monitoring. A buffered headphone monitor output jack also is included.

The key components in the Model 1200 are the loudness meters. The 12-B and its big brother, the 40-B, have been around for a few years as standalone units. Besides program level gain-riding, these meters are great for setting up audio processing chains.

Confirming your ears

The meters in the Dorrough unit simultaneously monitor peak and average audio levels. They confirm your ears in monitoring loudness comparisons. Unlike your ears, however, these meters don't fatigue.

How many times has this happened: You've been working all night setting up processing chains at the transmitter. After a few hours, "bad" starts to sound "good" ("Gee, it sounded great when I left the transmitter at 6 AM!").

The Model 1200 should do the trick. Attached to a good tuner, it also makes an excellent monitor for checking which station is winning the latest skirmish in the "Great Modulation War."

Time and space do not permit a thorough operational walkthrough of the unit, but the Model 1200 has an excellent instruction manual. It's well-written and simple to understand.

The unit works well. A minor short on the headphone jack to the case in my unit was the only flaw. That took a few minutes to clear up.

Real time comparisons

The ability to compare average levels to peak levels in real time is, in my estimation, the most exciting feature this unit has to offer.

Here's a case in point. We have all faced problems with marginal circuit performance. Telco lines are a great example: Let's say it's 25 minutes before air and the lines for an important stereo concert broadcast are still being equalized by the phone company.

The lines are tweaked, the frequency response looks great and the noise floor is where it belongs. However, with audio on the circuit it sounds "funny." There is a distortion problem. There isn't time to bring out the scope and/or distortion analyzer to interpret the readings.

This is where the Model 1200 comes into the picture. Looking at the audio from the remote you can see a problem. The difference between average and peak readings is 2 dB or so. There is an amplifier in the circuit that is misadjusted and clipping the audio.

Depending on time, telco readjusts levels, or the transmit level at the remote site is lowered to give enough overload protection. You can see on the meters where you can get a good compromise. Maybe there still is time for a quick cup of coffee and a chance to catch your breath before the show starts. Wow!

In conclusion, the Model 1200 is a very practical piece of test equipment that may fit the ticket for non-technically-oriented operators who are intimidated by scopes, analyzers and the rest. I would recommend the Model 1200 to those who want to "see" what's happening to their audio.

■ ■ ■

For more information on the Stereo Test Set Model 1200, contact your local Dorrough representative, or circle Reader Service 51.

BUYERS GUIDE INDEX

Neutrik the Choice at the BBC

by Dave Higton
Prod Mgr Test Instruments
Neutrik AG

Editor's note: What follows is an interview Dave Higton of Great Britain's Neutrik AG conducted with former BBC employee John Packman about automated and manual testing from a British perspective. Packman was head of BBC Transmission, which oversees all distribution and transmission of BBC program material.

Part of his responsibilities included providing test and measurement facilities for the entire BBC Transmission organization and buying equipment for its maintenance teams.

These duties spanned approximately 150 radio stations, about 1000 sites transmitting television and 11 satellite-fed World Service sites.

Packman worked at BBC Transmission for many years before leaving the organization last fall. He is now self-employed.

Higton: *Can you tell me something about BBC Transmission's audio testing requirements?*
Packman: Very often the installation test gear was of a more complex, higher priced nature. We used the Neutrik TT402A in both Operation and Maintenance and Capital Projects. Whereas you might have a cheap portable spectrum analyzer for maintenance, you'd have to have something perhaps two or three times as much on the installation side, where portability was not so important.

You normally talk about test equipment being an order better, but it really has to be an order better than the individual parts, which means it has to be two parts better than the end product.

Automation is used there for a different purpose, because you need to send a preordained sequence so that the receiving end knows what signal to expect. CCITT 0.33 is extremely important for testing networks. I don't think anybody uses it too much for testing local circuits, but if you can inject the signal at the studio center and leave 0.33 receivers scattered all over the country, and, say, test Radio 3 at midnight on a Tuesday night, you can get performance tests.

You don't want to send a man to spend a whole day to get the results. You need them back to a central point where they can be processed and any problem areas highlighted. There are telephones there, part of the ordinary public dial networks. You have Radio 1, 2, 3 and 4, and you really want to measure both the received and transmitted signals at the site.

So you need to be able to remotely control an eight-way switch. You can automate that, of course. You can call them up on the phone during the day and arm them to go to Radio 3 and wait for the signals.

Higton: *Do you have many automated desks within the BBC?*
Packman: At the moment there are few desks which can be remotely controlled, though I believe they are likely to come before very much longer. There are some very simple ones. Portsmouth Guildhall, for instance, has an unattended BBC interview studio, where people can actually go in and be interviewed without a reporter being there. "Mixing" is quite a strong word for it; there are only two microphones there! But automation is certainly coming.

Higton: *You've talked quite a lot there about automatic testing methods. What about manual testing methods?*
Packman: Well, manual, to my mind, is what you do once your automatic tests show you have a problem. But there's also a lot of room for manual testing during development and repair. In the context of transmission, it's more repair.

Higton: *What about the economics of automatic and manual testing, and the test equipment to do it?*
Packman: Take the case of testing a radio network. Radio 2 is on 24 hours a day. One of the immediate requirements you get is for something short. So the test has to be automatic. I wouldn't like to do 0.33 as a manual receiver. Even that takes 29 seconds. To do tests during transmission hours, you really have to have something automatic. You have to do it during the night, because there are too many people listening during the day.

Consider what would happen if you did it manually. You'd have to put an engineer at every measuring point. This could well be 3 AM. How many measurement points are you going to do? Let's say two dozen. It's dubious whether you could have one man working up on a dark mountainside.

You might have to have two men in every position. It depends on the site. You are really talking about employing 30 to 50 people at 3 AM. Almost two days' work, because how much work do you get out of them on the normal day before and after? A very inefficient use of staff.

Your costs of doing a manual test are very high. Which tends to mean not so much that you save a lot of money by doing it automatically; it unfortunately means that the tests are not done, which is not a very professional way of going about things. You tend to rely on people to complain that something's wrong.

Higton: *How has the BBC established how often these tests are to be done?*
Packman: No simple answer. All I can say is that the BBC would like to do it a lot more often than they, practically speaking, can. I just know that when they are done, typically a lot of problems are highlighted.

Higton: *What sort of problems do you get?*
Packman: From my observation of tests over the years, the problem most often is noise. It's something which creeps up on you all the time.

There is another aspect which has caught me out quite a lot, and immediately puts down quite an expensive solution. Modern transmission circuits are not linear. They compand, they do all sorts of things. Therefore you often need something like an FFT to get the right answers.

If you're feeding through something like an Optimod, how do you measure frequency response? You can sit there and do all these tests and then you say, "It's not going to work because that's processing it." To which some people say, "Ah, well you switch the Optimod out when you're doing the test." Which is all very well, but how do you test the Optimod?

There is certainly a tendency which worries me; people are bypassing all the processing sections of the circuit in order to do their tests. So an FFT, although it costs a lot of money, might be something that'll be forced on people in the future. The use of compressors and various other forms of processors is a distinct fly in the ointment these days to testing networks.

Versatility counts a great deal in this business. More and more to do live news and things like that means you've got a lot of measurement to do. Therefore automated measurements are extremely important. But you have to think in terms of coming back to manual if it's wrong as well. So the whole thing is flexibility, availability of manual use, and also the capability of fully automatic unattended operation.

Higton: *Let's make this very consumer-*

(continued on next page)

Symetrix Measures Up to WABS

by Bill Ashley, CE
WABS-AM

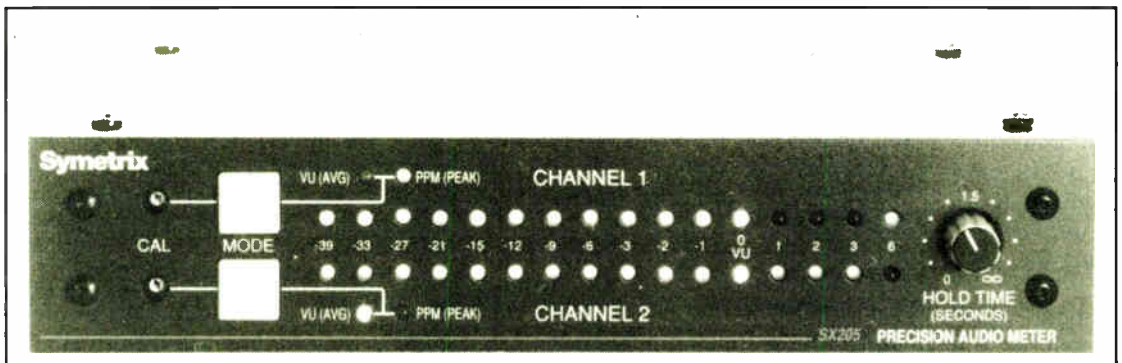
Arlington, VA USA From the earliest days, audio levels have been of paramount importance to broadcast and recording engineers. This is because, in the physical world, there are distinct limitations on dynamic range. If levels are too high, overmodulation or tape saturation occurs; too low, and your signal approaches the noise floor.

For many years, our measuring instrument has been the analog VU meter. This is an average responding AC volt meter, with specific ballistics, calibrated in audio terminology.

One VU (volume unit) is equal to 1 dB. In broadcasting and recording, 0 VU is usually made equal to 0, +4 or +8 dBm. The traditional analog VU meter suffers from limitations including limited range, lack of accuracy and fixed ballistics.

Enter now Symetrix with its SX205 Precision Audio Meter, an LED readout meter that is the newest member of the well known SX family. Like the other members of the family, it occupies very little space—one rack unit in height and one half rack in width.

Installation was simple: four screws hold the SX205 to its rack frame, the power transformer cord plugs into the rear panel and two TRS ¼" phone jacks on the rear panel accept the audio input.



The Symetrix SX205 provides accuracy at reasonable cost.

The Symetrix SX205 is a great device for continuous level monitoring, as an aid in setting up your processing or as a piece of test equipment on your bench.

The audio inputs are capable of being fed from either a balanced or an unbalanced source; the input impedance is high enough to bridge any source.

One important point regarding the power supply. By using a plug transformer, the SX205 qualifies as a low voltage device, therefore not requiring UL approval. The transformer itself, of course, is UL approved.

Ideal for stereo monitoring

The SX205 works wonderfully! It has two channels, making it ideal for stereo level monitoring. With the channels one atop the other, viewing both simultaneously is easy.

Both VU (US standard) and PPM (European peak reading standard) scales are available at the touch of a button. Its range is +6 VU to −39 VU. By switching between VU and PPM during a recording session, you're able to quickly determine your peak-to-average ratio, thereby achieving the best level consistent with headroom, dynamic range and signal-to-noise ratio.

My favorite use of the SX205 at WABS (a mono AM) is to parallel the two channels across the audio output of the modulation monitor. Then, with one channel set for average (VU) and the other set for peak (PPM), I'm able to see quite clearly how hard and how well my processing is working.

The smaller the peak-to-average spread, the harder I'm processing. For this use, I have the SX205 set for the bar mode with about one second of hold time.

While I prefer the bar mode, a dot mode is also available; switching from one to the other is accomplished using the front panel mode switches. The hold time, too, is adjustable. One knob controls both displays and is continuously variable from zero to infinity.

The only other controls on the front panel are the screwdriver-adjust calibration pots. These merely set the input level for referencing to your system's operating level; they do not affect the unit's accuracy.

Right on the money

Speaking of accuracy, I compared my SX205 to a laboratory grade AC voltmeter; it was right on the money! Obviously, when Symetrix put the word "precision" in the SX205's name, it wasn't just hype.

Two other features of the Symetrix SX205 should not be overlooked. One is a built-in 1 kHz sine wave oscillator, putting out +4 dBm from a rear panel TS phone jack. This is quite handy in recording and maintenance situations for system level setting.

The other feature, which will probably be of more interest to service technicians than to broadcast and recording engineers, is its ability to measure power amplifier output.

A rear panel barrier strip accepts the input, while two rear panel switches select the power level (100 W or 1 kW) and the impedance (2, 4 or 8 Ohms). A supplied overlay, calibrated in watts, adheres to the front panel display.

Frankly, I would have preferred the PC board and rear panel space occupied by this feature to have been used for an additional range or two for the line level inputs. An extra 40 dB of gain that could be switched into the input circuitry would have permitted the SX205 to measure noise levels, too.

Silkscreening criticism

One other small criticism I have of the SX205 is the PC board silkscreening. Many of the component designators are under the components they designate. For example, try to find R29 when you're in a hurry. It seems the silkscreen artist has never had to service a piece of electronic equipment.

All in all, though, the Symetrix SX205 is a great device for continuous level monitoring, as an aid in setting up your processing or as a piece of test equipment on your bench. It's compact, easy to use, very accurate, quite reasonably priced and I'm crazy about it!

■ ■ ■

For more information on the SX205, contact your local Symetrix representative, or circle Reader Service 26.

BBC Testing

(continued from previous page)

oriented. Why should anybody bother to test to such high standards? Does it actually make a difference in the end to the customer?

Packman: There does seem to be a certain pursuit of perfection, doesn't there? If you have a long distributed network you need to be able to exchange modules when they go wrong, and to be able to test those modules to a very high standard so that they slip in without having to retest the whole network each time.

So you do need to measure very accurately. You need to be able to forget the test equipment. You normally talk about test equipment being an order better, but it really has to be an order better than the individual parts, which means it has to be two parts better than the end product.

Higton: *Have you received complaints from the "man in the street" that you've had to follow up yourself?*

Packman: On many occasions, yes. A lot of them are actually extremely useful. A lot of people who complain actually do know what they're talking about, and can be very useful.

One of the things nowadays, since the advent of CDs, is that the studio noise and distortion is now considerably below the effects of the transmission network. When FM first came out, it was better than anything else around. Playback and sound reproduction facilities have now considerably overtaken it. I would strongly suspect that the weakest link in the chain is probably now the transmission.

I do feel that automatic operation is extremely important for network testing now. You just can't afford to have people stationed all over the country. And what's more, with things in use more and more hours every day, you can't afford long periods for manual tests. ■ ■ ■

For more information on Neutrik products, contact your local representative, or circle Reader Service 88.

Bonneville Picks ModMinder

by Bill Loveless, VP Eng
Bonneville International

Salt Lake City, Utah USA Bonneville engineers have always felt the FM broadcast industry has incorrectly applied AM's unique physical peak modulation limits and constraints to FM, which has few if any physical constraints. This has worked to the detriment of FM radio's achieving its full quality potential in the real world.

What Bonneville program directors want is loudness without trashing the audio. ModMinder from Modulation Sciences is a tool to deliver that goal. It is not an audio processor, but an accurate method of measuring FM modulation without losing loudness. By legally ignoring very short modulation peaks that do not contribute to radio interference, a station can use less audio processing while retaining the same loudness.

Bonneville found out about ModMinder early, because much of the device's on-air testing was done at

USER REPORT

WNSR, our station in New York City. As a result of those tests we decided to purchase units for all seven of our FM stations.

Surprising controversy

Since its introduction, ModMinder has created a surprising amount of controversy. The controversy has centered around Modulation Sciences' decision to equip the ModMinder with a peak response time of about 900 μs instead of the less than 200 μs common to conventional modulation monitors.

We examined the response time issue carefully during the WNSR testing. MSI supplied us with copies of its communications attorney's opinion of counsel, the Federal Communications Commission (FCC) Report and Order deregulating modulation monitors and the pre-1983 FCC rules dealing with FM stereo modulation monitors.

So, long before the recent FCC declaratory ruling about ModMinder, we were convinced that using a monitor that complied with the pre-1983 rules would ensure compliance with today's overmodulation rules. The FCC ruling confirmed our earlier conviction.

Most modulation monitors have peak indicators with response times of less than 200 μs. Therefore, the indicators false-trip on peaks that are much shorter than FCC rules ever required. To keep those false peaks from indicating as overmodulation, the average modulation must be reduced significantly.

However, by using ModMinder, with its FCC-approved 900 μs response time, brief, false peaks are ignored. The result is more modulation with less processing.

Reducing processing

We found that the more processing in use, the less increase in modulation ModMinder permitted. Measuring moderately processed stations with ModMinder allowed as much as a 15% increase in modulation, with no readjustment of the processing.

But we found something far more important in its long term implications for FM broadcasting. With ModMinder we *reduced*, and even removed, some processing from the air chain *without* any loss in average modulation.

The FM ModMinder, from Modulation Sciences

And because removal of processing made the audio "brighter" and less muddy, the net result was often a subjectively louder signal.

With all the attention focused on ModMinder's response time, many of its other unique features have been overlooked. For one thing, its response need not be 900 μs. By altering one resistor, the response time can be varied from about 200 to 2000 μs. At 200 μs, the ModMinder behaves much like a conventional monitor.

Digital display

ModMinder's digital modulation three-digit display indicates modulation to a resolution of 0.5%. The display updates once each second with the highest peak modulation of the previous second. In conjunction with ModMinder's peak flash indicator, the display eliminates the endless flasher threshold knob twisting normally associated with trying to determine peak modulation.

A true one-minute rolling peak counter displays at all times the number of peaks during the previous 60 seconds. This allows slightly higher modulation and displays continuously exact overmodulation counts per minute, as per the pre-1983 FCC ATS rules.

The remote control capabilities of ModMinder permitted significant improvement in modulation at several Bonneville stations by making it possible to locate the monitor at the transmitter. By avoiding the multipath inherent in off-air monitoring, ModMinder permitted a 15% increase in the accuracy of modulation measurements available at the studio of WNSR.

ModMinder has two types of remote interfaces; one for conventional remote controls, the other a modem-based computer interface. Bonneville stations have not made much use of the standard remote control interface, but many of our stations use the modem/computer interface to great advantage. It provides a serial data stream at 1200 bits/second for total remote operation of ModMinder.

Everything that can be observed or adjusted at the front panel can be done from a remote personal computer via an ordinary dial-up phone line. Modulation Sciences supplies an excellent introductory software package with ModMinder free of charge. It allows for full remote operation, password protection and a histogram data display not available with the ModMinder alone.

Histogram advantage

A histogram is a new and very useful way to display modulation information. It makes setting up processsing much easier. The ModMinder histogram shows how much time peaks spend at each percentage modulation. Using the arrow keys of the PC, a window showing any 25% of modulation may be displayed. The numbers below the modulation are the percentage of time that modulation occurs.

To sum up, the ModMinder is the first real innovation in measuring modulation in a long time. By applying modern digital technology to measuring and analyzing modulation, ModMinder will pave the way to improved FM audio quality. ModMinder is radical in that it allows for improved quality by reducing processing— without a loudness penalty.

■ ■ ■

For more information on the ModMinder, contact your local Modulation Sciences representative, or circle Reader Service 5.

Belar Keeping EZ-101 Neighborly

(continued from page 1)

in the FM band. When coupled to the LO and IF outputs of Belar's RFA-4 frequency selective RF amp, the frequency of any station in the market can be observed.

Digital averaging

A digital averaging function can be enabled to perform successive averages on each two-second reading. The approximation then becomes the average of the average, and so on.

We performed a number of tests on the unit to determine the effectiveness of this averaging function in reducing errors caused by modulation. A stable count with no modulation could vary as much as ±50 Hz with modulation and no averaging. After turning on the averaging function, we observed a maximum 10 Hz change.

One of the unit's features I particularly like is its ability to poll the three frequencies in sequence and display each on a 3.5-digit LED readout while performing out-of-tolerance tests. A two-color LED on the monitor is set to alarm if three center carrier counts exceed ±1 kHz (yellow) or ±2 kHz (red).

Similarly, two other LEDs give visual alarm status for a 2 Hz pilot or 500 Hz subcarrier variation. A failsafe circuit in the unit prevents invalid counts and low input levels from producing erroneous readings. When the optional relay interface board is installed, the FMM-4A can be used in a variety of ATS applications.

Keeping time

The monitor uses a temperature-compensated 6 MHz time base that has a frequency drift specification of better than 5 parts in 10^7 per year. That translates to better than 50 Hz per year at 100 MHz. Since we receive an independent frequency measurement each month, the unit's accuracy is easy to spot check. The 6 MHz time base has a buffered output for even more accurate calibration.

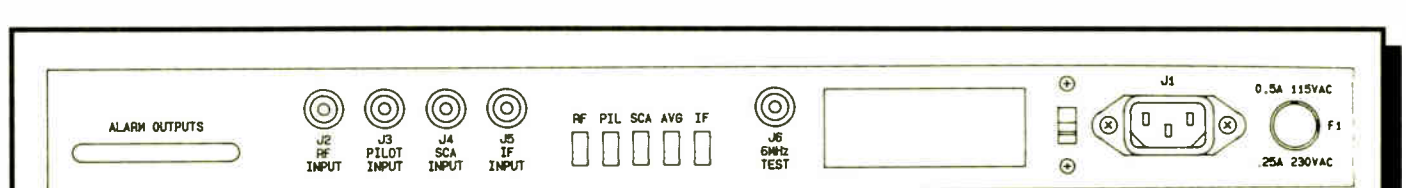
Where you are and what space you take up are two prime criteria in today's highly deregulated FM band. Monthly frequency measurements by an independent service are good as a check, but without continuous monitoring your station could be operating off frequency for up to 30 days.

An automatic monitor such as the Belar FMM-4A will let you know immediately if something causes your exciter, stereo or subcarrier generator to stray out of frequency tolerance.

■ ■ ■

For more information about the EZ-101, contact your local Belar representative, or circle Reader Service 45.

WEAZ uses the Belar FMM-4A to uphold the "good neighbor policy."



The FMM-4A, rear view.

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ANTENNAS & TOWERS

Want to Sell

Hughey & Phillips TI-2035 3500 W isolation transformer, 115/230 V primary, 117/120 V secondary, $900. N Winter, 210 W Cota, Shelton WA 98584. 206-473-3462.

Cablewave FCC15850J air dialectric cable, 320' of 1 5/8", in gd cond, have connectors to sell w/cable, on spool & in easily accessible location for pick up, $800/FOB. D Rozek, 475 South Ave, Beacon NY 12508. 914-831-8000.

Phelps Dodge 3 bay at 105.5 MHz w/radomes, $1300. R Martin, Box 5668, Colorado Springs CO 80931. 719-392-4219.

Phelps Dodge CFM LP-2 2 bay FM 94.5, complete, ready to ship; dummy load 50 kW RCA type 6000 & coax reducer, 3 1/8 to 4 1/16 w/gas pass & blocking rings, BO. L VanDam, WUPQ, Newberry Ave, Newberry MI 49868. 906-293-8522.

Rohn 14" face, 200', on ground, $500; Jampro JLPC 3 bay 98.3, $600; Andrews 1 5/8" air dielectric, 200', $800. D Crisp, Box 1280, Seaford DE 19973. 302-629-6636.

Custom made 900 MHz & 450 MHz STL's, all yagi's new & used. J Woddell, 1412 Mohave, Parker AZ 85341. 602-669-2483.

LeBlanc & Dick tower 1329', 7' face, designed for 2 TV & 5 FM ant's w/comm level @ 1100' that will support 12 2 bays, wind load 65lbs/ft 2 RS 222 specs includes EEG high intensity strobes & red lighting, right light contactor & strobe status controller included, BO. M Fields, 6760 Corprate Dr #340, Colorado Springs CO 80919. 719-548-1528.

Celwave LP FM 3 bay antenna, in service, ready for sale, tuned to 92.1 MHz, BO. K Diebel, 1207 Louisa, Rayville LA 71269. 318-728-4915.

Rigid line 3 1/8", 440', 20' sections, some hangers & 90° sections, $4000. C Haynes, 1850 Lynch, Jackson MS 39203. 601-948-1515.

Scala HDCA-10EB yagi antennas, 75 ohm, 104.1 & 106.9 MHz. D Leinen, Independent Resources, POB 23498, Oklahoma City OK 73213. 405-728-2525.

AUDIO PRODUCTION

Want to Sell

CBS Volumax FM stereo 440, $150. Roberto Blanco, 315-673-9049.

Stereo Generators for Gates TE-1 or TE-3 exciters (2), $350/ea. Roberto Blanco, 315-673-9049.

GE 4BA7 Limitor for AM, $30. Roberto Blanco, 315-673-9049.

Altec 438A compresser amp, gd cond, $10. Roberto Blanco, 315-673-9049.

Allison Kepex noise gates (2) w/pwr supply, $100/BO. J Zelinger, 4401 Sunset Blvd, Los Angeles CA 90027.

Ramko DA280 10 x 8 dist amp in rack mount w/pwr supply, $400/BO. J Zelinger, 4401 Sunset Blvd, Los Angeles CA 90027. 213-667-9310.

Lexicon 1200 mono, audio time compressor/expander, excel cond, $700/BO. J Zelinger, 4401 Sunset Blvd, Los Angeles CA 90027. 213-667-9310.

Harris ME-1 modulation enhancer, rack mount w/manual, $125. L LeBlanc, WKXL, POB 875, Concord NH 03301. 603-225-5521.

Orban 111B stereo reverb, brand new, BO. R Sundell, POB 734, Upland CA 91786. 714-985-0701.

Tascam PE-40 4 band, 4 chnl parametric EQ unused, $250. D Lerner, 115 W 23rd, New York NY 10011. 212-463-0795.

CAMERAS(VIDEO)

Want to Sell

Angineaux 18x15 zoom lens for RCA TK-44 cameras, $2000. Roberto Blanco, 315-673-9049.

GE 240 Color telecine fim chain camera 4 tubes, $3000. Roberto Blanco, 315-673-9049.

RCA TK-44 studio cameras (3), $2900/ea. Roberto Blanco, 315-673-9049.

Ikegami HL-79D 3-tube camera ENG configuration w/Canon J13 x 9BIE lens, w/(3) NiCad batteries, (3) chargers, hard case, AC adapter, (2) cables for power & Sony 4800 recorder, $18000. N Lindquist, POB 14920, Columbus OH 43214. 614-888-4788.

JVC BY-110 3-tube camera w/S-VHS upgrade, Fuji 16X lens, MK50 shotgun mike and KAM-50 mic grip, A/C adapter/charger, (2) DC11U NiCad batteries, VF400 4" B&W viewfinder, studio zoom & focus controls, (2) cables, custom case, service manual, high resolution & top quality performer, like new cond, $3495. B Hines, IPS, RD 1 Box 413A, Export PA 15632. 412-468-4115.

CART MACHINES

Want to Sell

Gates Model 994-06701-002 mono PB only, gd cond, $250. Roberto Blanco, 315-673-9049.

Tapecaster 700 RP mono RP in gd cond, $300. J Morrs, 3911 S First, Abilene TX 79601. 915-676-7711.

Tomcat Pacific Recorders PB (2) w/19" rack & manual, great for spares, $2650. M Barley, 1846 Roseneade #250, Carrollton TX 75007. 214-601-1294.

Harris Criterion Compact 90 mono, R/P, $795. B Mountjoy, POB 1240, Elizabethton TN 37644. 615-543-5349.

ITC PD II R/P gd cond, mono, $950; ITC PD II play, gd cond, mono, $700, or $1300/both. G McCoy, Box 100, Central City NE 68826. 308-946-3816.

Audicord A (3) side by side in 19" rack, mono, gd cond, BO. C Gustafson, 590 W Maple, Kalamazoo MI 49008. 616-345-2101.

BE 2100 stereo RP, brand new, BO. R Sundell, POB 734, Upland CA 91786. 714-985-0701.

ITC PD-II cartridge R/P's (3), gd cond, mono, $500/ea/BO. J Swett, 4025 Lugano Way, Flagstaff AZ 86004. 602-526-1975.

Fidelipac CTR 14 & ESD 10 stereo R/PB & elect splice detector, both units must be sold together, slight head wear. $1800. R Smith, 3407 W Olive #108, Burbank CA 91505. 818-367-6335.

CONSOL

Want to Sell

Collins 212 E-1 mono console, B channel, $80. Roberto Blanco, 315-673-9049.

Fostex 450 8 in 4 out production mixer, excel cond for production sound reinforcement, $500. R Smith, 3407 W Olive #108, Burbank CA 91505. 818-367-6335.

Teac 144 4 trk portable cassette, excel cond, $250/BO. R Fess, POB 250, Macomb IL 61455. 309-833-5561.

Sparta A-20 solid-state mono, 8 chnl, w/manual, fair cond, $500 plus shping. M Hagans, 218 W Hampton, Mesa AZ 85210. 602-964-3100.

LPB Alpha series 8 chnl, 24 input console w/3 inputs set up for mic level, program & mono mixdown output busses, in board monitor amp, in board pwr supply, 3 yrs old, $2500/FOB. D Rozek, 475 South Ave, Beacon NY 12508. 914-831-8000.

Autogram AC-6 excel cond, $2400/BO; AC-8, excel cond w/timer, $3400/BO; Sparta A-15B 5 chnl mono, gd cond, $400. J Swett, 4025 Lugano Way, Flagstaff AZ 86004. 602-526-1975.

Harris M-90, Auditronics 110 Grandson, mono & stereo inputs, 4 or 2 chl outputs, max 18 modules, test osc, stereo 8 pos, input sel, plus much more, BO. J Geogiades, WRRO, 216-373-1440.

Harris Stereo 53 mic inputs, 9 stereo line level inputs gd cond, BO. R Lafore, Box 1327, Valdosta GA 31603.

BE Spotmaster 8BEM100 8-chnl, dual-bussed mono, w/Daven step-attenuators, 16 inputs, $695. B Mountjoy, POB 1240, Elizabethton TN 37644. 615-543-5849.

Ampro AC-8D dual-bussed mono, 8 chnl, 4 inputs per chnl, 32 inputs, manuals, $995. B Mountjoy, POB 1240, Elizabethton TN 37644. 615-543-5849.

Sparta AS-30B stereo/mono, 4 chnl w/pwr supply, gd cond, $350/BO. M Black, Hobart William Smith College, Geneva NY 14456. 315-781-3456.

RECEIVERS & TRANSCEIVERS

Want to Sell

Shure M55 , $50. Roberto Blanco, 315-673-9049.

Fostex M22RP MS stereo studio mic. excel cond w/aluminum carrying case, $400. Don, 3142 Market Place, Bloomington IN 47403. 812-339-4446.

Neumann U-67 vintage tube mic, excel cond, BO. R Kaufman, POB 462247, Garland TX 75046. 214-271-7625.

RCA 77DX vintage ribbon mic, excel cond, BO. R Kaufman, POB 462247, Garland TX 75046. 214-271-7625.

Electro-Voice RE-20 gd cond, $100/BO. J Swett, 4025 Lugano Way, Flagstaff AZ 86004. 602-526-1975.

MISCELLANEOUS

Want to Sell

Bogen MO-100-PA amp, tubes, gd cond, $10. Roberto Blanco, 315-673-9049.

Broadcast cartridges (78), Fidelipac various lengths, $50. Roberto Blanco, 315-673-9049.

CTC-3 25 hz tone detector (2 in 1), $30. Roberto Blanco, 315-673-9049.

Gates M-5144-A carrier alarm operable, $10. Roberto Blanco, 315-673-9049.

Gates CB-77 turntable, 1970's, no tone arm, $50. Roberto Blanco, 315-673-9049.

IBM System 34 main frame computer, excel cond, $3500. Roberto Blanco, 315-673-9049.

IBM compatible 101 key French keyboard AT/XT style, new in box, $100. Roberto Blanco, 315-673-9049.

Nortronics 2 trk 9202 PB heads (3), new, $35ea; Nortronics 2 trk 9200 head in mount, new, $35. R Shroyer, 215 N 4th St, Yakima WA 98907.

Extel teleprinters (2), (4) boxes spooled paper, both excel cond, BO. R Fess, POB 250, Macomb IL 61455. 309-833-5561.

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Page No.	Advertiser	Reader Service No.
28	Allied Broadcast	28
2	ATI	69
11	Audiopak Inc.	61
10	Bext Inc.	64
8	Broadcast Technology Partners	13
3	Broadcast Electronics	17
13	Continental Electronics	47
21	Cutting Edge Technologies	50
17	Delta Electronics	34
7	Econco	55
16	Electrex	6
1	Fidelipac	25
22	Inovonics	20
9	ITC	32
18	Jampro Antennas	22
4	LPB, Inc.	11
4	Marti Electronics	39
15	Orban Associates	42
12	Sescom	75
6	Sono-Mag	9

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Nidec NAH-1202F6 motors (3) for Audi-Cord cart machines removed for shaft burnishing, otherwise in excel cond, $100ea. R Shroyer, 215 N 4th St, Yakima WA 98907.

Vikron PB heads (3) for Audi-Cord series E cart machines, new, stereo, $75ea; mono (2), $35ea; Dummy (3), $5ea. R Shroyer, 215 N 4th St, Yakima WA 98907.

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Want to Sell

Scala STL Antennae 950 mHz, gd cond, $375. Roberto Blanco, 315-673-9049.

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McMartin TBM-3500 mono moudulation monitor FM, $150. Roberto Blanco, 315-673-9049.

McMartin TBM-3000 FM freq monitor, $100. Roberto Blanco, 315-673-9049.

Moseley LPE-10 FM exciter composite input tube type, $500. Roberto Blanco, 315-673-9049.

Belar AMM-1 modulation monitor, gd cond, $790. Roberto Blanco, 315-673-9049.

Moseley Subcarrier detector SCD-2P, $25. Roberto Blanco, 315-673-9049.

Moseley SCG-4T Subcarrier gen 185 kHz, $70. Roberto Blanco, 315-673-9049.

Marti RPT-25R portable xmtr; Marti R-50/450 RPT rcvr; xmtr antenna, base rcvr antenna, cables & manuals, $1295/all. B Mountjoy, POB 1240, Elizabethton TN 37644. 615-543-5849.

Harris 9100 facilities control full ATS, logging, 16 status, 3 calculations, 8 analog, 16 control, 1 studio & 2 transmitter units, BO. J Georgiades, WRRO, 216-373-1440.

Gentner VRC-1000 control unit, fail/safe, command relay, cable accessory, cables, manuals, in use, like new, $3000. D Denton, 405 E Norman, Montgomery City MO 63361. 314-564-2275.

Wegener 1601 mainframe w/1683-08, 1645, 1646 cards and PS, $1500; Microwave Associates VR4XS, $1800 plus shpng. M Hagens, 1705 N Queensbury, Mesa AZ 85201. 602-962-7130.

TEST EQUIPMENT

Want to Sell

Hewlett Packard VHF Signal Generator model 608B 10-420 MC, $500. Roberto Blanco, 315-673-9049.

Simpson 458 oscilloscope old but in perfect shape w/manual, BO. R Fess, POB 250, Macomb IL 61455. 309-833-5561.

TRANSMITTERS

Want to Sell

BC-1E Gates 1 kW AM, $2000; RCA 250K 250 W AM, $1200; GE 250 W AM, $1200; RCA BTF-10B 10 kW FM gd cond, $6000; RCA BTF-3B 3 kW FM gd cond, $5000; Gates BC 5 kW AM xmtr very nice, $8500. Roberto Blanco, 315-673-9049.

FM Combiner 2-20 kW in, 40 kW out, 10 yrs old, $2800. Roberto Blanco, 315-673-9049.

Collins 820 D-1 AM, 1 kW, gd cond, inc all access & tubes, $4000. P Cahill, POB 400, Wanchese NC 27981. 919-473-3434.

FM Combiner 2-20 kW in, 40 kW out, 2 yrs old, tuned & tested, $3000. Roberto Blanco, 315-673-9049.

Lambda C-281M 325-425 VDC Regulated Pwr Supply, $500. Roberto Blanco, 315-673-9049.

RCA BW-73A FM Multiplex monitor, OK cond, $15. Roberto Blanco, 315-673-9049.

Wilinson FME-10 10 W xmtr FM, $1100. Roberto Blanco, 315-673-9049.

10 W FM translators (2) Tepco J-316 & FM amp model T-133, new in boxes, •3900/ea. Roberto Blanco, 315-673-9049.

CCA 10 DS FM xmtr w/meters in rack, 10 W, $750. Roberto Blanco, 315-673-9049.

RCA BTA-5T 5 kW AM, gd cond, BO. J Swett, 4025 Lugano Way, Flagstaff AZ 86004. 602-526-1975.

RCA BTA-521 on 1010 kHz; Collins 21E; Bird Wattwatcher, excel cond, $10,000. S King, 1703 Avondale, Amarillo TX 79111. 806-355-9777.

Collins 831 D-1 FM 3 kW, excel cond, w/all access & tubes, $6000. P Cahill, POB 400, Wanchese NC 27981. 919-473-3434.

QEI 675 FM bdct exciter w/changeable freq & companion stereo gen, $1695/both. B Mountjoy, POB 1240, Elizabethton TN 37644. 615-543-5849.

Will trade 5 kW FM xmtr for 2 to 3 kW FM xmtr. J strongquist, 2816 Haghers, Duluth MN 55811. 218-722-3017.

Collins 830F 10 kW FM on 98.5, BO. J Stanford, 2228 Gravier, New Orleans LA 70119. 504-822-1945.

Harris FM 10H clean, working, w/wo BX 15 exciter, BO/may trade. A Branch, Box 1979, Decatur GA 30031. 404-325-7847.

Sparta 680 solid state exciter on 96.7 MHz, used one time, excel cond, $1200. K Diebel, 1207 Louisa, Rayville LA 71269. 318-728-4915.

TV FILM EQUIP

Want to Sell

Delta Microtime 44 Quad Time Base Corrector, $700. Roberto Blanco, 315-673-9049.

Vidifont MKIV-A character generator w/disk, $9000. Roberto Blanco, 315-673-9049.

Central Dynamics VSP414 special effects production switcher, $4000. Roberto Blanco, 315-673-9049.

RCA PMS-5 monitor switchers rackmount tandem (2), $100. Roberto Blanco, 315-673-9049.

RCA PMS-18 monitor switcher rackmount, $150. Roberto Blanco, 315-673-9049.

Audio patch panels (5) pre-wired 24 across, 2 rows, $100/for all. Roberto Blanco, 315-673-9049.

Video patch panels (6) 15 across, two rows, $100/for all. Roberto Blanco, 315-673-9049.

RCA TG-3 sync generators (2), $200/ea. Roberto Blanco, 315-673-9049.

CBS 8400 Image enhancer, $300. Roberto Blanco, 315-673-9049.

Eastman 285 16mm television film projector, $900. Roberto Blanco, 315-673-9049.

RCA PM-14 (5) B/W 14" monitors, $60/ea. Roberto Blanco, 315-673-9049.

GE 240 Telecine camera, $3000. Roberto Blanco, 315-673-9049.

Eastman model 2 Pneumatic film chain multiplexer, $1000. Roberto Blanco, 315-673-9049.

Complete film package, 16mm projector, dual slide drums, multiplexer and GE Telecine camera, $4900. Roberto Blanco, 315-673-9049.

Spectrum 32 Dual Slide projector, complete, $900. Roberto Blanco, 315-673-9049.

RCA PM-9 B/W 9" monitors (5), $40/ea. Roberto Blanco, 315-673-9049.

Spindler-Sauppe S-32B used on multiplexer, w/lens, mounting plate & manual, $500/BO. P Wagenschein, Baylor University Division Telecom, 1218 S Third BU Box 7368, Waco TX 76798. 817-755-1511.

TUBES

Want to Sell

5762 (5) used good, $190/ea; (4) newly rebuilt, $575; (10) 7007 good $800/ea. Roberto Blanco, 315-673-9049.

Tubes, 1000's have receiving & amp number your looking for, some transmitter tubes too, $2most. A Branch, Box 1979, Decatur GA 30031. 404-325-7847.

ACTION-GRAM

Equipment Listings

Radio World's International Broadcast Equipment Exchange provides a FREE listing service for all broadcast and pro-sound end users. Simply send your listings to us, following the example below. Please indicate in which category you would like your listing to appear. Mail your listings to the address below. Thank you.

Please print and include all information:

Name _____
Title _____
Company/Station _____
Address _____
City/State _____
Postal Code _____ Country _____

Brokers, dealers, manufacturers and other organizations who are not legitimate end users can participate in the Broadcast Equipment Exchange on a ***paid*** basis. Listings are available on a US$25 per 25 word basis.

I would like to receive via surface mail **Radio World** FREE each month. ☐ YES ☐ NO

Signature _____ Date _____

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I. Type of Firm

A. AM station
B. FM station
C. Television station
D. Network office
E. Government office
F. Recording studio
G. Independent engineer
H. Manufacturer/distributor
I. Other (specify) _____

II. Job Function

A. Ownership
B. General management
C. Engineering
D. Programming/production
E. News operations
F. Other (specify) _____

Want to sell: ☐ Want to buy: ☐ Category: _____

Make: _____ Model #: _____

Brief Description: _____

Price: _____

Want to sell: ☐ Want to buy: ☐ Category: _____

Make: _____ Model #: _____

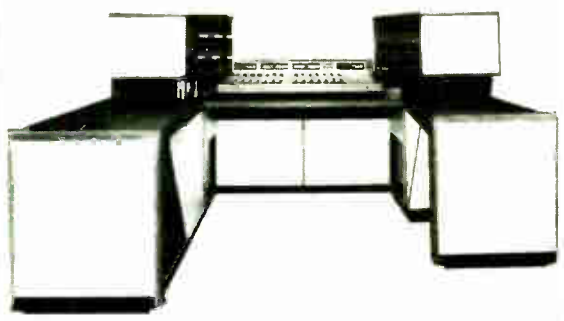
Brief Description: _____

Price: _____

Int'l Broadcast Equipment Exchange
PO Box 1214 • Falls Church VA • 22041 USA
Phone: 703-998-7600 FAX: 703-998-2966

MARKETPLACE

Send us your new product info. Be sure to include a black and white photo. Send all submissions to Radio World International Marketplace, P.O. Box 1214, Falls Church, VA 22041 USA.



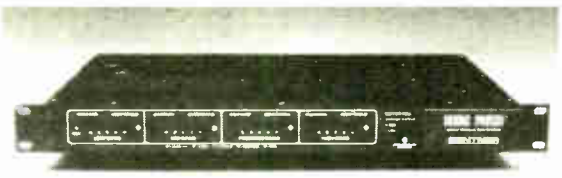
Studio furniture

Features of Murphy's Elite Series of studio furniture include non-chip rounded bumper base laminated to a solid hardwood molding, three built-in parallel wire passes, access panels along the entire back and ventilating reveal at top.

The underconsole racks are field relocatable.

The furniture is available in both stand-up and sit-down configuations with a two-to-eight week delivery time.

For information circle **Reader Service 82.**



FM Audio Processor

The Audio Prism is a 4-band FM audio processor designed to provide both clarity and power for your signal. It allows you to gain maximum modulation to keep your listeners listening longer. Digital control ensures maximum signal penetration, without the processing artifacts that can drive listeners away.

For AM, Gentner provides the Phoenix, a digitally controlled audio processor. The Phoenix combines superb audio performance and NRSC-1 compliance. The new NRSC-1 regulations mean tighter constraints on modulation. With the Phoenix, you can operate within these constraints while maximizing modulation.

For information, circle **Reader Service 90.**

Portable Radio Links

TFT, Inc., of Santa Clara, California, USA now has available its 8888 series of portable program quality radio links. These operate in the 370 MHz to 470 MHz range with up to 25 watts power output.

The 8888 series is frequency agile, with variable FM deviation and selectable receiver bandwidth. Included is DTMF signaling for system security and cueing commands. There are three mike/line inputs, and audio processing loop-through facilities.

For information, circle **Reader Service 59.**

Expansion connector

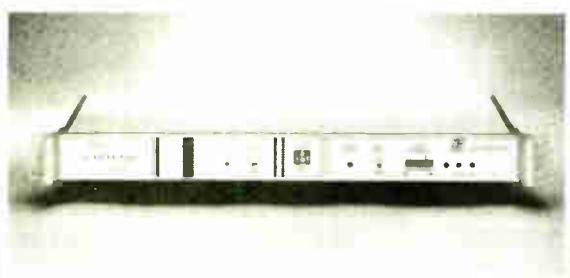
Cablewave Systems has a new expansion inner connector assembly.

This assembly is an option to the bullet anchor inner connector usually supplied with rigid line sections.

The sliding contact surface is made with a silver plated beryllium spring.

The connector is available for Cablewave rigid line sizes 3 1/8", 4 1/16" and 6 1/8".

For information circle **Reader Service 74.**



Composite filter

The Dividend FM composite filter, by Cutting Edge Technologies, operates on the portion of the composite spectrum above 53 kHz. This reduces main channel crosstalk to −60 dB or more.

The Dividend allows full stereo separation with no overshoots.

The filter also is designed to be phase linear.

According to Cutting Edge, users can typically increase modulation by five percent to reach the legal limit.

For information circle **Reader Service 92.**

Transmission test set

Neutrik's TT402A audio transmission test set can be used as either a standalone or a computer integrated audio measuring system.

Line monitoring, field trouble-shooting, quality assurance and data aquisition may be accomplished via the "one button- one function" front panel. The TT402A is able to both operate independantly in the field and in permanent installation.

For information circle **Reader Service 56.**

Stereo synthesizer

The Last Word, by Titus Technologcal Labs, is a microprocessor-based, automatic stereo synthesizer and stereo audio correction device.

The unit corrects audio problems such as loss of a channel, loss of signal or inverted polarity.

The TLW-2 can automatically insert an internal dynamic stereo synthesizer into an air chain.

The unit also provides full metering and audio monitoring.

For information circle **Reader Service 71.**

Audio volume meter

The AVM audio volume meter, by FM Systems Inc., is a hand-held, digital, battery-operated RMS meter calibrated in dBm in a 600 ohm circuit.

Steady digital meter readings are obtained by storing and displaying only the highest RMS measurement made during the measuring interval.

The AVM meter reads audio levels in a 70 dB range from +20 dBm to −50 dBm in 0.1 dBm steps with basic accuracy of ±1 percent ±1 digit.

For information circle **Reader Service 86.**

Condenser mic

The MKH 50-P48, by Sennheiser, is a symmetrical transducer, transformerless mic for digital recording which comes complete with an MZQ 40 stand adapter.

It includes a switchable 10 dB pad and proximity effect compensation filter.

The MKH 50-P48 uses a symmetrical push-pull capsule design incorporating optimum resistive loading for an ultra linear response.

For information circle **Reader Service 97.**

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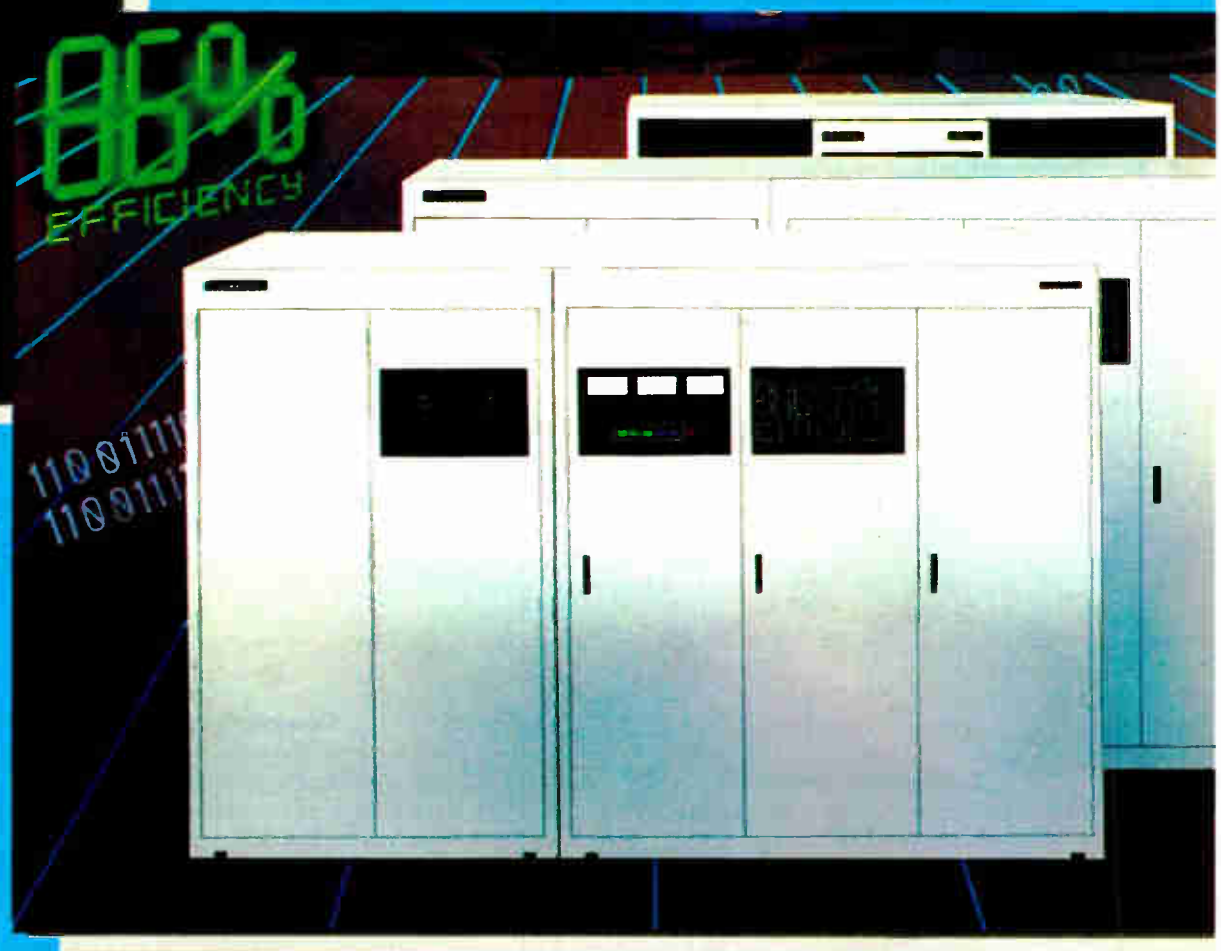

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