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THE SOUND ENGINEERING MAGAZINE

db

SEPTEMBER 1978
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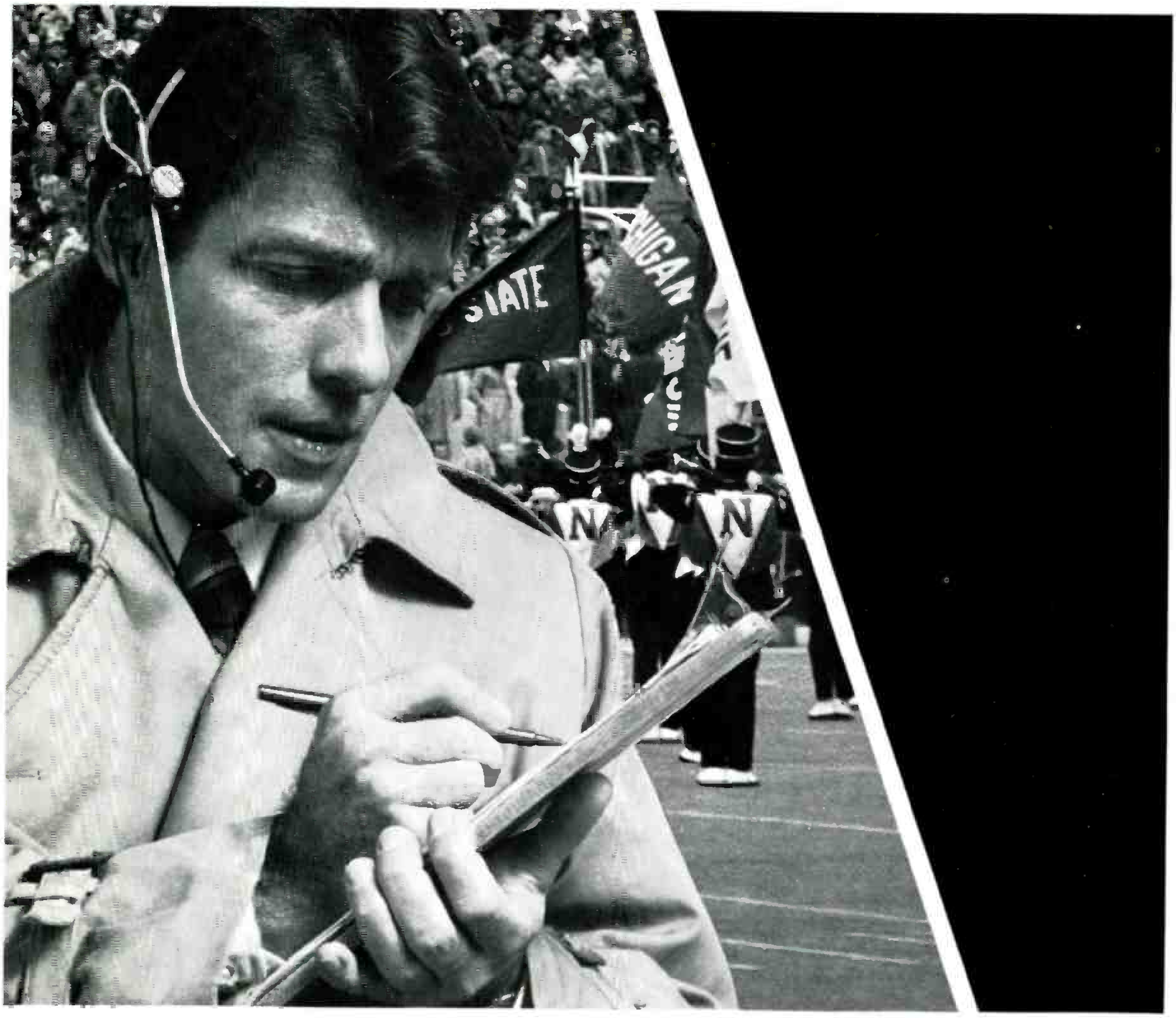
WHAT ARE THE WILD WIRES SAYING?

SPECIAL BROADCAST AUDIO ISSUE



Coming Next Month

● The subject for the October issue is disc recording.

● Irv Diehl helps us get started with a look at Basic Groove Geometry. How long is a 12-inch lp record? Well, it depends. Depends on what? Check with this article for the answer.

● When was the last time you saw a new disc cutting lathe? We saw a new one at the recent AES Convention and sent our new Associate Editor Suzette Fiveash (see the masthead) to find out about the Cybersonics system.

● Wouldn't it be nice to figure out a way to get a few more dB onto your next record? The folks at the CBS Technology Center thought so too, and have spent about five years developing a way to do just that. Charlie Repka reports on what they've got in the CBS DISComputer Mastering System.

● Direct to disc has been receiving a lot of notice these days. Whenever the conversation turns to this subject the name of Bert Whyte will surely get mentioned. We asked Bert (while he was between sessions) to tell us some things about this subject. He does in The Logistics of Direct-to-Disc Recording.

● Associate Editor Suzette Fiveash was busy last month with a lot of new assignments, but we were able to get her to interview Stan Ricker of the JVC Cutting Center in L.A. to find out what happens when you Bring Your Own Lathe to a direct-to-disc session.

● James Shelton, president of Europadisk Plating Company once took us to task for not saying enough on the subject of disc plating. So we convinced him to contribute A Look at the Record Plating Process.

About The Cover

● What are the Wild Wires Saying? This drawing by noted artist Charles Dana Gibson (did you ever hear of the Gibson Girls) appeared in Life Magazine (not the Time-Life publication of later). The year was 1922.

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db Letters

THE EDITOR:

You asked for comments from broadcast engineers. Here goes:

First. I go back to the days when James Langham was touting triode amplifiers as the highest thing in hi fi. I have been a broadcaster since 1948 and a chief engineer since 1955. I have both a.m. and stereo f.m. experience in several formats. So much for qualifications.

In those early days, we took what we had in the way of audio and transmitting equipment and did the best job we could of *faithfully* reproducing efforts of the recording industry represented on 78 r.p.m. shellac discs. I emphasize the word faithfully because we had no thought of attempting to alter what the recording studio had done.

Today our amplifiers are close to the fabled "wire with gain" we hoped for then. New techniques give us transmitters which are capable of low distortion, low noise and wide range—both a.m. and f.m. In fact, modern a.m. transmitters give us better quality than the f.m. units of a few years ago. We are equipped to broadcast either live or recorded material with a faithfulness which cannot be distinguished from home reproduction of the same material—if we want to.

This is the rub. At a time when we could offer a "wire with gain" between the record companies' efforts and the airwaves, we compress, clip, "process," boost, shape, and otherwise distort the product until there is little left of the many hours of effort put into a record by the company who made it, a company whose engineers are far more competent to tailor musical sound than any broadcast engineer will ever be: that is their specialty! Did you know that some so-called "more music" stations edit a record to cut its running time so they can pack in more records per hour? Did you know that some others run their turntables slightly fast (easy with frequency-sensitive motors) to accomplish the same purpose?

Somewhere someone came up with the theory that you had to be louder than the competition or you lost listeners. Why? The only person who would notice this effect are those casual listeners who tune across the dials of their car radios as they whiz through your town ten miles over the speed limit. They don't spend a dime with the local merchants.

The truth is that much of the blame can be laid at the doors of the equip-

(continued)

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ment manufacturers. After all, their business is selling new products and if every one of them waited around for you to replace your transmitter, etc., when they wore out, half would be out of business in a month. I spent eight years in industrial design engineering. When we came up with an idea for a product that solved an existing problem, well and good. But sooner or later you ran out of problems. At that point—if you wanted to keep your job—you dreamed up a few. Then your sales department set out to convince the potential customer that this "problem" had been sapping his manhood for years without his knowing it—but now WE had the solution! Do you think I'm kidding? Talk to some r & d people.

I agree with you that most loyal listeners prefer a given station because of its format, not because its sound is jazzed up or it is the loudest thing around. But, try to convince the owners or managers. It's funny, those guys are in the snake oil selling business but they are also the most gullible people around.

Let us *not* discuss rating services. I have seen them come up with results which would do credit to the aforementioned snake oil salesmen. The station for which I worked consistently had two out of three correct responses to telephone quiz questions which required a person to be listening—66 per cent of the local audience. Right? Wrong! The station wound up on the bottom of a national survey of stations in the area.

The man who said that millions have been spent in audience research just proves Barnum was right. Any broadcaster worth his salt who cannot find out what the listening audience in his own bailiwick wants in the way of radio by generating his own local survey had better go back to selling shoes. Please remember that a survey of listener preferences in New York City (or even in Dallas) bears no more relationship to the audience preference in, say, Port Arthur, Texas, than would a breakfast cereal survey of Earthlings relate to the eating habits of Martians.

As far as the "basic problems facing the broadcast engineer" in today's radio world, I have found that, in addition to the usual ones of not having enough spare parts, trying to get obsolete equipment replaced before it dies in the middle of a broadcast, and trying to get operators to handle your equipment in some sort of reasonable manner to avoid instant destruction, the only *real* problem is trying to keep your owner or manager from turning the station into a testing ground for every piece of junk offered to solve a problem you never knew you had.

(continued)

You know what Technics quartz-locked direct drive does for records. Now listen to what it does for cassettes.

Accuracy good enough for even the most demanding professional, that's what Technics quartz-locked direct-drive turntables are all about. And that's why radio stations use them and discos abuse them.

Now you can record your records as accurately as a Technics turntable plays them. With the RS-M85, our new quartz-locked direct-drive cassette deck. Not only does it have the kind of transport accuracy that's hard to beat, it has that kind of price, too. The reason for all this accuracy: The performance of Technics direct drive combined with the precision of our quartz oscillator.

The RS-M85's servo-controlled system compares the motor rotation with the unwavering frequency of the quartz oscillator and instantly applies corrective torque if any speed deviations are detected.

To complement that accuracy, Technics RS-M85 has a Sendust head with a high-end frequency response of 18,000 Hz, low distortion and excellent dynamic range.

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Also available is Technics RP-070. An optional full-function infrared wireless remote control.

Technics RS-M85. Compare specifications. Compare prices.

FREQ. RESP. (CrO_2): 20-18,000 Hz. WOW AND FLUTTER: 0.035% WRMS. S/N RATIO (DOLBY): 69 dB. SPEED DEVIATION: No more than 0.3%.

Technics RS-M85. A rare combination of audio technology. A new standard of audio excellence.

*Dolby is a trademark of Dolby Laboratories, Inc.

Technics
Professional Series

Circle 15 on Reader Service Card

Want to have some fun? Read the "used equipment" ads and see how many of the "problem solvers" turn up in the let's-get-rid-of-it-before-it-rots section.

One of the big excuses for all the a.m. audio processing has been the "need" to tailor the signal to the poor quality of today's auto and portable radios. If this had been done by the early f.m. broadcasters, just how much fidelity would today's f.m. receivers possess? We *forced* the receiver makers to produce a better product by making it known that the early units were not getting ALL of the signal we were transmitting. I might add that the fairly good quality f.m. sets we have today are cheaper in real dollars than the early sets. The same thing is happening to t.v. The way the t.v. station owners act, you would think they were planning on holding onto the station for the maximum three years required by the FCC and then selling off for a bundle. You may not know it, but the way the sale price is usually set is by taking twice the annual gross and adding it to the inventory of land, equipment, etc. This means that if you hype the gross way up, you can take out a bundle while you've got the joint and then sell off for another bundle and let the buyer try to unscramble it all. The FCC could solve the problem by rewriting the rules on Audio Proofs to require all equipment in place and set for normal operation (except compression on Frequency Response measurements only). They could also write a general rule restricting the use of response tailoring circuits (with a further restriction on using them to record stuff so the intent of the rule could not be circumvented.)

The scream from the 250 watter would come through loud and clear: "We can't sound as loud as the 10 kilowatt stations if we can't process."

Right as rain. Where is it written that a man with a hundred grand invested has the right to all the benefits given one with a cool million in the plant? Try PROGRAMMING for a change—you may find it works wonders!

I am sorry that I must ask you not to use my name. I would enjoy hearing from some of my fellow sufferers in the business (as well as chuckling over the apoplectic replies from those who feel I am a traitor to the cause) but I do work in the business and I do enjoy getting my paycheck every two weeks—mostly because my family enjoys eating—strange habit.

Thank you for wading through my mutterings.

A CHIEF ENGINEER

THE NEW TAD* DRIVER, IT GIVES YOU MORE BY GIVING YOU LESS.

The new one-inch TAD driver is truly unique. There is nothing else like it. Use it with your favorite horn and you'll get a frequency response from 800 to 22,000 Hz. So one speaker does the same job it used to take both a tweeter and super-tweeter to do. Saves weight. And money.

The secret is the TAD driver's beryllium diaphragm.

The diaphragm is the heart of a driver. It must be both light yet rigid. We used beryllium to achieve a standard of performance never before known to the professional. Every single tone — no matter how subtle or how complex — is captured and reproduced just the way it is played or recorded. The resulting sound is a revelation.

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If your job involves professional sound reproduction — on stage, in concert halls, in clubs or studios — you have an obligation to yourself to hear this remarkable driver.

With the arrival of the TAD driver the state of the art just took a quantum leap.

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A Division of U.S. Pioneer Electronics Corp.
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Telephone: (201) 440-8234

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Area Code

*TAD is an abbreviation for Technical Audio Devices, the professional products division of U.S. Pioneer Electronics Corp.

Circle 14 on Reader Service Card

db Calendar

SEPTEMBER

29 **Society of Broadcast Engineers Regional Convention,** Chapters 1, 2, and 22. Syracuse-Hilton Inn. Contact: Greg Dunn, WSYR-TV, 1030 James St., Syracuse, N.Y. 13203. (315) 474-3911.

OCTOBER

5-8 **High Fidelity Music Show,** New York City. Statler Hilton. Contact: Teresa Rogers, P.O. Box 67, New Hope, Va. 24469. (703) 363-5836.

5-8 **New York Management Seminars.** Contact: Heidi E. Kaplan, 14NR, New York Management Center, 360 Lexington Ave., New York, N.Y. 10017. (212) 953-7262.

5-6 **The Effective Engineering Manager.** New York City.

5-6 **Unlocking Creativity.** New Orleans.

23-24 **New Products: A Systematic Approach.** Chicago.

15-20 **Audio-Visual Institute for Effective Communications.** Indiana University.. Contact: Dr. E. L. Richardson, Audio-Visual Center, Indiana University, Bloomington, Indiana 47401. (812) 337-3853.

16-19 **Instrumentation-Automation Conference & Exhibit.** Philadelphia Civic Center. Contact: Instrument Society of America, 400 Stanwix St., Pittsburgh, Pa. 15222. (412) 281-3171.

16-18 **JBL Workshop,** Sound Reinforcement. Chicago. Contact: Nina Stern, James B Lansing Sound, Inc., 8500 Balboa Blvd., Northridge, Ca. 91329. (213) 893-8411.

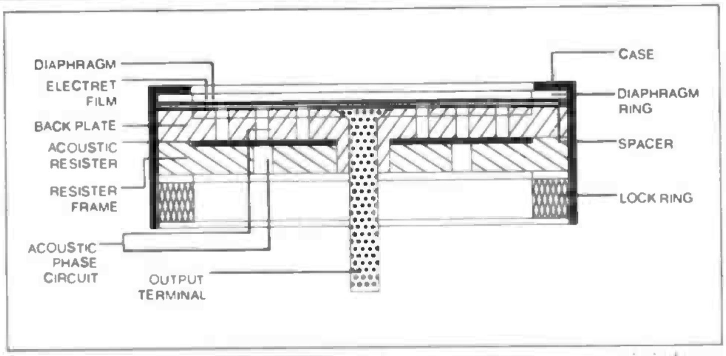
17-19 **INTERNEPCON/UK** Metropole Exhibition Centre, Brighton, England. Contact: British Information Services, 845 Third Ave., New York, N.Y. 10022. (212) 752-8400.

18 **National Radio Broadcasters Association, Sales Manager Seminar.** The Welsh Company, Tulsa, Oklahoma. Contact: NRBA, Suite 500, 1705 De Sales St., N.W. Washington, D.C. 20036. (202) 466-2030.

17-19 **Syn-Aud-Con Seminar,** Atlanta, Ga., Presidential Park Quality Inn. Contact: Synergetic Audio Concepts. P.O. Box 1134, Tustin, Ca. 92680. (714) 838-2288.

You can hear the difference because the difference is right here.

Take a look at what's taking the industry by storm. The Back Electret, another giant step forward from Sony.



Never before has it been possible for thin polyester film to be used in electret condenser microphones. That's because polyester film, acknowledged as the best material for microphone diaphragms, just can't hold a static charge for a long duration.

But Sony's engineers have made the impossible, possible. They've found a way to adhere the electret material directly to the back plate of the microphone. By thus putting the charge on the back plate, we are able to use polyester film in the diaphragm.

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But you don't have to look at Back Electrets to see why Sony is ahead.

NO MATTER WHAT KIND OF MIKE YOU NEED TO GET, WE'VE GOT IT.

Sony's microphone line is thoroughly complete. It ranges from professional condenser to semi-professional to microphones for public address, vocalists, and outdoor use. There's omni and uni-directional. And we think it's big of us to make sophisticated miniatures.

And all microphones are available with Phantom Power, battery operated, or both.

So if you need something to talk into, it makes a lot of sense to talk to Sony. Write to Sony, 714 Fifth Avenue, Dept. TK, New York, N.Y. 10019.

SONY AUDIO

Circle 23 on Reader Service Card

PATRICK S. FINNEGAN

db Broadcast Sound

Audio in Automation

● There are a large number of broadcast stations whose audio programming is handled entirely by some type of program automation system. Although the component units of such a system are very closely interlocked and run automatically, this does not lessen the need for quality in the audio they produce. Interlocking component units of a system does make it somewhat more difficult to service than a live studio arrangement. The design philosophy of a system can make internal measurements of individual units somewhat less than easy. This month we will touch upon some of these aspects in the measurement and maintenance of a major program automation system.

AUDIO AND CONTROL

There are at least two fundamental aspects of any program automation system, audio and control. The audio programming which the system generates and distributes is the primary objective—the same as in a live studio arrangement. Although the objective is the same and the tape players similar, the audio paths can be far more devious, looping through various switching and control devices for interlock purposes. Aside from the routing, there are also no convenient jack panels or patch bays as are found in the live studio. Finding suitable measurement points along the audio path can present a real problem in itself.

Because the system must run itself, there are many tightly interlocked control circuits. This interlocking is done to prevent a variety of sources playing on the air at the same time, as well as other control functions. So consequently it is not always a simple matter to remove a tape machine to the shop for servicing. In many cases, a number of by-pass switches must be thrown or jumper wires added to complete the control and audio circuitry around the machine before it can be removed.

CONTROL AUDIO

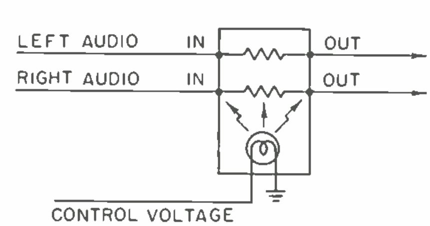
Although the switching and control pulses within the system are generally d.c. voltages, switched grounds, relay contact closures, transistor switches and the like, many of these initially start out as an audio signal within one of the tape players. In cartridge tape equipment, this is usually the 150 Hz auxiliary tone on the tape's cue track, while on reel-to-reel music machines, it is often a 25 Hz tone on the audio track itself. The audio control tones are converted to non-audio control pulses at the output of the tape player. Within the machines themselves, it is important to treat these tones as audio signals in their own right. Although our main concern is the program audio, if we neglect these control tone channels within the machines, system switching can become erratic or not function at all. If the music decks in a system are using a burst of 25 Hz tone on the audio track, for example, when aligning heads or making response adjustments to these machines we must make sure each unit ends up with adequate response at 25 Hz.

AUDIO AND OPTICS

As the audio is routed throughout the system, it will be turned on and off by devices such as the time honored relay, and in many cases by photo-optic devices. This device is a sealed unit containing a lamp and a light sensitive resistive element. In darkness, the resistance is infinity but in the presence of light it rapidly drops to zero or only a few ohms' resistance. By placing the resistive element in series with the audio circuit, the signal can be turned on and off easily and without direct connection to the control circuitry.

When troubleshooting problems in the system, it is well to know the routing of the audio through the system. There can be several problems along the pathway at different places. Knowing the circuit path makes it

Figure 1. A photo-optic device can be inserted in an audio path for a switching purpose.



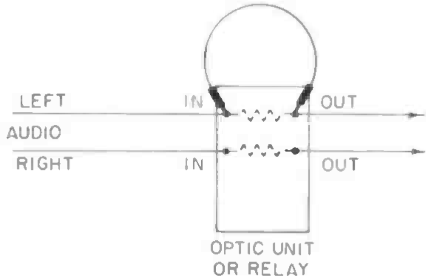


Figure 2. A quick test. Use a jumper clip across the resistance element or relay contacts to bypass the unit.

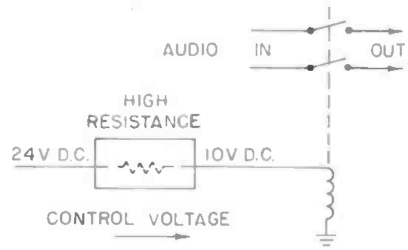


Figure 3. A high series resistance in the control circuit can reduce the control voltage below the point where it can operate the relay properly.

easier to signal trace when the audio fails to come through. Some types of sealed units have a pinhole at one end so that it is possible to determine if the light is on or not. Should the audio fail and you suspect a particular unit, observation of the pinhole will quickly determine if the lamp has operated. Even with the lamp on, the resistive elements may have failed and are effectively an open circuit. Should the unit be a plug-in type in a socket, then a quick replacement with a new unit will determine if the unit is defective or not.

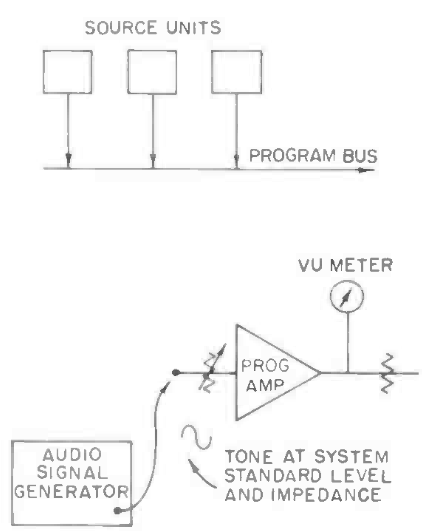
But if it is soldered to a p.c. board, it is advisable to make a few measurements before getting out the soldering iron. There will be a circuit diagram on the side of the unit to indicate pin connections to the internal elements. If the lamp is out, measure first to determine if there is control voltage present. But if the lamp is lit, then for a quick test use a jumper wire or screwdriver blade across the pins of the suspected resistive unit. If it is only a case of lower-than-normal signal, use an oscilloscope on the input/output to determine this. In the event of a plug-in unit where a substitution did not solve the problem, the same tests are in order. The problem may well be up the line and not at the photo-optic unit at all.

AUDIO AND DELAYS

Somewhere along the audio path you may also find relays switching the audio. In many cases, these are enclosed, multi-contact, plug-in relays. Audio failure can be caused by dirty or bent contacts, a defective coil, or low coil voltage. Problems can often be solved by quick substitution with another relay. But don't be too hasty about making that move! Only a couple of the contacts on that multi-contact relay will carry the audio; the others may be carrying d.c. control voltages or grounds. Pulling out the relay can interrupt control functions, and then all sorts of peculiar effects begin to happen in the system. Transients or multiple contact effects can turn on several sources and have them all playing on the air at the same time!

If the suspected failure point is at the relay, make some measurements as was done with the photo-optic unit. There will be a drawing on the side of the relay case to show the pin connections to the internal elements. Check first for adequate control voltage. Although voltage may be present, it may not be high enough to operate the coil properly. Low voltage means that some series element has changed resistance and is dropping part of the control voltage before it gets to the relay, or it can be the circuit which develops the voltage itself. Should voltage be present and correct, audio in but not out—there is a difinite relay problem. Poor contacts can

Figure 4. Preset the program amplifier to the correct gain and then use it to balance the individual source units.



Tandberg Presents the Next Generation

affect the audio response curve if they don't open the circuit altogether. Clean the contacts or replace the relay.

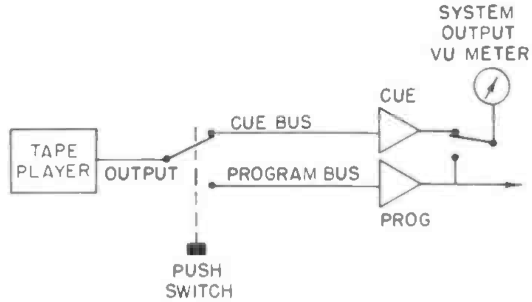


Figure 5. The cue channel is a separate circuit from the program channel, but it should produce identical results.

AUDIO SIGNAL LEVELS

Distribution of the audio signal within the system at a standard level is just as important as it is in a live studio. But the standard level here may not be the same as standard broadcast levels. This depends upon the design philosophy of the system. For proper set-up and adjustment of component units of the system, determine from the system manual what levels are standard for that system. If only the output meters of the system are used as the basis for adjustments, it is possible the individual units are set too high and input gain control of the program amplifier compensating to make the system output correct. Internally, however, the high level settings could cause cross-talk and distortion problems. Looking at the other side of the coin, the program amplifier gain may be set too high so that the individual units are running at too low a level. This could emphasize noise and interference.

Since tape decks ordinarily have no output meters on the units themselves in these systems, there are two ways signal levels can be set and units balanced. If there are no amplifiers between the source units and the program amplifier, feed a tone signal into the program amplifier at the *standard level of the system.* Adjust the amplifier gain control for zero dBm on the system output meters with the system properly terminated. Then run a NAB set level tape on each source unit of the system and adjust the units' gain controls for 0 dBm on the output meters.

If the system has intervening amplifiers between the sources and the program amplifier, another method must be used. In this case you must measure the output of each unit directly, but finding suitable checkpoints on the unit may be difficult unless they are on simple terminals at the rear of each unit. Finding such a suitable checkpoint, use your audio voltmeter bridged across the circuit. This will allow the normal circuit impedances to terminate the unit. Run the test tape and adjust to correct level on the voltmeter. The first method is the simpler one, if it can be done. At the same time, use the audio voltmeter to check the accuracy of the system's output meters. A zero dBm indication of the system meters should also be the same on the voltmeter. But you can also then actually measure the true output level the system delivers when its meters indicate 0 dBm.

IMPEDANCES

The impedances within the system may, or may not, be the standard 600 ohm broadcast distribution impedance. Since the circuits are all contained within the system, they may also be run unbalanced. The output of the system will be at 600 ohms balanced to interface with standard broadcast practices. Besides the fact a system may use its own standard impedance, it may not be a lumped impedance but distributed at several places over the circuit path. As far as the source unit is concerned when operating, it sees that impedance. But if you were to open the circuit path somewhere to make measurements or to feed in a signal, the path may have been opened out in the middle of this distributed impedance. So consequently, the measurements could be in error, or the adjustments produce erroneous results.

CUE CHANNELS

Most systems provide an audio cue channel so individual units can be checked and tape levels adjusted without also being on the air. This is a separate path and amplifier, with the audio from the unit put on the cue bus by a push switch and the system meters switched to the output of the cue channel. Since operational adjustments and listening tests are provided by this channel, it should produce identical results as the regular program channel. During a maintenance period with the system off the air and you are running response, level tests and so forth on the system, switch over to the cue channel and check it out for the same tests and compare results. They should be the same.

RECAP

A program automation system and a live studio operation have the same end objective, but each approaches it somewhat differently. Both require maintenance, measurements, and adjustments. Finding checkpoints in the automation is not always a simple matter, and it may have its own standard level and impedance, as well as different switching devices. Don't ignore the cue channel since most of the operational level setting and preliminary listening tests are done on this channel. ■

Voice Prints

● The idea of being able to identify a person by his voice print, just as you can by a finger print, is intriguing. It is possible—in fact it is being done. There are machines that will voice transcribe the sound of a voice into print, with some degree of reliability. But then we get into phonetics problems, dialects and all that.

The other day someone wrote to me, wanting to know where he could get, or if I could invent, a simple, inexpensive voice print unit, costing not more than $9.95! When I said it couldn't be done for that kind of price, he told me that such units were available, in fact had been for years. Further questioning educed the fact that he was thinking about voice *activation,* which is a different thing.

I told him that such a unit would be activated as readily by a passing airplane as by any particular person's voice, and he got the point. So the conversation turned to fingerprint recognition. This too, he assured me, has been done; there was a machine, invented some time ago, by which a thumb print was put on one side of a credit card. The presenter applied his own thumb alongside it. If it matched, the credit card was accepted, if not, the machine ate the card!

Here again, we have a different situation. A simple video scan of any two things, such as thumb prints, could be used with an electronic comparator network to determine whether the two are the same or different. But putting a particular print into a memory bank so that the same print would be recognized again is a little more complex. And putting a few million prints into the memory bank, so any print can be identified, is one the FBI is still working on.

Perhaps to the uninitiated voice prints and finger prints seem similar. A person can be identified by many unique things about himeslf or herself: no other person in the world will have exactly the same characteristic, whatever it is. The first of such identifiers to be used were fingerprints.

Then, fairly obviously, come foot prints (without shoes or socks, of course). After that came blood analysis, and possibly identification of genes in body cells. Then the phrenologist could analyze you by examining the bumps on your head, provided you didn't recently happen to bump your head. The dentist could take an imprint of your teeth, and so on down a long list.

THE REAL YOU

An intriguing thing about voice prints is the possibility of telling the difference between a real person and someone else impersonating that person. How is this possible? If they sound the same to our ears, would they not have identical voice prints? Not necessarily.

Perhaps an analogy that would help here is color identification. Have you ever matched a couple of colors by artificial light, particularly fluorescent lighting, and then found that, by daylight, the colors which appeared identical under fluorescent are, in fact, totally different?

Coming a little closer to voices, one of the problems in learning a foreign language, or even an unfamiliar dialect of your own language, is telling the difference between sounds that seem identical to you but appear quite different from each other to a person who normally uses that language or dialect.

I remember, I think it was Arthur Godfrey, giving a talk on the number of sounds that, to the Northern ear, sound the same as "all," as said by a Southerner. The point is this—to the Northern ear, all of those sounds seem the same. To the Southerner, they are not the same, and would never be confused. What are the differences? Something undoubtedly, that a voice print could demonstrate.

That may give an idea how a voice print might differentiate between sounds, but it differs from the purpose of telling one person's voice from another's, or indeed from anyone else's. That is one application for voice print technology that has received much notoriety, both in fact and fiction. It is one thing to tell the difference between different words spoken by the same voice, and between different voices, whatever they may say. The situations are different.

VOICE-OPERATE A TYPEWRITER

In one application you may want to voice-operate a typewriter. You don't care who is speaking, but you want the typewriter to transcribe whatever he or she says, whether he says it with a Northern or Southern accent—or perhaps some other intonation, such as English or Australian. A lot of people have difficulty even understanding a person with a radically different accent from their own. How much more difficult do you think it will be for a machine to take such different accents and correctly transcribe any of them into printed words?

This is the circumstance of using any of a wide variety of accepted dialect accents of the same spoken language. Then comes the problem in the fact that, whatever dialect you pick, the English language is far from phonetic. The letters *c* and *g* could be hard or soft, and the same vowels have a wide variety of sounds in different words, quite apart from the differences those same sounds may encounter between dialects.

Fairly obviously, such a machine cannot transcribe the words any more simply than a human being can. It must be given a word-recognition memory, equivalent to a person's vocabulary. Then, just as a human typist must do, it must be capable of transferring those words from their recognized audio form to the corresponding printed form.

You should be able to see that already we are a long way from a simple system. It is becoming quite complex. Just as a human being must do, the device must detect different dialect accents, and throw in a compensation, according to which dialect it determines—from a different set of memory banks and analyzers—is being used.

Such things are not impossible with modern technology, but they are far from simple.

DISTINGUISHING A VOICE

If you want to be able to distinguish one particular voice from everyone else's voice, it is not important what is said, but rather that the machine pick up on various intonations, or whatever other differences there may be between individual voices. If you think about this, you will start listening critically to voices.

Admittedly, given a complete voice print of a person speaking, you can identify differences from other voice prints, or other people speaking—preferably saying the same thing, to make comparison easier. You are conducting a visual examination of a print-out from an audio input. For the purposes of this discussion, we need not get into the details of how the print-out is arranged.

But we can profitably consider what the differences may be. All vocal utterances consist of a time sequence of varying amplitudes of a composite of audio frequencies within the audible spectrum. That is what must be displayed, in one form or another. And that is where you will discover differences.

From this viewpoint, you are using the same "raw input" as does the voice-operated typewriter: you are just doing something quite different with it. Perhaps you can begin to see why it is not the same as using finger prints. A person has just four fingers and a thumb on each hand. The only prints that a person can produce are limited to those quite fixed "inputs." The number of things a person can say extends to a vocabulary of thousands of words. The same words can be said with different emphasis, or in different tones. And different people say them with varying dialect pronunciation.

Our different problems require us to ignore, or to combine, whichever way

(continued)

you want to view it, differences that have varying significance, but are evidenced in the same domain—spectral analysis of amplitude/frequency composite sound, over time.

In one instance, you want to ignore the voice differences, the dialect differences, and come up with just words. If you then want to transcribe them from vocal words to printed words, that is another problem that involves a complex computer memory system to compensate for different pronunciations.

In the other situation you want to ignore different sequences of sound that identify words and concentrate on the more subtle differences that identify individual voices. One of the problems is that these differences—between words, between dialects, and between individual voices—all use the same set of variables, which are simple amplitude of various frequencies contained, varying over time.

LISTENING

As any machine that operates on voice analysis has to "listen" to the same sounds we do as human beings, it is useful to think about how we do those respective things.

Say you pick up the phone and recognize a familiar voice; how do you do it? What are the features about that voice that give it familiarity? Sometimes the recognition is instant, but at other times you may need to listen intently to distinguish whose voice it is. But what are the telltale sounds for which you listen?

Language is made up of vowel sounds and consonants. Both may play a part in voice identification. Each vowel uses a spectrum of audio for its sustained, or time-varying tone and each consonant has its more rapid variation in audio spectrum content.

The differences you hear may be made up of a special intonation on all of a person's vowel sounds. This is closely related to the differences that may be detected or observed between different dialects. Or the differences you hear may be more immediately triggered by a special way in which he or she sounds certain consonants. Some of these may also be associated with dialect differences.

Here is something else to think about. When I lived in England, the average Englishman would instantly pick up the fact that a person was an American, from his accent. But to an Englishman, the distinctly American sound would dominate, not what to an American would be a Northern or Southern accent; the Englishman would not know the difference!

Correspondingly, even in the British Isles there are literally hundreds of local voice accents that an Englishman can distinguish instantly, but with which the average American would have difficulty. What this says is that there are broader differences and narrower differences. But would you say that the difference between a Southern American and a Northern American is smaller than that between any American accent and any British accent?

I would say no, and I think you would too. In England, many Northerners cannot understand Southerners, and vice versa. The same is true in America. But in either country, those who get around hear more of people from other parts of their own country than they do of people from the other country. So the sensitivity to differences is more due to familiarity than to actual audio variances.

Perhaps I have said enough in this go 'round to give you something to think about. Can a machine have familiarity, or lack of it, in its memory system? One thing is evident: while none of these capabilities is impossible, the likelihood of any of these devices being available for a price like 9.95, is still a little way off. ■

JOHN M. WORAM

db The Sync Track

A Quick Look at the NRBA

• At about the time this issue of **db** arrives, some 2,500 people will be getting ready to converge on San Francisco, to attend the fifth annual convention of the NRBA. (On the other hand, they may already be there—or, on the way home again: you know how the mails are.)

In any case, the dates are 17-20 September, and the place is the Hyatt Regency Embarcadero Hotel. Some 150 exhibitors will be in the exhibit area, with hospitality suites open on practically every floor of the hotel. To some of our readers, this may bring up the question; "What (or, who) is the NRBA?"

The initials stand for National Radio Broadcasters Association. An NRBA press kit says, *"We're America's radio-only broadcasting organization. A volunteer, grass-roots organization of men and women who run radio stations. We think radio. We talk radio. We share a dedication to a healthy, productive future for radio. And we work hard to make it happen . . .We're for radio. If you are too, please join us."*

Well, **db** is for radio too, although too many of us recording-studio types tend to take it for granted. But, as I've found over the last year or so, a lot of our readers have some pretty strong thoughts on the subject. In fact, this entire issue of **db** takes note of reader interest in broadcast audio—in other words: in radio.

Never one to ignore an opportunity to get someone else to write my column for me, I asked NRBA president James J. Gabbert to tell our readers a bit more about the group. In his words, *"The National Radio Broadcasters Association is an association comprised of approximately 1,500 radio stations throughout America. These are all size stations, and cover all size markets. The goals of the association are relatively simple. By the end of World War II, there were approximately 600 radio stations in existence in America—today there are over 8,000. There is tremendous growth in radio, especially f.m. radio, and this has given the American public—more than any other nation in the world—the greatest diversity in radio programming.*

"Radio is a very competitive business, and what is happening today is that there is rarely a dominant radio station in any market. For instance, in Los Angeles, the number one rock-and-roll station enjoys only a 3.8 per cent share of the radio listening audience, whereas ten years ago it would have enjoyed a thirty or forty percent share. Because of this fragmentation, radio stations have targeted to specific audiences, both demographically and socio-economically. Every station is highly sensitive to its audience and reactions, and fights very hard for each listener.

"It is the NRBA's contention that the free market place—rather than the FCC—should determine programming standards, public affairs programs, and how much news should be broadcast. All of these should be established for each individual station by the listeners—its true constituency—and not by the federal government.

"Over the past ten years, the Fed-

eral Communications Commission has exerted more and more pressure in areas of programming and technology. One of the biggest reasons for quadriphonic sound's failure has been the FCC's lack of action in standardizing a system. The original "Petition for Rule Making for Quadraphonic Broadcasting" was filed by KIOI, San Francisco, in 1969. We are now in 1978, and no action has yet been taken by the FCC.

"Radio stations today are burdened with mountains of paper work to satisfy the federal government. The industry is saddled with many 1927 technical rules. The FCC currently inhibits new technology.

"The NRBA firmly believes that the free market place—the listening public—should be the judge as to whether new inventions are acceptable or not. In short, the NRBA is an association that is totally dedicated to reestablishing the control of radio to the people, and removing it from Washington."

Gabbert's point about the fragmentation of the radio audience may be particularly significant to the recording studio operator. With stations aiming at specific listener markets, it follows there is—or should be—a need for more music in just about every category. So, no matter what you're recording these days, there should be a station waiting for your tapes.

As for the FCC, let me quote from a recent NRBA memo. *"The Office of Management and Budget has awarded the FCC first prize for requiring the greatest amount of paperwork . . . more than any other independent federal regulatory agency (excluding the IRS). In a report to the President and Congress. OMB says the FCC has the largest number of recurrent forms reported.*

"Included in the list of the 15 most burdensome reports are four FCC forms and—you guessed it—radio station program logs are in first place!"

Hopefully, by the time this issue of **db** gets off the press, the Commission will have said something (by now, *anything* would come as a blessed relief!) about the status of a.m. stereo and f.m. quad. According to Gabbert's figures, it will soon be *ten years* since KIOI filed its petition. Oh well, I suppose these things can't be rushed.

As a matter of fact, another NRBA memo notes that, *"The FCC has postponed its review of both a.m. stereo and f.m. quad until August 8th. A heavy agenda was cited for the change and may delay hearings until September. In any case, no final decisions on a.m. stereo standards are expected until 1979 and a.m. stereo will most likely receive priority over f.m. quad. The Commission will probably issue a rulemaking asking for public comment."*

A heavy agenda? What are they up to now—designing *more* forms to fill out?

Well, despite what each of our readers may think (or not think) about a.m. stereo and f.m. quad, I hope all will rejoice with me that Uncle Sam has not appointed a federal watchdog to "guide" the recording industry. Who knows?—if there was such a thing as an FRC, we might all still be spinning around at 78 r.p.m. It's rather depressing that—after almost ten years—we are still waiting, and waiting, and waiting. (I certainly hope there's a "Stop the Press" announcement at the end of this column. Even if not, something *could* happen before September. Let's hope so.)

Anyway, this column started out by talking about the NRBA, so let me conclude with a salute to them for their efforts to keep radio moving ahead. With their help, the entire audio industry will profit. You too!

And for more information about the NRBA, write to them, at: 1705 De Sales Street, N.W. Washington, D.C. 20036 Suite 500 (202) 466-2030. ■

MARTIN DICKSTEIN

db Sound With Images

• Where were we? Oh, yes. Last month, we started to take a look at a conference that took place at the New York Hilton, June 6-9. This month, we will keep going with some more of the exhibits.

TELEFADER

Starting this month with the G's, GSA showed its Telefader, a device designed to permit optical fading (dissolving) in high intensity projectors. Because it is not possible to fade the light source on some high intensity slide projectors, and because the color temperature changes severely on those where it is feasible, this device was developed to provide a shutter module which goes in front of normal lenses and behind long throw lenses. The two-blade shutter can be set for a timing from full brightness to black from under 1.0 to 10 seconds. This unit can take signals from a computer, and is capable of operating projectors in any sequence or combination. There is also a flash capability where the unit cycles continuously in 0.9 seconds per cycle. With an under 1.0 second dissolve, the effect is that of a fast cut.

They also presented their Metrolite 35, a high intensity slide projector. With a 1200 watt HMI lamp, average life 1,000 hours, the output is 6,100 available lumens with a super slide, open gate, and 4 in. 2.8 lens and a 6 ft. wide image.

SLIDEMAGIC

Also among the exhibitors was Maximilian Kerr Associates, who showed their Slidemagic System. I've often discussed the possibilities of improving the look of slides for presentations, but never got into the actual means of producing slides. Here is a complete system for producing slides.

The software part of the system is specially designed to plan and prepare for the slide shooting. After ten years of developing and testing a method devised by Maximilian Kerr, this part of the system guides in the making of decisions as to what visual techniques to use, what colors work best, what type of styles are most legible, etc. It comes complete with manuals and 400 slides on graphic ideas, standard colors, stock graphics, and more.

The hardware side of the system is a full range camera system, designed to coordinate with the software portion. The camera is the SS-F2 Nikon; an optical grid provides perfect registration and reduces multi-image production costs. The entire package comes with a transparency platform, a duplicating stage, halogen lights for table-top shooting, and a host of accessories for maximum capability and efficiency. The producers even throw in two days of professional on-site training and a workshop to teach the functioning and handling of the materials.

FREEDOMIKE

Lectrosonics held forth at the south end of the exhibit. They showed their Freedomike™ System, which provides for wireless sound with up to four microphones. By using a single antenna and a precisely tuned receiver locked into just one signal by a special six-pole crystal filter and tracking oscillator, four receivers can be stacked, interwired, and fed to the regular sound reinforcement system already in use, running only one cable. Rechargeable batteries make the whole thing portable.

RANDOM ACCESS

Mast Development Co. was also well represented. They have a random access system which allows random selection of any of 81 slides in an average elapsed time of less than two seconds, with a maximum search time of only 3.5 seconds. The system consists of three parts: the projector (a Kodak Ektagraphic RA-960)/servo module, the electronics module with two short connecting cables hooking it into the projector, and the remote control unit, which is available in a variety of ways and can be as far away from the other units as 100 feet. The control unit has a focus capability which overrides the automatic focus of the projector.

They also have their Model 138-11 Random Access Dissolve Projection System which puts two projectors into a common output. This permits the use of this unit with t.v. film chain multi-plexers or flying spot scanners, or for rear screen effects for camera pickup. The system functions with fast dissolve between two projectors in either forward or reverse, random access or sequencing in either projector separately if desired, and a two-frame animation sequence capability.

Another Mast item is their new Automatic Random Access slide projector for use in exhibits, point-of-purchase locations, and information centers. This system offers two options. One is for the automatic sequencing of slides in an 80-slide tray with any slide chosen by random access staying on the screen until the sequencing is started again manually. The other is for slides to sequence normally at predetermined time intervals until a slide is chosen by random access. This slide then stays on for a predetermined time, after which normal sequencing begins again. An l.e.d. readout shows the slide that is on and the one selected.

Still continuing with Mast, they have what they call Omnisystem. It is capable of either two or three screen projection with either four (on 2-screen) or six (on 3-screen) projectors. The system is made in two parts. There's the Equipment Controller, and the Remote Control. Using this system, it is possible to project normally with a single drum at a time, or in dissolve. A digital readout is provided on the console to indicate which slide is up.

In conventional projection, slides in Tray #1 will go to 80, after which there is an automatic switchover to Drum 2. Slides are shown on the console as "1.15" (Drum number 1, Slide number 15), for example. (In dissolve, however, the slide is shown simply by a single number.) When Tray #2 is back to zero, the automatic sequencing in the normal projection mode starts Drum #1 again. In the dissolve mode, when a random slide is selected for projection, as the system is finding the slide in Drum #1, for example, Drum #2 is also automatically looking for the following slide in the sequence. This prevents any out-of-sync presentation of slides. If the number 0 is entered, and the all-select button is pushed, all slide drums will "home" and standby for restart of the entire program.

If during a normal presentation it is decided to select a slide at random, the slide at which this decision was made is entered automatically in the memory. After the random slide is shown, and it is desired to return to normal, pressing the Recall button will reset the drums where they were interrupted, for the continuation of the show. There is also a "clear" button for changing random selection or recall position. Another provision is for feeding a tape recorder through the system and the control of the environmental conditions of the room containing this console. And more, too—including the use of a pre-programmed audio tape with cues on one track for an automated show using all the capabilities of the control system.

CARAMATE

Otto David Sherman, Inc., slide developer for unique graphics and illustrations was represented, as was Singer, with their Caramate system to show slides on a self-contained rear screen synchronized with cassette sound. These units are also capable of front screen projection for larger audiences, and come with a special lens which can be flipped in place of the normal lens to enlarge the image in either mode of projection, thus permitting the slide to fill the screen whether it is vertical or horizontal.

Spindler & Sauppé also presented dissolve and control equipment.

VISUAL CHOREOGRAPHER

A device called the Visual Choreographer was shown by Talijon, Inc. This unit is described as "a time-plotting device designed to interface between multi-imaging programming systems and design personnel to achieve maximum synchronization of visual-to-audio cues accurately and efficiently." The system is said to be able to cut programming time by 60 percent and to increase accuracy to 100 percent. Real-time plotting is accurate to 1/20 of a second.

The designer sets up segments up to 8 minutes in length, then analyzes and converts to microprocessor cues. Correcting and rechecking is then possible until the show is as desired.

Telescript Monitor Prompting Systems shown by Telescript, Inc. This is the device that goes on the front of t.v. cameras and rolls along at the proper speed for the person in front of the camera to speak normally while looking directly into the lens.

Tiffen had a cassette player/recorder which incorporates a dissolve unit. The new Model SD-70 was recently introduced and contains provision for sync sound and cues with dissolve capability of 1.5, 4, and 10 seconds as well as fast cut. These four rates can also be used to alternate between the two projectors. In addition, the machine permits an output of the 1000 Hz signal for external use without the use of the internal dissolver. They have even designed a carrying case which will transport a two-carousel projector stacker with projectors in place.

To complete the alphabet, UniSet modules were shown by Kniff Woodcraft Corp. These come in individual shapes and are numbered so that set designs can be produced easily from 26 different geometric shapes. There is also a scale planning model.

Well, there you have it. Some things new, some old, but all interesting to those in the a/v or allied fields. If you care to get more information on any of these exhibitors at the Visual Communications Congress, don't hesitate to write to us and we'll try to help as soon as possible. ■

db New Literature

BATTERIES

A short-form catalog surveys and makes possible fast selection from the spectrum of possible batteries. Source: Panasonic Co., 1 Panasonic Way, Secaucus, N.J. 07094. (Ask for "Short-Form Battery Catalog.")

PROGRAMMER CALCULATORS

"The Hewlett-Packard Personal Calculator Digest," a magazine-format review of programmed and programmable calculators, presents a fascinating array of personal computers and pre-programmed application packs, covering everything from engineering math to a game of craps. Source: Hewlett-Packard, 1501 Page Mill Rd., Palo Alto, Ca. 94304.

LOUDSPEAKER HANDBOOK

Pocket-sized (3" x 5½") handbook provides a fast education in loudspeakers, including descriptions of the parts comprising speakers and an alphabetized lexicon of speaker language. Source: The Little Speaker Company, Inc., 78 Stone Pl., Melrose, Mass. 02176.

THUMBWHEEL SWITCHES

The switches described in this brochure offer flexible output code selection and up to 20 switching stations. Source: Switchcraft, 5555 N. Elson Ave., Chicago, Ill. 60630.

BURN-IN TEST SYSTEMS

Burn-in testing of semiconductor devices is the function of the systems described in this brochure. Source: Marin Controls Co., P.O. Box 490, Belmont, Ca. 94002.

INTERCOM

A catalog describes a line of headset intercom systems, particularly designed to cope with noisy backgrounds. Source: David Clark Co., Communications Div., 374 Franklin St., Worcester Ma. 01604.

AUDIO POSITION LOCATOR

An information sheet describes the Selectake II tape position locator. Source: 3M Company, PO Box 33600, St. Paul, Minn. 55133.

POLYPHONY

This is the title of a magazine devoted to electronic music. Subscriptions (U.S.) are $4.00. Free copy available. Source: Marvin Jones, Polyphony, P.O. 20305, Oklahoma City, Oka. 73156.

RF INTERFERENCE HANDBOOK

Detailed directions for correcting 25 different forms of rf interference are contained in this handbook. Not free, ($5.00), but worth looking into. Source: Sony Corporation, 9 W. 57th St., New York, N.Y. 10019.

ELECTRONIC WIRE

A 32-page booklet offers technical data on conductors, insulating materials, and circuit identification as they relate to electronic wire. Source: Brand-Rex Co. (Technical Data, ECI-78), P.O. Box 498, Willimantic, Conn. 06226.

ELECTRONIC COMPONENTS

Components used in speaker and audio systems, computers and other electronic systems and sub-systems, are listed in a catalog. Source: World Business Corp., 1669 E. Del Amo Blvd., Carson, Ca. 90746.

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Editorial

Since September is NRBA month (see this issue's *Sync Track*), it seemed to us to be a good idea to take a long look at what's going on in broadcast audio these days. For there's surely no denying that radio and recording are inextricably linked, with each dependent on the other for a large measure of success.

But how many recording engineers *really* appreciate the special problems of the broadcaster? Or for that matter, how many broadcasters understand what's going on in the recording studio? This month, we'll tackle the former question. (If we ever figure out what's going on in the recording studio, we'll get back to you.)

As our letters to the editor indicate, there's a lot of controversy about how a radio station should "sound," a point not to be overlooked by broadcast *or* recording engineers. So, we asked Eric Small to take a look at **The "Sound" of Broadcast Audio.** Although the subject is mostly subjective, Small's *Amplitude Distribution Histogram* may at least give the broadcaster a more effective tool for measuring his "sound," whatever it is.

Dolby f.m. also influences the sound of broadcast audio. And—as in the early days of recording studio noise reduction—there are a lot of misconceptions about its effectiveness. And then there's the question, "Is the station broadcasting Dolby-encoded programming, or isn't it?" The folks at Dolby Laboratories have been busy dispelling the misconceptions and working out ways to let you (and your receiver) know what's going on. But, see our **Dolby F.M. Up-date** for more information.

Talking about sound, what about the sound of a.m. stereo? Like many of us, WYFA-AM is waiting for the Federal Communications Commission to act. In Larry Zide's short photo story; **AM Stereo: Ready, Set, . . .?** it looks as though at least one station could be on the air with the sound of a.m. stereo almost within minutes of a favorable nod from the Commission. Let's hope it's not too much longer.

And, as Joe De Angelo tells us in **The Dawn of AM Stereo,** the concept wasn't discovered yesterday. He gives us a brief review of the various a.m. stereo systems now contending for FCC approval, and is enthusiastic enough to suggest that a.m. radio need not be permanently relegated to the strictly low-fi category.

Speaking of sound (still), what about the sound of some particularly offensive (or libelous, obscene, vulgar, etc.) remark, transmitted to millions of impressionable ears, over your very own radio station? Not exactly music to your ears, especially if the FCC is tuned in. (Apparently, they *are* capable of taking fast action, now and then.) Obviously, digital delay can help, as Richard Factor explains in **There Will Be A Short (Expletive Deleted) Delay.** With a minimum of grief, the interviewer can get through the program without fear of life's little embarrassing moments, when the interviewee gets carried away (verbally).

For that small broadcast facility with more ambition than budget, Louis B. Burke, Jr. tells us something about **Home Made Console . . . Why Not?**

And, since listeners often choose their favorite radio station according to program format (at least, we think they do), we thought it would be informative to explore the rationale behind broadcast format services. No doubt these services need no introduction to the broadcaster, but may be of some interest to our other readers. In **The Development of the Musical Format Service,** Loring Fisher explains the concept in detail.

Finally, for a change of pace, it's off to London, with John Borwick as our guide for a look at the recent **APRS Exhibition.**

J.M.W.

ERIC SMALL

The "Sound" of Broadcast Audio

An Amplitude Distribution Histogram is a computerized method of determining the precise nature of the sound being produced.

PROBABLY no subject consumes more human energy at a radio station than the never-ending discussion between the Program Director and Chief Engineer regarding the "sound" of the station. Is it hard or soft, boomy or shrill, heavy or light, muddy or crisp? Maybe it's too raspy or the bells don't tinkle or, most frequently, it's just "not loud enough."

The Chief Engineer defends him or herself against the above onslaught by producing last year's FCC audio Proof of Performance. It likely shows far better compliance with the Rules than the FCC requires. Harmonic distortion figures are often 50 per cent better than required, and 100 per cent better is not unusual. Frequency response is usually equally good. The Program Director responds by asking, "Why, if all stations must comply with the FCC Regulations, do various stations sound so different?"

THE PROBLEM

A fundamental problem exists with the FCC audio Proof requirements—the data is gathered under totally artificial situations and is thus worthless as an indicator of how the station will sound under normal conditions. Before doing a Proof, all limiters and AGC devices are either disabled or patched out; then the equalizer is switched out or set to flat. And, oh yes, the reverb system is shut down. Now, the system gain is so high that attenuation must be added in various places to replace the loss that the AGC devices had been providing dynamically.

Finally we are ready to begin measurements. The oscillator is plugged into the console, usually at the mic or turntable input. Good grief! The entire disc playback and/or cartridge record/playback system has been bypassed. Every major device that colors the sound (except possibly the transmitter, if it is an a.m. station), has been removed. And as a final blow to the validity of the Proof measurements, there are basic questions about the psychoacoustic validity of the Proof requirements.

All of this results in audio adjustments and equipment discussions critical to the "sound" of a station being made according to the whims of a "Golden Ears." This is akin to designing a bridge by instinct. Some of the bridges will stand, a few may even be innovative, but a lot will fall. And even worse, because everything is done subjectively, with no agreed-upon measurements, there is no way to transfer the innovative technology to later projects. Everything is folklore and rule-of-thumb.

It is not obvious why this problem exists, in light of the tremendous advances made in high quality sound reproduction over the past twenty years. There is, unfortunately, a basic dichotomy between the goals of sound reproduction and broadcasting. High fidelity reproduction has always had as one of its cornerstones nearly absolute gain linearity over the amplitude range of interest. A broadcast system is, by the FCC's own definition, acutely non-linear. I realize that such a statement flies in the face of the position clung to by many audiophiles and purist broadcasters. They feel that any audio processing by a station is bad and a distortion of the artistic intention of the musical performance. I agree with them. However I think they miss their mark when they attack the radio station. Their ire would be better if vented on the FCC. It is within the Commission's power to modify the modulation rules so as

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Eric Small is the president of Eric Small & Associates.

not to penalize the broadcaster who chooses not to process audio.

The over-modulation rules effectively demand that the station employ a non-linear system in order to remain legal. The FCC definition of over-modulation, as applied to broadcasting in general, is rather vague. However, the recently written Automatic Transmission System Over-modulation Rules are specific, and indications are that these standards will be applied to all broadcasting. The Rules state that ten or more occurrences of modulation greater than 100 per cent in f.m., are to be considered over-modulation. A "window" of five milliseconds is provided to allow multiple occurrences to be considered as one incident. In statistical terms, this means that modulation may exceed 100 per cent only 0.0833 per cent of the time. Put another way, you must be 99.91667 per cent sure that 100 per cent will not be exceeded in order to remain legal.

If a station did a live pickup of a concert and fed it directly into the transmitter with no processing and perfect amplifiers, the modulation would be 7-10 per cent rms. But modulation that low—for either an a.m. or f.m. station—would seem unacceptable to the vast majority of public or commercial broadcasters.

With the need for peak limiting established, the need for AGC follows almost automatically, to prevent excessive limiter operation. To comply with the Rules, the limiter must be a virtual "brick wall." Too-frequent running into that wall by the signal will result in unacceptable distortion.

Unlike limiting, all of the other mayhem done to audio signals, such as equalization, multiband limiting, reverb and the like, certainly does not have any direct basis in the Rules. However they all represent a *desire* to make the most effective use of the channel.

AN OBJECTIVE APPROACH

Suppose a 'scope were set up as an instantaneous modulation indicator and a very fast motion picture camera photographed several thousand times a second. Say this set-up ran on a typical music program off the air for several minutes. And when the film was developed, 128 bins were set up, each labeled in one per cent modulation intervals, 0-1%, 1-2%, 2-3% . . . 127-128%. Then cut the film into individual frames (about a million of them). Examine each frame and sort into the bin appropriate for the modulation level read. In a few months, when the sorting process is completed, a count of the contents of each bin is plotted. The resulting graph is called an *Amplitude Distribution Histogram* or, ADH. It reveals a great deal of information about high quality audio signals in general, and the particular audio chain of the station being studied, in particular.

Historically, broadcasters have attempted to judge their modulation by staring at the modulation monitor meter for a few minutes while jockeying the peak lamp control. If the range between the point where the peak flash lamp operates regularly and where it does not flash at all is small, then the peak modulation control is said to be tight. This generally indicates a good limiter.

The range over which the modulation meter swings also has significance, but exactly what, is uncertain. The wider the range of swing, the greater is the dynamic range. Too much dynamic range sounds good but results in low average modulation. Too little range sounds awful (square waves), but is loud. The problem with trying to make judgments of modulation from the monitor is that the resulting impression is very dependent on the program material and the exact response time of the modulation meter.

The Amplitude Distribution Histogram and its relatives replace the obviously subjective ("it looks okay to me") approach outlined above, with an objective, numerical approach to describe modulation characteristics. The amplitude distribution technique is to modulation measurement what t.h.d. and i.m. are to distortion measurements. Can you imagine the state of audio today without the ability to describe distortion numerically? The Amplitude Distribution Histogram is also useful in another way. It provides a new and very powerful way to think about modulation.

Here is some of the kind of information that can be derived from an ADH:

A) For the broadcaster, probably the most interesting part of the plot is the leading edge of the curve. Its shape describes the ability of the system to control over-modulation. In other words, it indicates the limiter characteristics plus the over-shoot of the transmitter.

B) Dynamic range is frequently referred to, but rarely described quantitatively. The range of modulation covered by 68 per cent of the area under the curve (standard deviation) is an easy, convenient way to describe dynamic range. It is expressed by converting the range of modulation bracketed by the standard deviation into dB by the formula:

$$\text{dB (dynamic range)} = 20 \text{ Log} \left(\frac{\text{upper modulation}}{\text{lower modulation}}\right)$$

C) The mode of modulation is that histogram bin that has the most number of modulation occurrences. If loudness is your goal, the higher the mode of modulation, the better the limiter and compressor are working together to produce a fully-modulated signal. If quality is the main objective, then the mode should lie well before the onset of limiting.

D) The arithmetic mean of modulation, more popularly called the average, is the sum of each modulation level recorded, divided by the total number of readings made. Going back to the high speed camera analogy, it is the value of modulation read off each frame, divided by the number of photographs taken. The mean of modulation represents how effectively the carrier is being filled. However it can also represent how badly a signal is being mangled. Infinite chopping (square wave signals) would produce modulation that approaches 100 per cent mean. Voice passed through such a system retains intelligibility but loses naturalness. Music is destroyed, almost beyond recognition.

There is a tremendous amount of information that can be gleaned from the ADH. I have touched on only the more obvious items. The mathematically-inclined reader is urged to consider how such statistical measures as skew and kurtosis relate to audio performance.

ADH is a research technique. I don't know if an ADH display system will ever find its place alongside the multimeter, oscilloscope and the t.h.d. meter as a general test instrument, but I do know it provides a readily-understandable tool for approaching modulation statistically—and in the real broadcast world of compressors, limiters and transmitters that bounce, ring, and tilt. Steady-state test signals—even random noise—provide little information about how the system really behaves dynamically. Statistical analysis of real program material offers the only means to understand and quantify system performance.

Today, producing an ADH in my lab takes a PDP-11 minicomputer and a raft of peripherals, not to mention a several-hundred-line computer program. Hopefully, the hardware can be simplified to allow a device to be marketed that would provide both the broadcasters and their regulators with a way to describe modulation in a rational, quantitative manner. ■

JOHN M. WORAM

Dolby F.M. Update

Dolby F.M. has been around for some time now, but it has not always been easy for the listener to know this fact. Herein, a proposal for an automatic switching system.

IN RECENT years, f.m. broadcasting has expanded its market penetration to stand on a virtually-equal footing with a.m. Technically, what distinguishes the two broadcasting methods is f.m.'s theoretical ability to reproduce program material with exceptional fidelity and relative freedom from atmospheric interference. However while advancing f.m.'s competitive position relative to a.m.. f.m. stations have borrowed various commercially-successful a.m. formulas such as promotions. contests. speed-rapping personalities, *and* various signal processing techniques. It has been assumed that one medium's success formula will produce a profit for another. and so. the tried-and true techniques developed for a.m. radio have been applied—almost by rote—to the f.m. side.

THE FM "SOUND"

The result has been a station-produced "sound" which is assumed to attract the attention of the listener who tunes across the dial. However, in many cases that "sound" makes f.m. almost sonically-indistinguishable from a.m. Switching back and forth between "loud" commercial a.m. and f.m. stations on a good car radio quickly supports this contention; it is difficult indeed to differentiate the medium which was originally intended for high fidelity broadcasting.

Unfortunately for the critical listener, these trends have resulted in the development of a number of audio signal processing devices which can diminish the broadcast fidelity of recordings and live performances. These devices are used by nearly all of the 4,000 f.m. stations in the United States, in degrees ranging from barely perceptible to extremely audible. For purposes of our discussion, we will

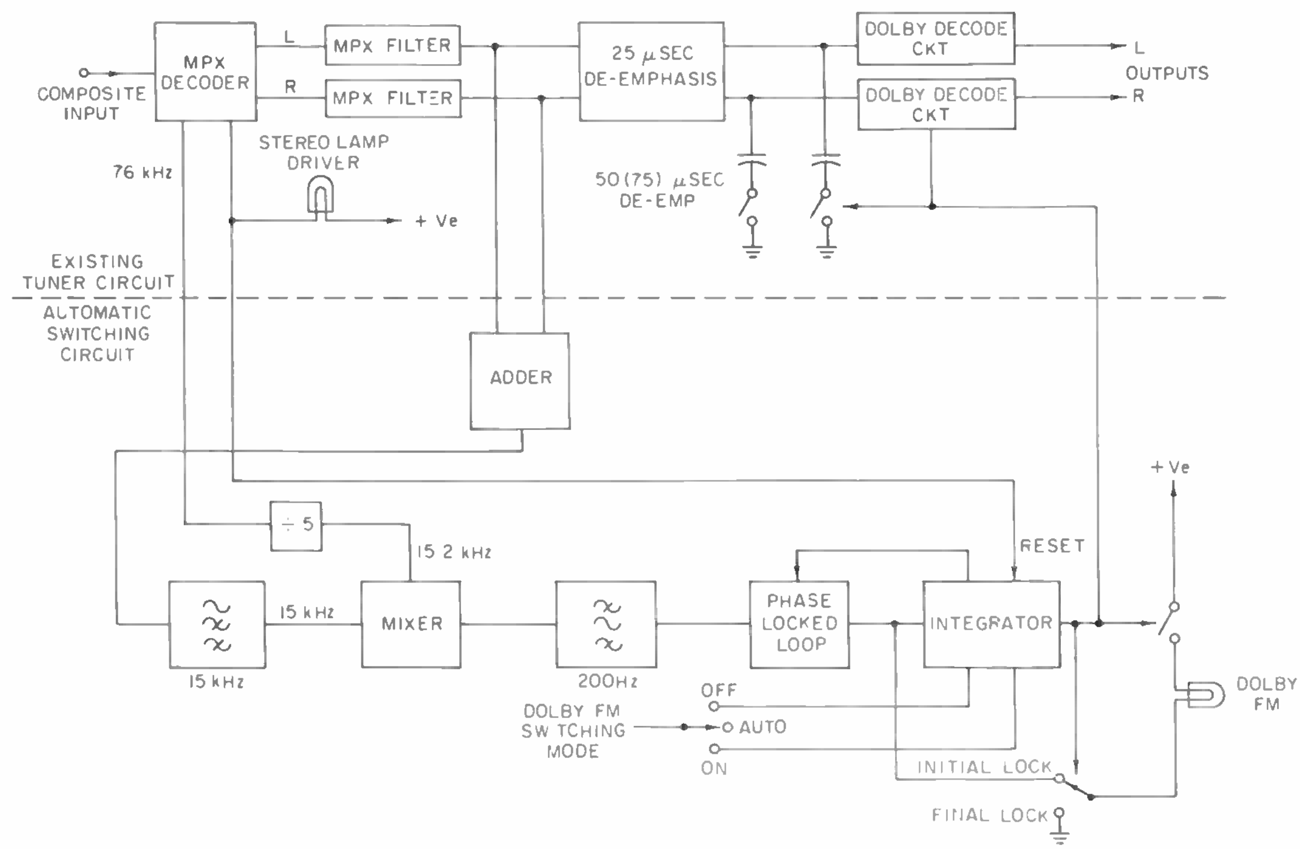

Figure 1. Automatic Dolby f.m. switching.

categorize the use of signal processing (limiting and/or compression) in three groups:

Minimal Processing uses the level-controlling device only during the most extreme program peaks—on the average. say, once per program selection.

Moderate Processing uses the level-controlling device regularly, but the amount of gain reduction rarely, if ever, exceeds 5 dB. To put it another way, the gain reduction indicator moves slightly in rhythm with the program material, but never approaches half-scale deflection.

Heavy Processing is constant use of the device on all programs, announcements and commercials. with rarely a microsecond of dead air.

F.M. PRE-EMPHASIS

The need for any f.m. broadcast signal processing stems from the 75 μs pre-emphasis required of all North American f.m. stations. 75 μs pre-emphasis is a single-zero network with the 3 dB point at 2.1 kHz, and boosts mid- and high-frequency information at a rate of 6 dB per octave. This results in a boost of about 17 dB at 15 kHz, relative to 1.5 kHz. Yet modern recording technology is easily capable of a flat frequency response up to 15 kHz.

This places the f.m. broadcaster in a dilemma. On the one hand, he can maintain a low average modulation for a wide dynamic range without high frequency over-modulation. On the other, he can use high frequency limiting and achieve a higher average modulation, but with restricted dynamic range. The former approach provides excellent fidelity, but inefficient use of channel space. The latter improves transmission efficiency, but forsakes fidelity. as the limiter provides a high-frequency loss at the transmitter which is not recoverable at the listener's receiver.

ALTERNATIVES TO LIMITING

An obvious alternative to f.m. limiting would be a reduction of transmission pre-emphasis, bringing it more in line with today's recordings. After all, it is the sharply pre-emphasized high-frequency program material that tends to over-modulate the transmitter, and so requires limiting.

The European standard of 50 μs offers some relief, but still produces a 13 dB boost at 15 kHz. A reasonable balance between the need to modulate the transmitter efficiently and to keep receiver noise down on the one hand, and the spectral content of modern recordings on the other, would be 25 μs. (Eliminating pre-emphasis altogether is hardly necessary, as even on modern recordings the level of high frequency harmonics is lower than the average low- and mid-frequency program material.) However, while 25 μs provides the reasonable balance, if uncorrected it would produce a dull sound on domestic receivers.

THE DOLBY NOISE REDUCTION SYSTEM

Enter the Dolby Noise Reduction System. In 1974, the FCC authorized B-type Dolby-encoded, 25 μs f.m. transmissions, provided the two techniques are used together. The loss of highs caused by a combination of 25 μs pre-emphasis (at the station) and 75 μs de-emphasis (at the receiver) is subjectively compensated for by the dynamic action of B-type Dolby encoding. High frequency headroom is increased by as much as 8 dB at 10 kHz, and a 5


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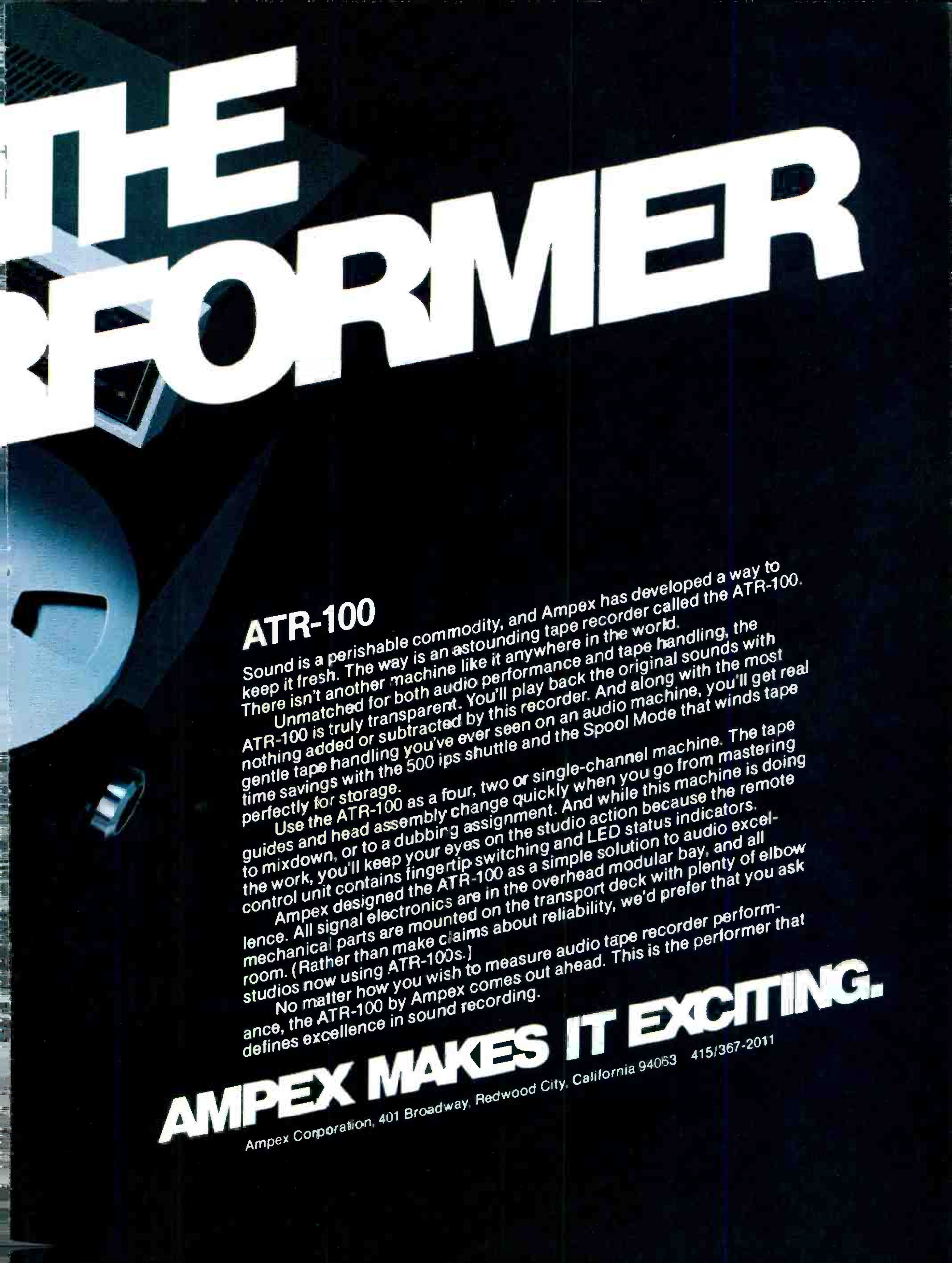
THE
RFORMER
ATR-100
Sound is a perishable commodity, and Ampex has developed a way to keep it fresh. The way is an astounding tape recorder called the ATR-100. There isn't another machine like it anywhere in the world.
Unmatched for both audio performance and tape handling, the ATR-100 is truly transparent. You'll play back the original sounds with nothing added or subtracted by this recorder. And along with the most gentle tape handling you've ever seen on an audio machine, you'll get real time savings with the 500 ips shuttle and the Spool Mode that winds tape perfectly for storage.
Use the ATR-100 as a four, two or single-channel machine. The tape guides and head assembly change quickly when you go from mastering to mixdown, or to a dubbing assignment. And while this machine is doing the work, you'll keep your eyes on the studio action because the remote control unit contains fingertip switching and LED status indicators.
Ampex designed the ATR-100 as a simple solution to audio excellence. All signal electronics are in the overhead modular bay, and all mechanical parts are mounted on the transport deck with plenty of elbow room. (Rather than make claims about reliability, we'd prefer that you ask studios now using ATR-100s.)
No matter how you wish to measure audio tape recorder performance, the ATR-100 by Ampex comes out ahead. This is the performer that defines excellence in sound recording.
AMPEX MAKES IT EXCITING.
Ampex Corporation, 401 Broadway, Redwood City, California 94063 415/367-2011

dB improvement in signal-to-noise ratio accrues as well, under weak signal conditions.

The Dolby process begins with feeding the peak-controlled signal program to the Dolby Model 334 F.M. Broadcast Unit, which applies B-type encoding, followed by time constant correction. Response between the 75 μs and 25 μs breakpoints is shelved by 6 dB per octave, so that the resulting signal comes out of the transmitter's conventional 75 μs network with a 25 μs characteristic.

This processing should follow the high-frequency limiter because to ensure proper decoding, the B-encoded signal should be unaltered once it is encoded. Limiter modification or the 25 μs characteristics is required (although some units, such as the Optimod 8000, require no component changes: the Model 334 is connected to points within that unit by means of a special cable supplied by Dolby Labs).

WHEN TO (AND NOT TO) USE DOLBY FM

Probably, Dolby f.m. is best suited to stations in the minimal processing category, because the process exists to provide listeners with an effectively "transparent" program channel—that is, an air sound which closely replicates the sound of the program itself. Judicious, occasional limiting should be virtually inaudible; when combined with the Dolby/25 μs technique, the listener can recover the signal virtually intact, in the form it left the studio, with a receiver equipped with Dolby decoding and 25 μs de-emphasis. Because the limiting acts only on rare occasions, there is virtually no difference between the sound of the program (console output) and the air sound (the demodulated, decoded, and properly de-emphasized signal).

For stations which apply virtually no limiting at all, the reduction in high-frequency pre-emphasis (for 25 μs, the 3dB point is 6.3 kHz) allows a 6 dB higher average modulation level. For stations using minimal processing, a marked improvement in coverage should not be overlooked by the broadcaster particularly concerned with high fidelity.

While moderate limiting is more of a compromise between highest fidelity and optimal modulation level, it can be inaudible most of the time, depending upon the limiter, the program material, and the sophistication of the listener. In such cases, the benefits of Dolby f.m. also accrue, with the added benefit of a reduction of intermodulation distortion caused by the gain reduction action of the limiter.

In the case of heavy processing (strong high-frequency limiting and multi-band compression) the benefits of Dolby f.m. are questionable. The dynamic relationship between the signal's spectral components is deliberately distorted to produce a very full, "fat" sound. This is desirable for a variety of popular formats, particularly when a large percentage of listeners are away from home, using automobile or car radios. Loudness is often the goal, overriding concerns for highest fidelity, and thus Dolby f.m. is of no particular consequence.

COMPATIBILITY PROBLEMS

In fact, with heavy processing, Dolby f.m. can produce compatibility problems. If the station broadcasts with a dynamic range of less than 25 dB, the signal level in general tends to be above the level at which the level-sensitive Dolby encoding takes place. Thus, the signal with its 25 μs pre-emphasis may sound consistently dull with 75 μs de-emphasis. As a result, a number of stations in the heavy-processing category have tried, and rejected, the Dolby f.m. technique. Because of their commercial success and popularity, this has led to some misunderstandings about Dolby f.m.'s compatibility in general. It's important to remember that if other forms of signal processing are used moderately so that the signal has at least 25 dB of dynamic range in the first place, the Dolby encoding sufficiently offsets the difference at high frequencies between 25 μs pre-emphasis and 75 μs de-emphasis, and so compatibility *is* most satisfactory.

AUTOMATIC DECODING SYSTEMS

A new feature of Dolby f.m. is currently under test in San Francisco, in cooperation with the FCC and Dolby f.m. stations KKHI, KQED, KRE, and KSAN: an automatic decoding system enabled by a pilot identification tone. These stations have been equipped under experimental FCC approval with Dolby f.m. encoders incorporating pilot tone generators, while specially modified consumer Dolby f.m. receivers are located throughout the nearby Bay area. The pilot tone is a crystal-controlled sine wave in the 15 kHz region, injected into the audio signal by the modified Dolby encoder at about 80 dB below 100 per cent modulation. This ensures inaudibility, while phase-locked-loop detector circuits in the modified receivers detect the tone for automatically switching the receivers to the Dolby/25 μs mode, and lighting an indicator l.e.d. This technique, so far proving successful, frees the listener from needing to know whether or not the station he has

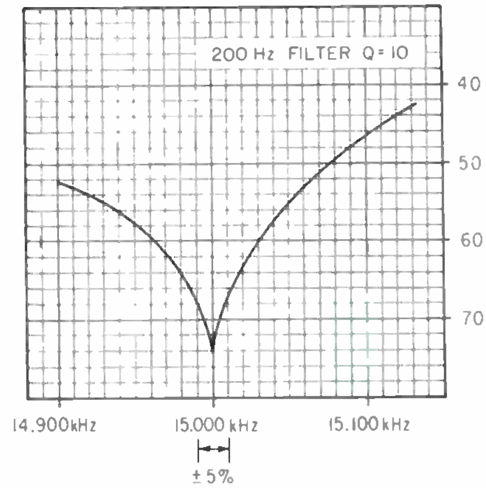

Figure 2. Selectivity of tone detection system, showing levels of continuously present adjacent frequencies required to unlock the phase-locked loop. The level of the 15 kHz tone maintaining the lock is —74 dB.

tuned in is broadcasting Dolby f.m. The system has been designed to handle a variety of tones, which could be used for a number of automatic signalling/switching purposes, such as four channel encoding/decoding processes. It is hoped that this new feature will be available to consumers and stations within two years.

DEVELOPMENT OF THE DOLBY PILOT TONE SYSTEM

Since the Dolby pilot tone system should find future use in a variety of broadcast applications, it may be of value to readers to describe how it works.

APPLICATIONS

The need for such a signalling system is not unique to Dolby f.m. applications. There are at present several system under investigation for control of transmitters, control and information display in consumer equipment, and for special applications such as traffic information. Such systems are often complex, as the broadcast authority tries to envisage all its present and future requirements, and to ensure that its chosen system is compatible with others under consideration.

The problem was to find a simple, inexpensive identification system (preferably costing less than a Dolby decoder) which would be simple to implement yet reliable in operation. One of the free benefits of the Zenith/GE pilot tone system is that the presence or absence of the tone must necessarily be detected in order to decode the stereo signal correctly. It is therefore a simple matter to provide an indicator lamp at little extra cost. By contrast, the Dolby-encoded signal has no special audible characteristic which can be easily detected. Indeed, this is the other side of the compatibility coin. No pilot tones are provided, since the information for correct decoding is carried in the amplitude and frequency characteristics of the signal itself. Thus, a separate signalling system must be added.

In the interests of simplicity, in-band signalling would be preferred. Some literature is available as to the inaudibility and suitability of such signalling, using the lower end of the spectrum, the middle frequencies, high frequencies, or a combination (see References 9-11). A general conclusion was that at high frequencies a level in the region of —60 dB below peak program level is usually imperceptible. This seemed promising, and research was commenced to see if this conclusion could be substantiated and, if methods could be devised to extract the signal reliably.

Our initial experiments consisted of merely passing the output of the f.m. detector through a very high-Q filter, and of analyzing the output of the filter. A frequency of 15 kHz was used as the signalling tone—chosen since it is right on the edge of the band (in the United States and most other countries). A level of —60 dB below Dolby level was used for the tone, and a simple phase-locked loop drove an indicator bulb. (Dolby level is defined as an audio deviation of ±37.5 kHz at 400 Hz.)

The first result was somewhat of a surprise. There is a significant amount of program information in and around 15 kHz, particularly on those stations employing heavy limiting. Moreover, spurious signals are caused by the clipping of program in the 5 kHz region, which produces harmonics at 15 kHz. Even using a very high-Q tuned circuit, there were many times when the phase-locked loop was thrown off lock. Secondly, young listeners could (just) detect the tone, at least in the absence of program. The approach did not look promising.

A study of standards laid down for minimum signal-to-noise ratios allowed by broadcast authorities for a complete studio-transmitter chain gave figures of 55 dB (CCIR/ARM) for the IBA (Independent Broadcasting Authority —England) and 53 dB CCIR/ARM for the FCC. (CCIR/ARM refers to a method of noise measurement. It has been described in A.E.S. preprint 1353. See Reference 12 at the end of this article.) It seemed reasonable therefore to choose a level of about —70 dB relative to Dolby level for the 15 kHz tone, which is quite inaudible under the loudest possible listening conditions using high quality monitor loudspeakers in a one-third octave equalized studio. With this level of tone, however, there was little possibility of extracting the tone at the receiver using the simple circuit described above, at least at a realistic cost. Thus, a new detection method had to be devised.

The method chosen was to derive an appropriate stable frequency in the receiver which is mixed with the upper portion of the baseband signal, thus transposing the high frequency components to more easily-manipulated frequencies. FIGURE 1 shows the block diagram. The 19 kHz multiplex pilot tone is used, indirectly, to derive this stable frequency, since this is controlled by the broadcast authority to—typically—one part in 10,000. It happens that most receivers today use phase-locked loops to perform the stereo decoding, and usually therefore there is a 76 kHz signal derived from the 19 kHz pilot available. Thus, a simple division by 5 yields a highly stable 15.2 kHz reference decoding tone, which lies conveniently near the edge of the 15 kHz audio spectrum. Since all decoders must perforce extract the 19 kHz component, all stereophonic tuner designs can be adapted to produce the 15.2 kHz tone.

This tone is then heterodyned with the 15 kHz tone, which is extracted from the program by a simple 15 kHz filter to produce a 200 Hz signal which is then passed via a bandpass filter to a tone decoder. Inexpensive commercially available phase-locked-loop tone decoders typically have an acceptance range adjustable between 2 to 14 per cent. We chose a value of 5 per cent, or ± 10 Hz, which is achievable without close tolerance components or other stability precautions. In comparison, to achieve a similar resolution at 15 kHz with direct detection would require an acceptance range of about 0.07 per cent. The output of the loop is used to switch in the Dolby decoding circuit, to change the tuner time constant from 50 (or 75) μs to 25 μs, and to operate the Dolby f.m. indicator.

Even with the very narrow acceptance range provided by this scheme, there are occasions when the signal content at 15 kHz is still sufficient to cause the loop to lose synchronism unless further steps are taken. On initial tuning to a station transmitting the 15 kHz tone, the loop is given a fast response time, and turns on in approximately one second. Once lock is achieved, the time constant is increased so that on the relatively rare occasions when the program's 15 kHz component is large, the loop maintains lock. Detuning the receiver produces a quick turn-off of the circuit, since the disappearance of the normal stereo signalling information is used to reset the system.

There is one further condition that must be met. Occasionally, the broadcaster may wish to turn off the Dolby encoding—for example, to make an A/B comparison, or when playing very old 78 r.p.m records, in which case the encoding tends to enhance the surface noise during compatible reception. It is relatively easy to provide a 3- to 5-second delay in the turn-off, so that the circuit still integrates out the 15 kHz burst of program information, yet turns off fast enough to cope with such intentional switching.

SYSTEM EXPANSION AND FUTURE APPLICATIONS

The system described is expandable to include more than one piece of transmitted information (for example, the identification of quadriphonic encoding). With an acceptance spread of 20 to 40 Hz, several frequencies can be used at the top of the 15 kHz band, or in the vicinity of 15.2 kHz. FIGURE 2 illustrates the selectivity of a loop (here tuned to 15 kHz); thus, the loop is only upset by a continuous 15.040 kHz signal when that signal level is greater than 16 dB higher than the 15 kHz tone. In practice, all tone levels would be identical.

Eventually—if and when there is a world standard on a more sophisticated data-transmission system along the lines proposed by several European broadcasters—such a system could also incorporate a Dolby f.m. signalling function which would co-exist with the simpler arrangements proposed here to deal with the immediate situation.

Based on quantities of 10,000, an approximate cost for the components in the detector circuit needed in each receiver is about \$3.75; thus the full Dolby f.m. decoding circuit, including automatic switching, is about \$9.00. A dedicated integrated circuit could be designed which would reduce the cost of the decoding circuit substantially. There is a little extra cost to the broadcaster, since the crystal controlled identification oscillator is incorporated into the Dolby B-type professional decoder. Units in use at present will be modified on an exchange of modules on a no-cost basis. ■

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LARRY ZIDE

A.M. Stereo: Ready, Set,....?

A brief visit to a local a.m. station, ready and waiting to "go stereo."

The new headquarters of WYFA-AM, waiting for stereo.

AS WE GO TO PRESS, we're *still* waiting for the Federal Communications Commission to tell us how and when a.m. stereo will become a reality. There is no doubt however, that it is coming "soon." Accordingly, progressive a.m. stations are taking steps to assure that they will be in a competitive position, once the FCC acts.

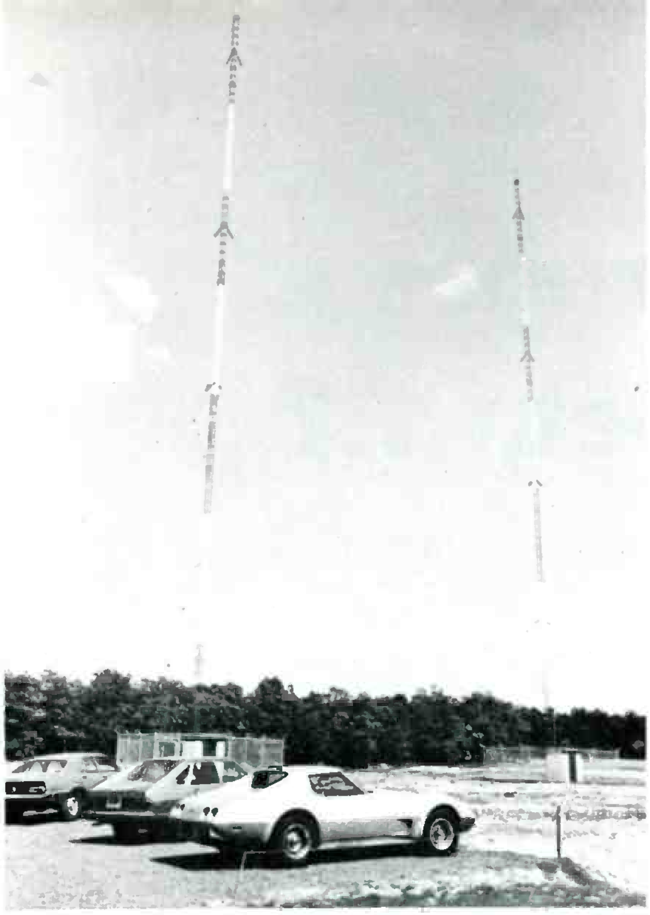
WYFA-AM is one such station that is all ready to go. Located in Patchogue, on Long Island's south shore, the station received its license in September, 1977. Their 1580 kHz frequency allocation is not new (it's the former WSUF-AM) but over the years there have been several owners. When a fire destroyed the old facilities, the present management decided to build a completely new one, from the ground up. Moving day was May 1 of this year, and the first official broadcasts began on June 20.

As a 10,000 watt, dawn-to-dusk operation broadcasting a middle-of-the-road format (MOR, as they say) they felt that stereo (when it comes) would be a natural for their new quality-sound operation.

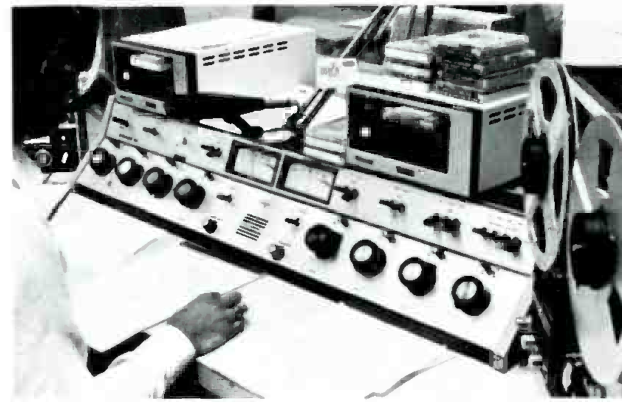
All studio and transmission equipment is housed within the new building, and chief engineer Gene Pfieffer reports that it took some elaborate shielding to keep r.f. from sneaking into the audio lines. In the control room, a single mast control/combo operation works through a new Stereo 80 two-channel console, manufactured by the Harris Corporation. All music is on tape, pre-programmed for convenience.

Control and stereo signal processing electronics are lo-

Signal processing and control electronics, with the 10 kW transmitter on the right.



Just outside the building, WYFA's twin towers reach 188 feet.
(No, they're not leaning over. Our wide angle lens is the culprit.)



An operator's-eye view of the Harris Stereo 80 board, flanked by Revox and TEAC tape recorders.

cated just outside the Master Control Room, where relatively light signal processing is achieved. For the present, the two stereo channels are mixed to mono for transmission, although the transmitter is already equipped with left and right channel modulation meters. In fact, WYFA's entire audio chain is "stereo ready." As for the transmitter (also from Harris), all that is required is the installation of the proper module (there's space waiting inside for it). Then, at the right moment, it will be "ready, set, GO!" for WYFA-AM STEREO. ■

JOSEPH De ANGELO

The Dawn of A.M. Stereo

Improved electronics have made the a.m. stereo dream into a possibility.

THE CONCEPT of a.m. stereo is not a new one. In the late 1950's, RCA Laboratories made a study of several a.m. stereo systems. RCA noted that, although quadriture systems were capable of good performance, the receiver design was complicated by the necessity of recovering the (unmodulated) carrier, for use in the demodulation process. They concluded that synchronous consumer a.m. receivers were impractical, which was certainly true with the technology available at the time.

But now, with modern integrated circuit technology and the development of phase-locked loops, carrier recovery is no longer a very complicated process. Therefore, a.m. radio may be about to undergo an exciting change. During the summer of 1979, it is expected that many a.m. stations will begin broadcasting in stereo. There seems to be no doubt in the minds of broadcasters, manufacturers and advertisers that—for a.m. stereo—the time has come at last!

WHAT ABOUT QUALITY?

In an effort to regain the listener rating shares that have been lost over recent years, the a.m. broadcast industry wants to start providing the same high quality stereophonic sound currently being enjoyed by the f.m. listener. With this in mind, the industry is getting ready to supply its listeners with an a.m. stereo service.

But, a.m. radio has long been considered a strictly "low fi" medium, and listening to most "modern" portable or car radios will quickly prove this point. But this does not answer the question, "Where does a.m. lose its quality?"

Joseph De Angelo is with the Harris Corporation

To find the answer, let's look at some typical audio performance data. FIGURE 1 shows measurement data taken from a representative example of a modern state-of-the-art thousand-watt a.m. transmitter. With such equipment on board, it can be seen that a.m. stations are capable of, and some indeed are broadcasting a full fidelity signal. So, since degradation at the transmitting site does not have to be the "weak link," this leads us to an examination of the receiver.

THE NAMSRC

The National AM Stereo Radio Committee (NAMSRC) represents three of the five manufacturers who have submitted a.m. stereo proposals to the Federal Communications Commission. As part of its work, the organization

FIGURE 1

Harris MW-1A 1 kW Solid State AM Transmitter Audio Response and Harmonic Distortion Measurements Taken at 95% Modulation:

Hz	Response Deviation	Distortion %
20	—.25	.30
50	—.30	.40
100	—.20	.27
400	θ	.27
1kHz	θ (Reference)	.25
5kHz	—.2	.60
7.5kHz	—.3	.80
10kHz	—.3	1.10

I.M. Distortion: 1.8% at 95% modulation
SMPTE Test 60/7000 Hz
4:1 Ratio

Noise (Residual AM): —62 dB below 100% modulation.

conducted audio performance tests on some typical monophonic receivers currently on the market. FIGURES 2 through 4 list the results. It can be seen that these receivers, like many others on the market, leave much to be desired in terms of audio quality. In comparison to the transmitted signal, consumers are simply unable to receive a good high fidelity sound, when listening to a.m.

INTRODUCING AM STEREO

Hopefully, the introduction of a.m. stereo will change this state of affairs. There is every reason to expect that new a.m. stereo receivers will indeed offer true high fidelity reproduction. In the new stereo receivers, frequency response can be essentially flat to 10 kHz, although selectivity requirements place restraints on higher audio response. Depending on the a.m. stereo system chosen, we will also note significant improvements in distortion performance. For example, Harris Corporation tests have proven that a.m. stereo need have virtually no distortion (less than 0.3%) under any receiving conditions (sky wave, selective fading, narrow bandwidth, or mis-tuning). In general, stereo reception is 20 to 30 dB across the audio band for each of the five proposed systems.

THE FIVE AM STEREO SYSTEMS

Five proposed a.m. stereo systems have been developed by the Harris Corporation, Motorola, Magnavox Corporation, Belar Laboratories, and Kahn Communications. Let's briefly touch upon the methods used by each proponent.

HARRIS — A quadriture modulation system, with a reduced L-R component equivalent to L and R modulation of two carriers separated in phase by thirty degrees.

FIGURE 2A

Audio Performance Data—Delco AM Pushbutton Auto Radio Model 70BP1.

Harmonic Distortion Versus Percentage Modulation: 1 watt output, 1 kHz modulation, volume control set at tone tap position (identified by slight plateau in volume control versus knob rotation). 1030 kHz.

Modulation %	Distortion %
10	4.5
20	2.0
30	1.5
40	1.7
50	1.7
60	1.9
70	2.1
80	2.5
90	3.2
100	4.6

FIGURE 2B

Frequency response at 30% modulation, volume control set at tone tap position, 5 MV/M signal at 1030 kHz supplied to receiver, tuning adjusted for minimum output at 5 kHz modulation, tone control at mid-range detent.

Frequency	dB Deviation From Reference
100 Hz	+6.2
400 Hz	+3.2
1000 Hz	θ (Reference)
2500 Hz	—4.8
5 kHz	—17.0
10 kHz	—39.0

FIGURE 3A

Audio Performance Data—Panasonic AM/FM/PSB Portable Radio, Model RF-1080.

Harmonic Distortion at 1 kHz, with 1030 kHz, 3 MV/M signal supplied to receiver.

Modulation %	Distortion %
10	1.6
20	1.0
30	1.2
40	0.9
50	1.0
60	1.2
70	1.8
80	2.3
90	2.4
100	2.0

FIGURE 3B

Frequency response at 1030 kHz, 30% Modulation:

Frequency	dB Deviation From Reference
100	+1.0
400	+5.0
1,000	θ (Reference)
2,500	—8.0
5,000	—12.0
10,000	—23.0

MOTOROLA — A quadriture modulation system with predistortion of the entire signal (not just the L+R sidebands) to force the modulated envelope to carry L+R information.

MAGNAVOX — A simple L+R amplitude modulation of an L-R *phase* modulated carrier.

BELAR — A system similar to Magnavox, except that the L+R information is used to amplitude-modulate an L-R *frequency* modulated carrier.

KAHN — An independent sideband system, with predistortion to force the modulated envelope to carry L+R.

COMPATIBILITY

Though consumers will need new a.m. receivers for stereo reception, each of the proposed systems is compatible with the mono receivers in use today. Compatibility with present-day mono envelope detectors—along with good stereo performance—has been a major hurdle for the five proponents to overcome.

At least 99.9 per cent of the mono receivers now in use employ envelope detectors for demodulation. The envelope detector is a non-linear device, and in general will generate distortion. However, under one special condition, distortion does not result. The special case is monaural a.m. with no negative over-modulation of the envelope, and amplitude and phase symmetry of the upper and lower sidebands. To generate a.m. stereo, it is necessary to make the upper and lower sidebands different in phase and/or amplitude. Therefore, in *all* a.m. stereo systems, envelope detections will generate mono receiver distortion. The five proponents have taken two approaches to solving this problem: 1) minimization of envelope detector distortion. 2) pre-distortion of the transmitted signal in an attempt to "cancel" the envelope detector dis-

FIGURE 4A

Audio Performance Data—Low Cost Pocket Portable. *Harmonic Distortion* at 1 kHz, with a 3 MV/M 1030 kHz signal applied.

Modulation %	Distortion %
10	2.4
20	1.8
30	2.1
40	2.4
50	2.7
60	2.8
70	3.0
80	3.4
90	3.6
100	3.2

FIGURE 4B

Frequency response at 1030 kHz, 30% Modulation:

Frequency	dB Deviation From Reference
100	—34.0
400	—10.0
1000	*0* (Reference)
2.5k	—8.0
5.0k	—25.0
10.0k	—35.0

FIGURE 5

(a)
No
Filter

10% 5% 0

0% 0% 0%

0% 3.3%

Belar Magnavox Motorola Kahn Harris

(b)
Elliptic
Filter

10% 5% 0

20% 15%

16.8% 19.9% 18.1%

15.2% 3.9%

Belar Magnavox Motorola Kahn Harris

(c)
Butter-
worth
Filter

15%

10% 5% 0

9.8% 11.7% 11.7%

9.4% 3.2%

Belar Magnavox Motorola Kahn Harris

(d)
Synchro-
nously
Tuned
Filter

10% 5% 0

4.0% 4.4% 3.4%

6.4% 0.9%

Belar Magnavox Motorola Kahn Harris

15.2 9.4 6.4

Monophonic receiver harmonic distortion for 80 per cent modulation at 3 kHz on the left channel only. (A) No bandpass filter-idealized case. (B) Elliptic filter, typical of receivers having ceramic i.f. filters. (C) Butterworth filter, typical of receivers with lumped L-C i.f. filters. (D) Synchronously tuned filter, typical of receivers having cascaded i.f. amplifiers and i.f. transformers.

tortion. The proponents have met this challenge with varying degrees of success. as shown in FIGURE 5.

SOME EXCITING NEW RECEIVER FEATURES

A.M. stereo will offer the consumer a new two-channel, high fidelity medium. plus some interesting features yet to be introduced in receiver designs.

All the a.m. stereo systems (except for Belar) incorporate a low-frequency identification tone. Unlike f.m. stereo transmission. where a 19 kHz pilot tone is used for synchronous detection. the sub-audible tone is strictly used for signaling and/or stereo/mono receiver mode switching. The stereo identification tone can also be used for other purposes. such as carrying low-speed digital data for station identifications that would appear on l.e.d. or LCD displays.

SUMMARY

Possibly. the FCC will approve and adopt one a.m. stereo system in the spring of 1979. Several leading manufacturers. including Pioneer and Sansui. have officially recommended to the FCC that it adopt the Harris system. However, most manufacturers are taking few chances, and are busy breadboarding most systems. The major semi-conductor companies are also gearing up for this new market by designing single-chip a.m. stereo detector i.c.'s.

The automotive market represents the biggest potential for a.m. stereo radios. Auto radio companies predict that a.m. stereo/tape units will probably surpass the f.m. stereo/tape unit combinations in popularity. The large home receiver manufacturers also forecast large a.m. stereo markets opening for their products. Several receiver manufacturers plan to capture an early share of the a.m. stereo market by having models available within three to four months of final FCC approval. A.m. stereo will have a dramatic impact on receiver sales. Estimates go as high as $20 billion. to supplement the 425 million mono a.m. receivers in use today.

A.m. stations will bring stereo listening to far wider audiences than are presently served by f.m. since a.m. has the ability to reach remote rural areas and mountainous regions where there is no f.m. penetration. The industry is getting ready for the day when it will be able to supply the listening public with new high quality audio, via the medium of a.m. stereo.

What will be the eventual impact on the recording industry? It's a bit too early to say, but it seems to be a safe bet that recording engineers and producers had better stop thinking of a.m. radio as strictly low-fi. ■

RICHARD FACTOR

There Will be a Short (Expletive Deleted) Delay

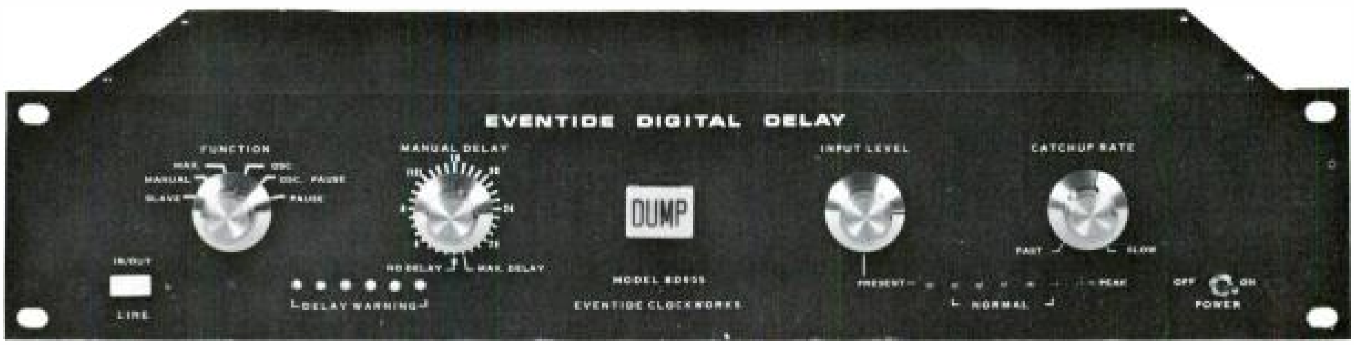
Solid-state delay equipment reduces the "sacred" seven-second pause.

MANY YEARS' standard practice has been reinforced by a recent Supreme Court decision: Broadcasters are RESPONSIBLE for, and must exercise CONTROL over programs they transmit. Transmission of patently obscene words (there are at least seven, according to one celebrated recording) is a serious no-no, and can subject the broadcaster to fines and possible loss of license, no pun intended. Transmission of libelous material can result in expensive legal action.

During normal operations, programs which are not clearly inoffensive are pre-recorded and subject to review by management and/or lawyers. However, live programming, and especially telephone "talk shows," are not subject to prior review, and the latter are not even subject to common sense restraint on the part of the telephone participant, as he is usually anonymous and free from fear of retribution, legal or otherwise. Because of the desirable features of live and 'phone shows, such as timeliness and audience participation, a method to reconcile the contradiction between instantaneous broadcast and responsible censorship must be employed.

The method normally employed is DELAY: The "live" program is delayed for a short period, typically seven seconds. Obviously, little immediacy is lost because of this short delay, but the time is sufficient for a control operator, usually the air personality in the case of telephone shows, to censor any undesirable material that he may hear at the input of the delay by cutting off the output of the delay. Several problems are inherent in this procedure. Among them are the physical realization of the requisite delay, and the necessity of substituting something for the program material deleted (or, of course leaving up to seven seconds of dead air, a normally unthinkable alternative.) The methods used to achieve the delay, their advantages and disadvantages, are the subject of this article.

Richard Factor is vice president of Eventide Clockworks of New York City.



BD955 Broadcast Delay Line

THE SEVEN SECOND DELAY

The standard "seven second" delay alluded to above is not sacred. It evolved, as have many standard practices, as a result of necessity. Before the days of solid state technology, there were only a few possible means of delaying signals for any significant period, and all of them were mechanical. The only possible solution was to record the signal, and play it back later. Disc recording was a possible but clearly impractical method. Tape recording was obviously the way to go because the medium could be re-used, and was potentially continuous when spliced into a loop. The seven second delay was derived from the physical distance between the record and the play machine!

The actual system consisted of "recording" the program in process on one machine, then spooling the tape past the playback head on a second machine, and then onto that machine's takeup reel. It was clumsy, but it worked. If something was said that was unsuitable for airing, the playback fader was brought down until the stretch of tape was passed.

The above primitive scenario still exists (with good reason) at some stations today. The obvious disadvantages in terms of space, inconvenience, personnel, and equipment tied up can only be offset by frequency, or rather infrequency, of use.

The advantage of the method is, of course, that it requires no capital expenditure. Any station that doesn't have two tape machines probably can't pay its phone bill either, and therefore doesn't really have to worry about telephoned obscenities being aired. Other than as a historical perspective, however, method 1 is of little relevance. There are several obvious improvments over the "two-tape-machine-and-don't-bump-into-the-tape-thank-you" school of delay, and the standard delay methods employed today take advantage of them.

LOOPS

Tape can be spliced into a loop. This obvious statement has been pyramided into an industry. Tape loops are used in the ubiquitous cartridge machine for storage, and in many special purpose devices for delay. They are used in concert hall simulators, rock & roll special effects echo units, and, of course, broadcast delay equipment. The first use was probably home brew; a station engineer got tired of setting up the two tape machines in method 1 and put a few tape guides on the outskirts of one deck. Then, simply adding an additional PLAY head before the ERASE head enabled the machine to be used normally in RECORD mode, and the delay signal was derived from the newly installed PLAY head after the loop had traversed its peripheral trajectory.

The advantages of this scheme are apparent. The most obvious is that only one tape machine is required, and the tape loop, being continuous, does not require constant supervision. The disadvantages are that one tape machine is still required, and that the tape loop can break, and certainly will deteriorate. If the advantages and disadvantages seem to have elements in common, it is because they do. After all, the purpose of a tape machine is to record and play, not to delay. Again usage factor becomes significant. If the tape machine is to be used a few hours a week for delay, it probably saves a lot of money. If on the other hand it is to be used a few hours a day for delay, the savings are not so obvious. It is not the ideal solution because it does require setup and some supervision, if only to switch modes and make sure that a new, clean, properly spliced loop is in place, and the machine is unavailable for normal use during those hours it is used for delay. At this point, scheduling plays a part. If the talk show is at night and the tape recorder is used for production during the day, using tape delay is an economy. If both activities are simultaneous, attempting to use one machine for both will ultimately turn out to be foolish, given the ever-increasing cost of personnel.

Notice that up to now, nothing has been said about filling dead air. Any straight delay method will have pockets of up to seven seconds of no-program material. This time must be filled in some fashion. One method is to have the air personality use a "live" microphone—one placed after the delay system—to announce that a portion of the program is being edited. This may be effective, but can be disorienting to the announcer who must keep track of what's happening in two different time frames! Another method is to play a cartridge containing the announcement, a jingle, etc. Since this must be instantly available, it requires one more cartridge machine that would ordinarily be used for the program in question.

ROM

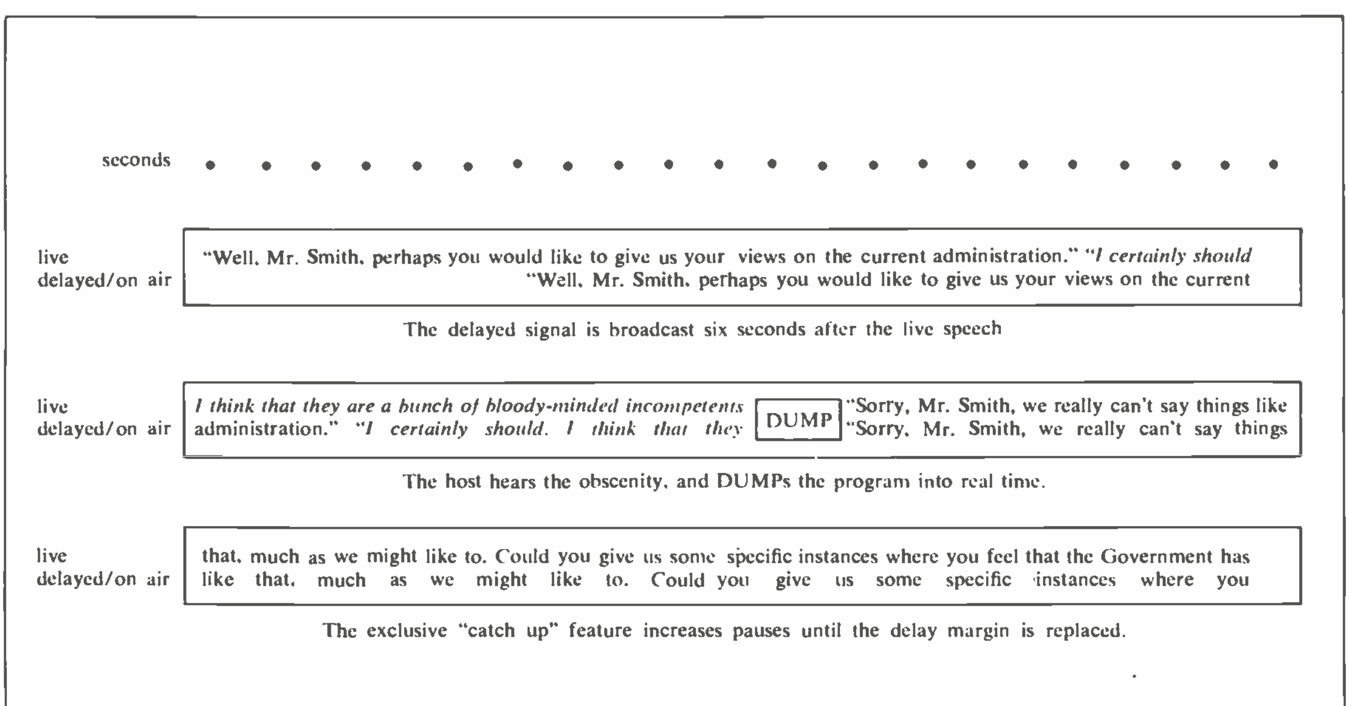
An interesting variation of this theme, and one which not-so-coincidentally eliminates the necessity for the cart machine, is a system devised by Pacific Recorders. This consists of an automatic tone generator using a ROM (Read Only Memory) to store the tone sequence. The device is triggered manually, after which it automatically provides the timing function necessary to kill the offensive material, and simultaneously fills the dead period with anything from a musical signature to a Sousa march (or presumably, a Bronx cheer!) Another variation on the tape delay scheme has been introduced by ITC, a tape cartridge machine manufacturer. Their unit takes advantage of the fact that the standard broadcast cartridge is already an endless loop, of any desired length. By using a four foot tape wound in the cartridge, a seven second delay is accomplished. A slight rearrangement of heads is all that is required to turn the cart machine into a delay line. The same comments with respect to supervision and economics apply here.

Until recently, a station devoting a large portion of its broadcast day to live programming had no alternative to tape. However, the decreasing prices of solid state memory devices has made the digital delay line an increasingly attractive option.

Briefly, a digital delay line functions by sampling the input signal and storing the samples as numerical values in a memory of some sort. After a specific period, the stored number is converted back to an analog voltage and sent on its way. The advantage of this system is that once the number is stored, it is not subject to change or degradation. There is no mechanism comparable to oxide flaking or tape head wear in digital technology. Thus, employing digital techniques should result in orders of magnitude less supervision, and greater reliability. Because it is completely non-mechanical, there should be no preventive maintenance requirement, and its frequency response, dynamic range, and freedom from other tape problems such as wow, flutter, and potential jamming as in the case of the cartridge, are constant. The only disadvantage has been, historically, the price.

The amount of delay achievable depends upon the digital storage capacity available. The required number of "bits" of storage is a product of the delay time, the dynamic range, and the frequency response. The relationship is linear. Cut the frequency response in half, and the delay can be doubled for the same amount of storage. Because broadcasters require long delays and good frequency response, the digital memory requirement is for-

Diagrammic exposition of the "catch up" feature of the Eventide BD955 Broadcast Delay Line.



midable. To achieve seven seconds of delay at a 15 kHz frequency response with good dynamic range requires about 2.5 million bits of storage! As few as six years ago, a chip with 256 bits of memory was considered "large," and most small computers had only a few thousand bits of data storage. Now that RAMs (Random Access Memories) with about 16 thousand bits of storage are routinely available, consideration of long delay lines without concomitant consideration of refrigeration plants for cooling ten thousand i.c.'s is possible. As if by magic, three manufacturers have announced such products within the last year.

When considering use of solid state delay lines, it should be remembered that although in the one limited instance of obscenity/libel policing, they seem to perform a function identical to that of the tape delay, they are entirely different beasts. For instance, the non-mechanical but very high cost of the storage medium (plastic/metal oxide tape *vs.* silicon chips) contrasts with the high cost, low versatility, mechanical transport mechanism *vs.* the fairly cheap and incredibly versatile capabilities of the delay line control circuitry (actually a small, special purpose computer).

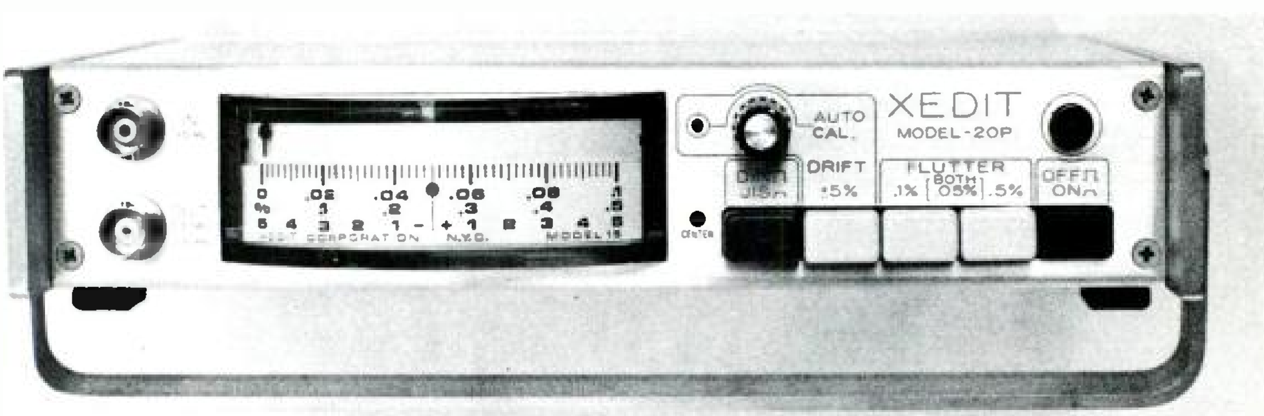
In recording studios and some radio production facilities, the delay line has been used for years as a special effects tool for such purposes as adding reverberation, choral effects, flanging, speeding up/slowing down of tape, and even more exotic effects such as tunnelling and continuous short-duration repetition. There is nothing fundamentally different about these recording delay lines except that the control circuitry was designed for special effects, and the memory capacity is much, much smaller. Adding some features, such as variable delay, exacts a small cost penalty, and at least one of the broadcast delays, the Eventide DB955, has a front panel control that allows varying the delay from milliseconds (for doubling and other special effects) to hundreds of milliseconds (for echos and video/audio synchronization in cases where t.v. audio is transmitted by landline and video by geosynchronous satellite), to the multiple seconds required for obscenity deletion.

RETHINKING METHODS

One possibility allowed by the solid state delay is the rethinking of the method of use. The tape machine is limited by physical constraints. One cannot physically chop out a piece of offending material and splice the ends together in real time and maintain program continuity. However, if one could do that, it should be possible to continue the program without interruption! All that would be necessary in that case would be to find some way of building the delay up to a satisfactory value before allowing uncontrolled material back on the air.

To continue with our analogy, let us assume that one can make instantaneous splices in tape while in motion. We have just spliced out an obscenity and our delay is zero. What now? Let's keep listening as the program goes along: The announcer is still speaking. He says "Sorry we had to cut you off, but you can't say that on the radio. . . ." What do we notice about his speech? He pauses microscopically between words! Why don't we splice some blank tape between each of the words? Well, of course (ending the analogy), we don't because we can't move fast enough.

However, solid state circuitry certainly can, and does! When the caller utters a profanity, the operator presses the "DUMP" button, at which time the delay instantly goes to zero and an auxiliary relay closes. (The relay may



be used to control a phone cut-off or whatever.) The zeroing of the delay effectively eliminates the profanity, and the caller is gone. The process of catching up now begins: A level detector circuit waits for pauses in the announcer's speech. As each one occurs, the delay is automatically increased, lengthening the pause and increasing the delay margin between the input and output.

A series of front panel l.e.d.'s indicate how much delay margin is present. Because this process of editing without interrupting the program is virtually inaudible on the air, it makes for more natural programming and fewer and certainly less obvious breaks and transitions.

Unfortunately, the catching-up process does require more time to build up a satisfactory delay margin than simply stopping the program and filling the otherwise dead air. For this reason, it is usually desirable to go back to live programming before the entire delay is available. To this end, an internal jumper may be connected to the various indicators to operate a master indicator telling the operator that it's safe to resume live programming. This indicator may also be connected to a relay which automatically silences the phone for the requisite period.

COST vs RESPONSE vs DELAY

As was stated earlier, the seven second delay was arrived at more by tradition than by consideration. There is nothing in FCC regulations requiring seven seconds (or any delay, for that matter). The requirement is simply that the broadcaster exercise control over his transmissions. Because the cost of solid state delay is critically dependent upon delay, it can literally be looked upon as so many dollars per second. When selecting options on a digital delay, one should remember that dirty words are typically "Anglo-Saxon monosyllables," and are easily uttered in a couple of hundred milliseconds. It hardly seems necessary to provide seven seconds of delay to guard against this sort of thing. In fact, one station which purchased an early model solid state delay specified a 1.5 second unit for reasons of cost, which has proved to be adequate over many years of operation. This is not to say that there aren't good reasons for getting longer delays. For instance, an eloquent speaker might manage to utter a libelous remark and not have it recognized as such until it had already been transmitted. In such cases, the additional margin provided by a long delay may be warranted.

Another consideration is frequency response. Cutting response in half eliminates half the storage requirement and a good percentage of the cost. This is a bit more difficult to justify than compromising on delay time. If you are sure that only speech (especially telephone speech) is to go through the delay, it is perfectly okay to get a restricted response unit. Remember, though, that the entire program, not just the speakers or callers, must be delayed—or you will need some very fancy synchronization indeed!

For broadcasters occupying or considering occupying a large percentage of the broadcast day with talk programming, it is probably a good idea to consider a solid state delay line for obscenity policing even if appropriate tape equipment is already in place. As personnel costs increase, the lack of routine maintenance and setup required by these units will ultimately result in a substantial saving, while their new features allow programming decisions based less upon equipment limitations. The cost of a full capability delay unit is now comparable to that of a quality tape machine. ■

LOUIS B. BURKE, JR.

A Home Made Broadcast Console...Why Not?

Here's an easy-to-follow recipe for cooking up your own console.

THE AUDIO CONSOLE is often the single most expensive piece of equipment in either the control room or production room. Because it is the proverbial heart of the broadcast system, broadcast engineers are naturally concerned with the quality and reliability of this important piece of equipment. So much so, that the thought of designing and building one's own console is immediately rejected—if it is considered at all.

But, while the initial thought of actually designing and building an audio console is somewhat awesome, the question of its feasibility is beginning to raise its head more and more often as some radio stations all-too-quickly outgrow the capability of their existing equipment. When this happens, the station engineer is faced with the age-old conflict of reconciling need with cost. The "old faithful" console should be replaced with one which features additional input and output capabilities, along with a multitude of switching components for additional flexibility. However, the cost of the new console must be strongly considered and compared against features available in order to arrive at the best dollar value.

I recently became involved in a situation where I was faced with such a conflict. The needed console would require a mix capability of six channels, two mic inputs and four high-level inputs, with a few switchable remote high-level inputs. Shopping around only aggravated my frustration because I was finding out that a console with the aforementioned capabilities fits into a unique category. Simply stated, it is too small to justify the expenditures required to incorporate features that would be excessive and unused, yet the features we did need were not present in the smaller, more inexpensive consoles available.

In this day and age when the price of everything is continually rising, why shouldn't a business that has prided itself in revolutionary technical concepts from its inception also pride itself in a concept that is both economical and technically sound? At this point, my competitive nature compelled me to at least give a project of this magnitude some serious thought, and after looking at the possibilities from numerous angles, then testing my

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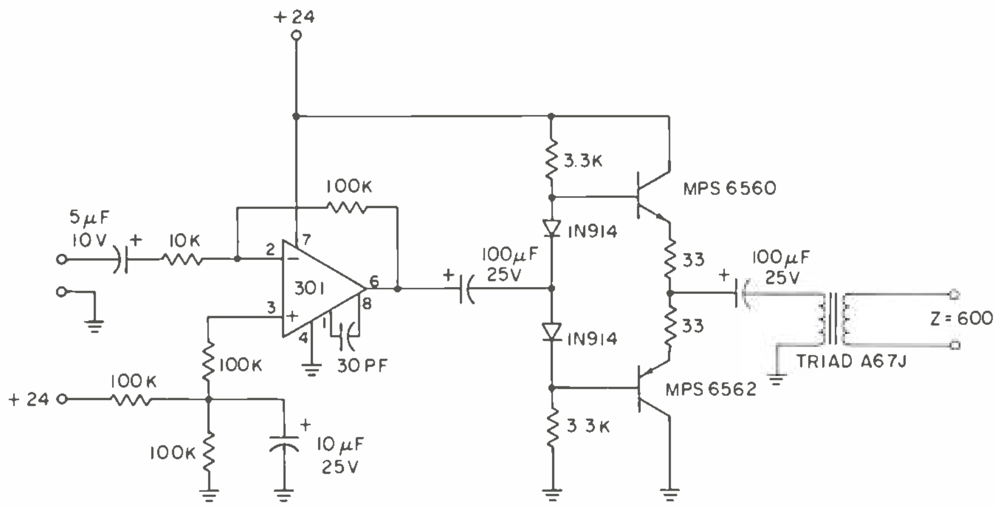


Fig. 1. The basic circuit described by the author.

ideas on the drawing board. I decided that it might indeed be feasible to build my own console.

After spending more hours of "back to the drawing board" than I care to remember, and confronting more problems than I thought possible, I did build my own custom designed console, and at a much lower cost figure than what I would have paid for a commercially built console with comparative spec's and features.

GETTING THE ELECTRONICS

Following are some of the questions I asked in getting to the finished product, and some of the answers I found. One of the first real questions in confronting the construction of your own console is how to acquire the necessary electronics. I decided there are two options: buying the necessary printed circuit boards from one of the many manufacturers, or, designing and building my own boards.

If the first option is taken, there are numerous modules available; just browse through any current trade magazine and take your pick. Any of the available modules are of excellent quality and are capable of producing fine results when packaged into an audio system. But the cost of these boards will certainly put a sizeable dent into your budget. If the second option is considered, there are several very good books available on circuit design by such notable authors as Walter Jung and Dennis Bohn, and many papers by engineers presently involved in research for the major parts manufacturers.

I decided on the second option. I knew in the early design stages many parameters would be decided, such as input and output impedances, acceptable frequency response, harmonic distortion limits, signal-to-noise ratio and maximum output level. So, it was off to find all available books and articles by the aforementioned writers, then to read and theorize for many hours. I also bought a ream of paper and several sets of batteries for my calculator in preparation for the task that lay ahead.

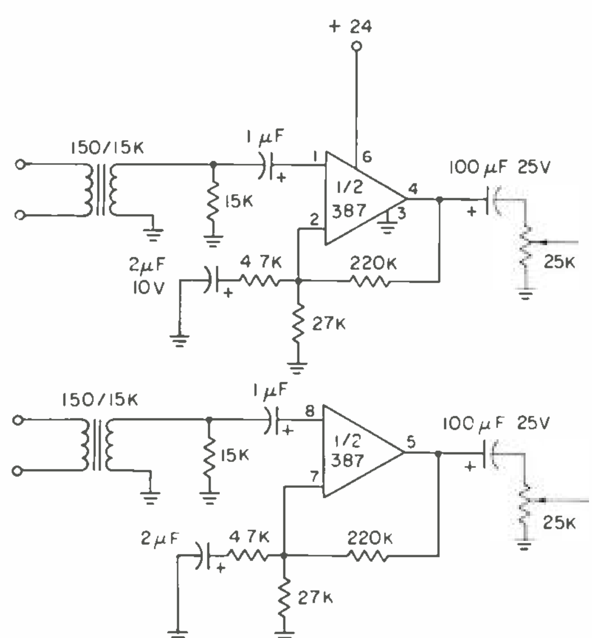
The ideal design, I finally decided, would be to use one plug-in module to serve all the necessary electronic functions. This would reduce the number and types of spare replacement parts that would have to be kept on hand for maintenance, thus reducing the maintenance inventory and cost. Further, it would simplify trouble shooting; hopefully to nothing more than replacing a suspected malfunctioning part or board with one that was known to be working. As a matter of fact, it would be wise to build up at least one extra board to be used for just such a purpose.

The next step was to purchase a "proto" board assembly and arm myself with several rolls of Number 22 gauge solid wire in preparation for bread-boarding many of the circuits that seemed promising. After many days of plugging in hundreds of components and evaluating each individual circuit, I finally settled on a very simple arrangement using an integrated circuit driving a discrete pair. (See FIGURE 1)

AMPLIFIER

This amplifier provides: 20 dB of gain, with audio response flat to within a couple of tenths of a dB across the audio spectrum (20 Hz-20 kHz), maximum harmonic distortion of 0.5 per cent, with a signal-to-noise ratio of 85 dB below maximum output, which is +22 dBm. Granted, these are not the spec's of a $50,000 console.

Figure 2. The dual microphone preamplifier.



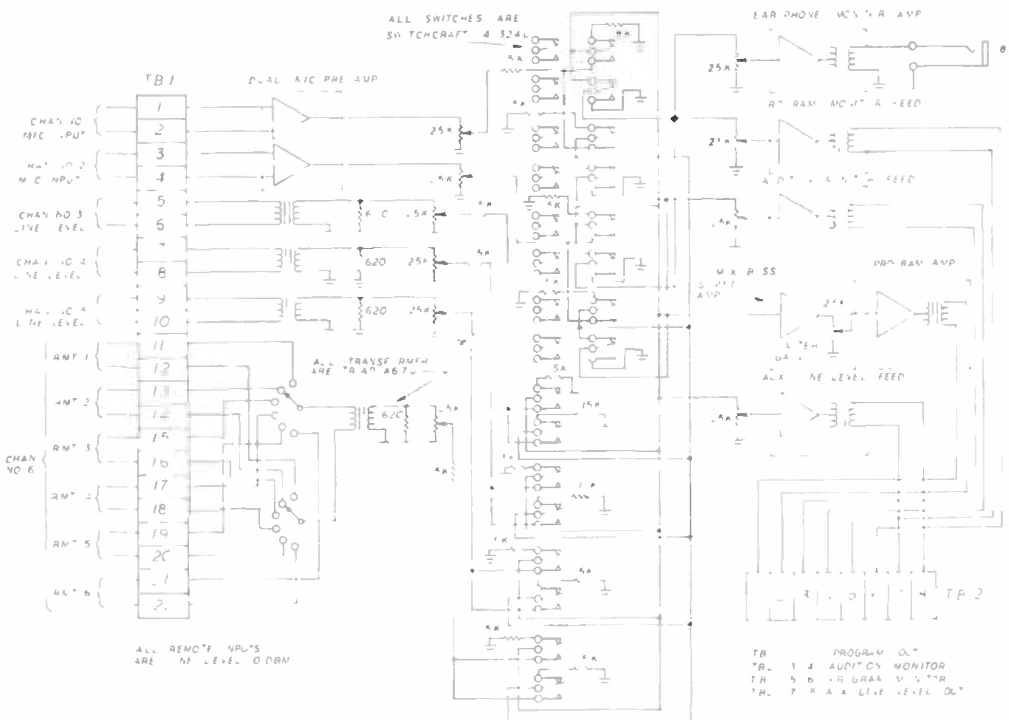


Figure 3. A block diagram of the console.

but then again, they aren't too bad, especially considering that this amplifier costs approximately $30.00 to build. If we assume an operating level of 0 dBm, this amplifier provides head-room of 22 dB before clipping, which should be adequate for broadcast use. Another desirable feature is that the board operates from a single-ended power supply of 24 volts d.c., thereby also keeping the power supply simple and inexpensive.

Having decided on using this as my "do-it-all" audio module, it was now necessary to start planning on putting this circuit into printed-circuit board form. I acquired the necessary equipment and began to tape-up the art work, using a 2:1 scale. After completing the art work, I took it over to one of our local photo shops and requested a negative reduced by 50 per cent the size of the art work. Now, having a negative of the correct size, it was a simple matter to have the printed circuit boards manufactured by a local business involved in that type of work.

Upon receiving my first run of finished boards, I immediately began stuffing parts and soldering, anxious to test the first completed module. Appropriate connections were made to a makeshift test socket, and once checked, the power switch was turned on. Since there was no sign of smoke, and none of the components felt unusually warm to the touch, further testing followed. A short time later I was pleasantly surprised to find that everything was working just as expected, meeting all of the technical specifications set forth in the design criteria.

The only remaining question would be the reliability factor. I decided to conduct a reliability test by running up a 1 kHz signal to full output capability (+ 22 dBm) and letting the circuit operate in this mode for the next 30 days, around-the-clock. One month later the board was still playing at full output, just as I had left it. Another set of measurements were made at this time, showing no deterioration in quality. Feeling somewhat encouraged by now, I began thinking about the only remaining circuitry to be developed, the mic pre-amp.

MIC PRE-AMP

The development of the mic re-amp seemed to go a little faster than did the previous project. Within two weeks, I was involved in testing my first printed circuit board of the mic pre-amp. (See FIGURE 2) The results were very gratifying inasmuch as it worked fine the first time out. The total cost of the mic pre-amp was approximately $40.00. Since the pre-amp is one of dual capability, only one card was required for my project. Having completed the assembly and testing of all of the boards, it was now time to move on to the next phase of the project, packaging.

Packaging is oftentimes, at least for me, a function of design. By this I mean I always try to package the components in such a way as to provide shortest possible wiring runs in the interconnecting of sub-assemblies. Sometimes it becomes difficult to achieve this goal because of the front panel layout. I found myself changing the design to effect a better wiring scheme, and then changing the wiring runs to better accommodate the design. Obviously, something had to be done to change this situation. I decided to adopt a final design and worry about how to locate the wire and cabling runs later.

We have now reached the point at which we can look at the console as a total package rather than several modules and sub-assemblies that must somehow fit together inside the console housing. In order to determine the size of the housing, it was necessary to know the exact dimensions of the front panel and the interior mounting plate on which the modules were to be mounted. The front panel was designed primarily for operator convenience, consisting of an aluminum back panel, one eighth of an inch thick, with a stainless steel cover panel which was engraved with lettering denoting the function of each of the controls. All of the metal work was done by a local precision sheet-metal shop for a total cost of $138.00.

The front panel dimensions were 24 inches wide and 8 inches high. Because I used a stainless steel cover panel, no screw mounting holes were visible anyplace on the front panel. The front panel was mounted to the console housing by using a piano hinge, with the panel mounting holes counter-sunk in order not to protrude and cause a problem when the cover panel was installed.

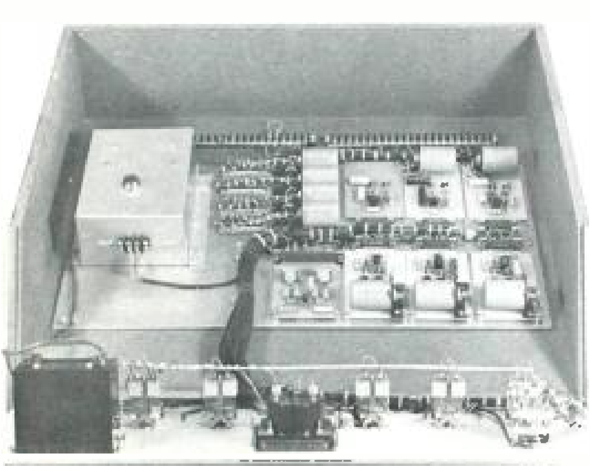
The console housing was built by a local cabinet shop, fabricated out of particle board covered with formica. The outside dimensions of the housing were 25 inches wide and 22 inches deep. In order to maintain a low profile appearance, the height of the housing was kept at 9 inches.

PACKAGING INTERIOR COMPONENTS

Packaging of the interior components was accomplished by using a plate of aluminum as a base-plate for mounting the printed circuit board edge connectors, transformer mounting bracket, solder terminal strips, power supply, and cable clamps. The base plate was one-eighth of an inch thick, 22 inches wide, and 12 inches deep. It was positioned inside the housing in such a way as to allow enough clearance for the front panel parts, such as switches, pots, meter, and clock to adequately clear any of the sub-assemblies mounted on the base-plate. It also provided enough clearance for the barrier blocks to be mounted to the base of the console housing behind the base-plate, affording easy access to input and output connections. A ground bus was fabricated out of number 10 copper wire, straightened, tinned, and formed to size by bending at the proper length, and held in place through the use of wood screws.

The actual wiring of the printed circuit edge connectors was very straightforward, and seemed to fall into place simply because of the mechanical layout used. Good grounding techniques were observed throughout as a precautionary measure. I was concerned with two potential problems at this point—one that the mic cables were tied in with high-level cables, a possible source of cross-talk, the other that the power supply was to be incorporated inside the console housing. To reduce the possibility of hum and noise from the power supply, it was built inside a small aluminum chassis, with the

Figure 4. An interior view of the completed console.



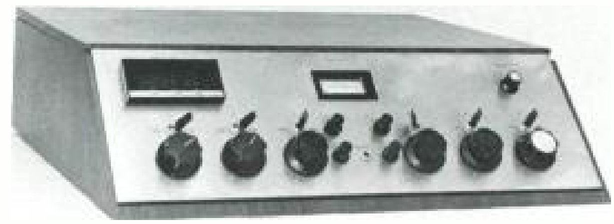


Figure 5. The console constructed by the author.

chassis providing the necessary shielding. Neither of these two situations developed into a problem as I found out later on during final testing.

During the early design stages, it was decided that no monitor amp would be incorporated in the console to further guard against cross-talk problems. All monitor output feeds are 0 dBm, sufficient for driving almost any studio monitor system. Mic muting was accomplished by simply grounding the inputs of the monitor driving amps through the front panel mic switch, thus eliminating the need for any relay muting.

Having completed the wiring, final tests were made. Upon completion of the final test, the results were as follows:

Audio frequency response
(referenced to 1 kHz) .... −0.5 dB
Distortion (thd) program/audition outputs ... <0.5%
Signal-to-Noise (referenced to +4 dBm) .. −66.0 dB
Output level (before clipping) .... +22 dBm
Output impedance (balanced) .... 600 ohms
Number of inputs: 2 Mic (150 ohm)
9 line level (600 ohm) .... 11 total
Number of outputs: Program, aux-line level,
program monitor, aud. monitor .... 4 total
Cross-talk: (referenced to 0 dBm, 1 kHz) .. −60 dB

The console described here was custom designed and built to fulfill a specific need. It should be obvious that larger consoles demanding additional input and output features could also be home built for a fraction of the cost of most commercially available models.

Flexibility is usually a function of switching capabilities provided by the manufacturer. One can easily see that adding the necessary switches to this basic design would provide all the features and compare favorably with the performance of commercial consoles costing considerably more.

The final cost analysis breaks down as follows:

Metal work	$138.00
Audio modules	180.00
Mic module	40.00
Power Supply	31.50
Clock Assembly	150.00
vu Meter	40.00
Switches	90.00
Wire and Hardware	20.00
Console Housing	40.00
TOTAL	$729.50

The total costs versus quality and operating performance has proved this to be a worth-while project. ■

LORING S. FISHER

Using Musical Format Services

Syndicated musical programs often provide broadcasting stations with quality at a savings.

IN ORDER to understand the growing popularity of musical format service, it is necessary to be aware of the needs that developed in the broadcast industry. At present, there are approximately 7,400 radio stations, a 50 percent increase over the number of stations operating ten years ago and including a three-fold increase in f.m. service alone; within the past five years, the number of f.m. stations has virtually doubled. This constitutes the most rapid growth for radio broadcasting since the adoption of stereo broadcasting techniques in the mid-1950's. But technological improvement has been but one of several contributing factors to the development and growth of f.m. broadcasting.

With the increasing number of stations coming into operation in such a short time, the needs of the industry have been strained, in an effort to provide the public with optimum use of the frequency spectrum. The necessity for non-duplication of programming in the larger markets imposed an additional operating burden on many stations attempting to meet new programming requirements. Lack of qualified manpower, coupled with the cost of broadening and expanding programming service, led to a proliferation of program automation concepts and techniques. Several companies offering automation services were formed and while some did not last, others still exist.

Loring S. Fisher is Director of Marketing and Operations at Bonneville Program Services, Tenafly, N.J.

SALES VS. PROGRAMMING

With the spate of stations attempting to offer diversified programming, while at the same time operating profitably, syndicated program packages became popular items. Another reason for the acceptance of packaged entertainment services grew out of the operational make-up of most stations. Typically, the station management personnel have sales backgrounds. The result, then, is a philosophical conflict with respect to programming. Classic examples of the sales *versus* programming syndrome exist in many radio stations throughout the country and have for many years. It is the broadcasting version of the chicken-and-the-egg paradox. Does programming attract an audience, which in turn allows a station to generate increasing sources of revenue through advertising, or, is sales revenue necessary first. in order to provide the means whereby service to the community can be expanded?

It appears that, too frequently, determination of program schedules or content is affected by sponsor likes or dislikes. At times, a block of programming time may be structured in a way to satisfy a sponsor without regard to any specific need or interest expressed by the station's community of license. On the other hand, it may certainly be argued that if the programming is of interest and satisfies the needs of even a small number of persons, then it is justifiahle.

Thus, we find that the scrutiny of audience specialization and programming fragmentation becomes yet another consideration. In competitive marketplaces, contrasted with smaller rural areas, a programming concept which neglects listener preference frequently results in an audience tune-out. To counter such a result, specialized programming. and more structured adherence to specific programming guidelines, has evolved. It is important to note that broadcasting competition depends on the proliferation of radio signals and many not bear a close correlation, in terms of geography or population density, to the area being served.

Thus, audience profiles start to evolve where some stations specialize in a teen or youth audience or young adults, some attempting to bridge the two, while others may seek other demographics, such as women-only or adults over 45. Some stations satisfy minority interests by block programming ethnic programs in a native tongue, or other entertainment or informational content of particular interest to specific groups.

Frequently, in a rural community, the listener may be subjected to any or all of these extremes while listening to a given radio station for extended periods of time. Some of this may be predicated on the source of revenue dollars rather than taking into consideration the economics of the number of people listening to the station at a given time and the impact that the choice of program may have on audience size.

Additional specialization is manifested by those stations dealing in the news only or all-talk programming structures of varying degrees. Most controversial of these recently has been the so-called "topless" format. (Ed. note: "topless" radio is a derogatory reference to a format catering to the male "macho" listener. Fortunately, it has not enjoyed a spectacular success.) Certainly this illustrates a case in point where the broadcaster's discretion and sensitivity influence his role in the shaping and structure of changing social attitudes and opinions.

COMMERCIAL PRACTICES

As a parallel to programming activity, it is also important to note that commercial practices of stations vary widely. Not too many years ago, back in the late 1950's or early 1960's, many listeners thought that "f.m." stood for "free music"! They were most offended when commercials started to appear on what they regarded as a sanctuary from the obtrusiveness and objectionable nature of the commercials on a.m. radio and television stations (particularly at the height of the so-called loudness controversy). F.m. stations responded in some measure by attempting to suppress or tailor any commercial material which might seem to have an objectionable profile.

With diminishing viewer interest in television fueling a renaissance of radio broadcasting, more commercial dollars found their way onto the radio portion of the spectrum and a.m. broadcasters, especially, discovered that programming to specific audience demographics was beginning to reap handsome financial returns. F.m. stations offered a continuing diet of mostly music on an uninterrupted basis and some listeners who enjoy this type of program started to gravitate away from a.m. However, after several years of this, we find that f.m. programming has become as varied (and in some cases as blatant and objectionable) as its a.m. counterpart. Even so, many f.m. stations have continued to meet a reluctance on the part of the advertiser to spend money in conjunction with f.m. programming; the number of commercials on these stations is significantly lower, on average, than those of a.m. stations, a circumstance which leads many listeners to prefer f.m. In fact, the application of the fundamental laws regarding free enterprise and competition has caused a.m. broadcasters to reduce the number of commercials in order to deter their listeners from fleeing to f.m. At the same time, f.m. broadcasters are simultaneously realizing that they too are *radio*, rather than a "separate" or "special" kind of communications medium. So, those factors that did represent a difference to the discerning listener are now becoming minimized.

PERSONNEL PROBLEMS

In addition to facing the reality of increasing competition from other broadcasting stations, many stations are faced with a shortage of trained saff personnel. There is now much information before the FCC relating to operator requirements and there appears to be a general reduction in the standards required for an operator on duty. Part of this trend is due to the proven reliability of new technological developments and part of it has been the result of the increasing need to relieve the broadcaster of operating burdens which might affect his ability to operate profitably.

CHANGES IN THE WIND

With this background, we find that some of the original concepts and precepts upon which broadcasting was founded are gradually and constantly changing. Technological innovation and pioneering are no longer the mainstay of broadcast and communication. Today, the public is more demanding in terms of programming content. However, given the lack of trained programming people, and the economics of most radio stations, this is the area most often lacking in development. It seems to be a truism that everyone has his own ideas as to how to program. Some are more successful than others but all must provide enough of a listening base to merit and sustain continued financial operation. In some cases, programming decisions evolve on a day-by-day, minute-by-minute basis, changing as specific events dictate. Other programming formats are more structured, more tightly specified and controlled. But these, too, attempt to maintain a balance, providing relevance to instantaneous needs to satisfy certain regulatory stipulations as well as an underlying premise that spontaneity of communication is the cornerstone upon which radio broadcasting is built.

Prompted by the trend toward reduced cost and minimal staff requirements, as well as an attempt to realize the benefits of several stations utilizing similar or virtually identical programming prepared from a common programming source, the acceptance of the distribution of syndicated material has made this a much more refined communication technique than simply a means of lowering operating costs! In a number of markets around the country, several stations are enjoying dominant and significant shares of the audience, utilizing material which is virtually identical, on an hour-by-hour basis, to material being used by stations in other geographical markets. Much of the success is the result of substantial effort on the part of program suppliers to produce a superior product with optimum utilization of the proper operating concepts and philosophies to fulfill the needs of the listeners to whom their particular product is aimed. Depending on the nature of station operations in various markets, such efforts may produce marginal audiences or extremely significant ones.

USING RIGHT CONTROL

It is interesting to note that key elements in the success of virtually any broadcast operation, or for that matter any organizational structure, depend on control and discipline. Knowing what is to be done, how to do it, and in turn executing what is deemed necessary, becomes the set of rules by which the game is played.

Thus, some suppliers of musical program material have particular expertise, developed through experience in many markets. This makes their advice on programming matters as vital to success as that of the lawyer or engineer in his own area of knowledge. It is on the basis of this experience that certain recommendations may be made to a station to enable it to obtain maximum benefit from the particular entertainment format chosen. In this respect, the supplier becomes a programming consultant which is, in fact, the proper characterization of this aspect of the service.

Where an agreement is made between a broadcast station licensee and a programming service or consultancy, the licensee must certainly be aware that he must retain ultimate control over the programming material. No agreement can be based on a relinquishment of that obligation.

However, where there is a desire on the part of the licensee to obtain a specific result deemed to be in the public interest for the community, and it has been found that such a result is achieved by a particular musical programming format company, it is not unreasonable for the station to follow these proven methods in obtaining the desired objective, with the music programming format company occupying a consultant's role and providing advice with respect to the total program structure of the station. As a particular example, stations using the Bonneville Program Service agree to the voluntary limitation of the number of commercials-per-hour substantially below that of other stations. Surely this policy may be said to be in the public interest, if the public is tired of hearing heavy commercial content on other radio stations.

TALK REGULATIONS

Another basic issue is the amount of talk which is recommended. Many licensees have a programming commitment in which ten or fifteen percent of the total operating time is represented by programs in the categories of news, public affairs, and "other." But it would seem that a qualitative, rather than quantitative, measure should also enter into the evaluation of how a station is fulfilling its programming requirements; quantity alone cannot be the criterion. It should also be pointed out that many stations make no excuse about the fact that they "bury" much of their so-called talk commitment at times that would be deemed relatively harmless to the station performance in attracting a large audience. The stereotype block of religious or quasi-religious program material on Sunday mornings, coupled with blocks of talk typically heard on Sunday evening, or during the wee hours of the morning, are representative of the manner in which this programming commitment is satisfied by many licensees. Surely it would better serve the public interest if relevant and meaningful material were presented for audience consumption even at those times.

Gone are the days of network programs and other productions running half an hour in length and longer. The impact of television programming has made a marked effect on the style and content of programs of interest to the radio listener. Part of this has been further fueled by the programming structure of competitive radio stations. In general, most listeners will not tolerate lengthy interruptions of program material normally of an entertaining nature. Stations whose primary program structure is abundantly one of talk provide a complementary program selection for those people whose interest may be attracted in that fashion.

CONSULTANT-BROADCASTER AGREEMENTS

It should be noted that if a broadcast licensee undertakes to engage the services of a programming consultant, that consultant should be capable of structuring program recommendations in a manner that will produce the desired end results by the client broadcast station. Certainly, it would behoove the consultant to make—and the station management/ownership to accept—only such recommendations as are within the structure and guidelines of FCC rules and regulations. If the recommendations of the consultant do not produce the desired results sought by the client station, then it would be a foregone conclusion that any working agreement would and should be terminated.

If the consultant is to protect his interests, a definitive set of working standards should be established for the mutual benefit of all parties involved in the agreement. The fact that some consultants may have any number of agreements in effect should not be a matter of great concern. In all likelihood the nature of the recommendations or operating constraints would differ as a function of the particular consultancy.

For example, a music program service performs a necessary vital public interest function where there is a desire for diversification of entertainment, allowing a licensee to provide high quality musical offerings in situations where they might not otherwise be feasible, due to cost, personnel availability, and technical considerations.

Thus, when a decision is made by a licensee to engage a consultant, he does so in the interest of providing the highest quality service possible to his listening audience. It is axiomatic that where not only the public is benefited, but the station as well, by the economic stability which may follow, the legitimate interests of the FCC are satisfied. If the licensee fails to meet the obligations of ascertaining and programming for the needs of its community, the Commission has a legal obligation to intervene. However, there is no evidence that by employing a music program format service there is any impairment of a licensee's ability to serve its community. On the contrary, the licensee is better able to provide high quality programming designed to deal in a significant, meaningful way with the needs and interest of its community. ■

JOHN BORWICK

The APRS Exhibition

The London exhibition is a showcase for new European equipment, as well as some notable American products.

FOR THE ELEVENTH YEAR running, the Association of Professional Recording Studios put on a glittering exhibition of Professional Recording Equipment at the Connaught Rooms, London on June 21-23. The APRS is a unique trade association in that the bulk of its members are British recording studios (a total of 110 members) although it more recently added a category for manufacturers (56 members) and educational establishments (6).

So the APRS Exhibition, resembling those we are accustomed to see at AES Conventions, displays only professional sound recording items—with no domestic hi-fi. This year there were 80 exhibitors, representing over 100 brand names and I can't think off-hand of any important brand that was missing. All the top American manufacturers have active European representation and, if I seem to give less space to them, it is because I am assuming that their newest products have already been launched at U.S. meetings. Some of the European, and especially British, manufacturers do, however, make a habit of unveiling new designs at the annual APRS Show.

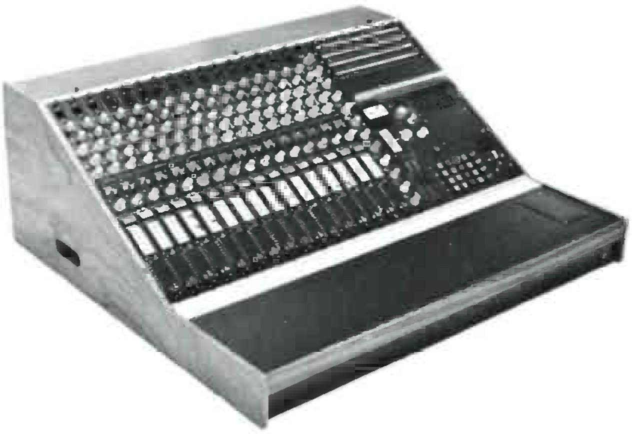
TAPE MACHINES

Speaking to record producers lately, I have begun to wonder if there aren't signs of the pendulum swinging back from multi-multitrack recording to simpler, but possibly more sonically pure, recording techniques. But you wouldn't think so if you walked around this exhibition. Studer made a big feature of their latest A800 recorder in its 24-track version, the trick being that two of these monsters were linked together as master and slave through the Tape Lock 2000 system. This effectively synchronized the machines to give 46-track working.

For producers able to make it on only 32-track, Telefunken let me play about with their Magnetophon 15A in a 32-track version. Though using only 2-inch wide tape, the specification still made pretty good reading, except that dynamic range was 2 dB less than for 24-track, and crosstalk was down to 45 dB, compared with 50 dB and 58 dB on the 24-track and 16-track respectively. All 32 amplifiers were housed in drawers in the console, with the v.u. meters and record/sync/play buttons arrayed on a sloping front panel; transport was very smooth. Telefunken, of course, is the originator of the Telcom C4 noise reduction system, which seems to combine the virtues of Dolby's four-band splitting with dbx's across-the-board level companding. If you fasten 32 Telcom units to the Magnetophon 15A, the signal-to-noise and crosstalk figures I just quoted for 32-track zoom up to 90 dB and 66 dB—which looks pretty respectable. This show was also chosen by Telefunken for a first showing of their new compact ¼-inch recorder, the Magnetophon 12A, and the Tachos cassette loading machine.

Lyrec of Denmark has carried cueing and remote control of its multitrack recorders to extreme lengths, with a microprocessor giving the possibility of searching to three different tape positions, recycling between any two and

General view of just one of the six handsome Connaught Rooms housing the APRS Exhibition.



Libra Theatre Sound Mixer.

The Calrec Microphone (new styling).

AUDIX portable MXT-1000 audio mixer.

The Alice 6006E disc jockey mixer is around $3000.

16-track Studer A800 with sophisticated microprocessor control unit.

Klark-Teknik DN36 Analogue Time Processor.

storing up to sixteen. There is spot erase, varispeed and solo button—very rare on tape recorders—enabling any soloed track to be heard in its correct stereo location. Also new from Lyrec was a high-speed cassette duplicator with vertical loop-bin and slaves in pairs, giving useful space economy. As always, the Swiss Nagra IV-S, now with 10½-inch NAB spool adaptor, was attracting the attention of everyone needing full professional performance from a battery/mains portable, and the baby Nagra SN weighs only 1.3 pounds and so is just right for recording by parachute or on a mountain climb.

Ampex occupied a large area, giving them space to show the MM-1200 in a 24-track version, and the ATR-100 2-track linked to their MQS-100 video/audio synchronizer and VPR-1 helical-scan video recorder (reproducing the "Muppet Show"). They also featured the lower-cost ATR-700 recorder, designed for the market now catered to so prominently by Studer's little brother brand.

A chandelier's-eye view of part. of the Studer display.

A young visitor tries his hand at mixing down a multi-track tape on the Trident stand.

Revox. MCI is moving into Europe in a big way and. besides a huge JH114 24-track recorder. they introduced a neat JH110A 8-track version using 1-inch tape and JH110M broadcast recorder. specially for European radio stations. 3M too gave an impressive demonstration of the M79 24-track with Audio Kinetics XT24 Intelocator.

MIXING CONSOLES

Alongside the MCI recorders was a very large example of their automated JH-500 series mixing desk. As is usual with these custom-built jobs. the manufacturers do not make up standard consoles for exhibitions but prefer to show a real live sample on its way to a customer. This multi-function model was destined for the new studios of Red Bus Records. Cadac is also into computer-assisted mixdowns. with their CARE (Cadac Automated Remix Equipment) system. but the real novelty on their stand was the new "In-Line" console philosophy. This brings all the input, output and monitoring facilities for each channel together on one, necessarily rather long. module. A central routing control module assigns the modules. gives numeric l.e.d. display of the route selected and has a RAM (Random Access Memory) to recall. examine or alter the routing at any time.

Another British company basing their channel strips on a single long module was new to me, Solid State Logic Limited of Oxford. They modestly announced themselves as designing "the most technologically sophisticated, flexible and engineer-oriented production recording consoles ever built." Their console certainly looked very impressive. with a compressor/limiter/expander/noise gate on *every* channel. 14-control parametric equalizer. and overload indicator on *every* channel, 8-vca subgroups. pre-timed automatic fade from 1-60 seconds. etc. As an optional extra. a floppy computer system has an alphanumeric keyboard for communicating cues and other data in English. with a 24-line t.v. display unit showing print-outs of track directories or graphics. My eye was originally caught by a full-size t.v. monitor screen fed in parallel with the desk-mounted one. displaying song title, take numbers. track allocations etc., as a 24-track tape was being remixed. Certainly this is a company name to watch.

Neve Electronics is a name already well known. Of course their NECAM computer-assisted mixing system. with its motor-driven faders following the memorized movements on replay. is a natural visitor draw at exhibitions. But they specially featured the latest Model 8078 Multitrack Recording Console. This 40-channel 32-track desk has a separate 32-track monitor/mixdown section in recognition of the commonly met situation of a console doing double duty for laying down tracks and then meeting the different demands of the mixdown session. There was also a new. smaller Model 8066 20-channel 16-group 16-track console and—something that appealed to me for my occasional classical music location assignments with just a few microphones—the Model 5422 compact 8 x 2 suitcase console.

Helios Electronics is yet another British company with a worldwide reputation for sophisticated consoles and recording vehicles. Their specialty is custom-built ergonomic design ("human engineering") and their main console on show had the typical compact central track monitor panel with side-wings to keep all necessary controls within arm's reach. They also introduced a Nordic range of broadcast

Derek Tilsley of Neve (right) shows off a comprehensive example of their NECAM, computer-assisted mixing desk.

consoles, arising from their years of supplying desks to international broadcast standards. Another promising newcomer was Libra Electronics who, by concentrating on the needs of theatre sound installations, had developed an attractive console with flexible routing of any input to any configuration of outputs (loudspeakers) and standby modules for presetting several configurations for instant cueing.

AUXILIARY UNITS

Add-on boxes for response shaping, time shifting and level companding can now be had with such versatile possibilities and advanced illuminated displays that they rival music synthesizers and real-time analyzers in their complexity. Audio Developments, for example, showed the ADO70 proGraphic Equalizer which has gone back to the old idea of stepped attenuators—controlled digitally via modern semiconductor analogue switches. A 16-frequency graphic equalizer in approximately ½-octave bands has each band controlled in 2 dB steps from +14 to —14 dB with the settings indicated on a column of l.e.d.'s A control fader generates a binary code and a keyboard sets up all functions, including access to the individual frequency modules. A memory device will store up to 16 response curves for future instantaneous use, and allow curves to be displayed within milliseconds of each other for immediate comparison. Remote control and display are possible, allowing the shared use of a proGraphic Equalizer or a central bank of them.

Klark-Technik, a British firm also famous for graphic equalizers, showed two analogue time processors and a neat DN70 Digital Time Processor with digital readout in milliseconds of delay for each of its three outputs—with optional readout in feet or metres for sound reinforcement applications. Audio & Design had added the E950 Paragraphic Equalizer to their range. This has a 6-section stereo or 12-band mono format with tunable frequency over 4 octaves in each section, and variable bandwidth from 6 octaves to one-eighth octave. So up to 6 feedback nodes could be tight-notched in stereo and accurately tuned to remove a minimum of audio content.

MICROPHONES

After a period of non-change, microphones seem once again on the march. The Calrec 4-dimensional CM4050 soundfield microphone (already described in **db**—see July issue) was there in a newly designed casing with an improved control unit. Since the complete assembly costs around £2,100, I was told the company is planning a hire scheme for smaller recording teams who want the special flexibility of this 4-capsule microphone for a limited schedule of recordings. Neumann had a new gun microphone for high directivity and a whole range of thin capsule extension tubes for their KM83, 84 and 85 microphones. AKG pioneered the extension tube idea in their CMS series and at this show they featured a new thinner two-way "woofer plus tweeter" dynamic microphone, the D222. Both Sennheiser and Beyer have infrared cordless headphones with associated infrared light transmitters. They reported some penetration into studios where foldback to performers' headphones can produce a maze of tangled cables, doing it the old-fashioned way with wires instead of "wireless." Truly we are in a restless industry, as each year's APRS Exhibition has demonstrated. I can hardly wait for APRS 1979. ■

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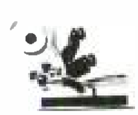

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THE RESONATOR is more than a reverb. Designed for use with any console, including Tascam. $359.00. **Dyma, Box 1697, Taos, N.M. 87571.**

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FOR SALE: Westrex 3DIIH, $3,495.00; Haeco SC-2, $4,800.00; Haeco SC-1, $1,495.00; Grampian D, $385.00; Grampian BI/D, $325.00; Westrex 2B, $525.00; all Haeco cutterheads new, other cutterheads reconditioned and in specs. **International Cutterhead Repair, 194 Kings Ct., Teaneck, N.J. 07666. (201) 837-1289.**

INTERFACE 200 series, 8 x 2 board; Ducan sliders; brand new; list price, $2,000, yours for $1,250. Hewlett-Packard/Altec 8050A real time analyzer, $2,000. Demo UREI equipment; 1176-LN limiter; 527 equalizer; other assorted pieces. Make offer. **Ro-Cel Electronics, Inc., 731 Butler St., Pittsburgh, Pa. 15223. (412) 781-2327, Don.**

MODERN RECORDING TECHNIQUE, by Robert E. Runstein. The only book covering all aspects of multi-track pop music recording from microphones through disc cutting. For engineers, producers, and musicians. $10.50 prepaid. **Robert E. Runstein, 1105 Massachusetts Ave., #4E, Cambridge, Mass. 02138.**

IVIE DEMO MODELS. While they last. IE 10A, $500; IE 20A, $150; full factory warranty. $2 postage & handling, certified check or money order. **Theatre Technology, 37 W. 20th St., New York, N.Y. 10011.**

HAECO CSG-2 compatible stereo generator, new condition, $1,700. Used Westrex 1574D cutter amps, $400 each. Pultec MH-4 mixer, $125. HAECO LX-1 crossover, $600. Fairchild 670 stereo limiter, $650. **Frankford/Wayne Recording, 134 N. 12th St., Philadelphia, Pa. 19107. (215) 561-1794.**

RANSTEELE custom heavy duty wall/ceiling speaker mounting brackets for JBL, Altec, Big Reds, etc., $160 pair. Ransteele 200W/ch stereo cutter drive amplifiers; used Westrex and Haeco cutterheads; complete line of custom disc mastering room audio equipment not available anywhere else! Reconditioned Scully lathes; complete turnkey mastering rooms. **Ransteele Audio, Inc. 1697 Broadway, New York, N.Y. 10019. (212) 265-5563.**

PROFESSIONAL AUDIO COMPONENTS: AKG mics; Badap 1; Crown; dbx; Delta-Lab; Eventide; Frazier; Gauss; GLI; Ivie; Malatchi; MasterRoom ; Neumann mics; Otari; Pentagon; RTR; Sennheiser mics; Switchcraft; Tascam; Uni-Sync; and UREI. These products are on demo in our showroom and in stock for immediate delivery. Our shipping is insured and prepaid. **Barclay Recording & Electronics, 233 E. Lancaster Ave., Wynnewood, Pa. 19096. (215) 667-3048 or 649-2965.**

FOR SALE, large Moog synthesizer. All components, 2 keyboards, ribbon controller, 2 sequential controllers. Excellent condition. **Scott-Textor Productions, 220 E. 54th St., New York, N.Y. 10022. (212) 688-4330.**

SCULLY 284 12-track recorder, excellent condition, 15-30 i.p.s., $6,400. Orban Parasound parametric equalizer, brand new, $500.00. **Sound Trap Studio. (201) 895-4077.**

TAPCO and Electro-Voice: mixers, equalizers, amps, mics, and raw loudspeakers. Write or call for low mail order prices. **Sonix Co., P.O. Box 58, Indian Head, Md. 20640. (301) 753-6432.**

SOUND COMPANY for sale: BGW 750's: 8 Cerwin Vega cabinets; TAPCO 6201B's; Anvil cases; everything needed for medium shows; also Mitchell light systems. Tulsa, (918) 492-0161, after 6.

USED RECORDING equipment for sale. Professional mics, speakers, miscellaneous gear. **Dan (415) 232-7933.**

PENTAGON C-1000 2-track reel-to-cassette 8X duplicator with S-1000 slave (12 cassettes/run); Pentagon RS-2150-8. 0.150 mil reel slave. **(212) 924-6663.**

SCULLY 8-TRACK w/Sync Master, mint condition; Accurate 8 x 8 mixing console; best offer over $6,000.00 takes both. **Le Mans Sound. (201) 359-5520.**

CUTTERHEAD REPAIR SERVICE for all models Westrex, HAECO, Grampian. Modifications done on Westrex. Avoid costly down time; 3-day turnaround upon receipt. Send for free brochure: **International Cutterhead Repair, 194 Kings Ct., Teaneck, N.J. 07666. (201) 837-1289.**

TASCAM Model 10 mixer; 12 inputs; headphone monitor; one meter out, but otherwise perfect. Sacrifice, $995.00. **(318) 234-3911.**

IVIE SOUND ANALYZERS, all models in stock. **Theatre Technology, 37 W. 20th St., New York City 10011. (212) 929-5380.**

MCI JH10 DECK with auto-locator (factory-reconditioned), $2,000.00; Tascam Series 70 8-track recorder with new heads (perfect), $1,995.00; 32 Altec light meters with power supply, $1,500.00; Sound Workshop 223A electronic crossover, $250.00; Sound Workshop 882M mixer, Model 18 monitor mixer, patch bay, with console cabinet, $1,695.00. **Paul Westbrook Audio, (512) 578-4401, 578-8072.**

TASCAM Model 10, 12-in/4-out, accessories; Sony TC8-54 4-channel; associated equipment. **Steve Kurtz, 2421 Mapleview, Kalamazoo, Mi. 49002. (616) 323-9410.**

SOUND WORKSHOP professional products from stock! **Paul Westbrook Audio. (512) 578-4401, 578-8072.**

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24-TRACK DATAMIX consoles in excellent condition. Contact **H. Selby. (212) 477-7500.**

PROFESSIONAL AUDIO DEALER. Over sixty product lines. **Sound System Labs, P.O. Box 3100. Camarillo, Ca. 93010. (805) 482-5081.**

ONE SPECTRASONIC 24 x 24 custom console with two producer's desks. Additional patch bay, etc. Must sell, best offer. **(215) 271-7333.**

SPECTRA SONICS 1024-24 recording console; 24-in/out. Producer's desk. 18 months old, excellent condition, $25,000. **(805) 966-6630, 966-1271.**

IMPRESS your manager. Save money on: Technics, Ampex, Scully, Nortronics, Stanton, Inovonics, UREI. Electro-Voice, Belar, Orban, etc. **Val-tronics, Inc. Call collect, (717) 655-5937.**

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WEST COAST engineers/producers: Bay Area sixteen-track; fully equipped with tube Neumanns, Schoeps, etc., EMT; DDL's (2); acoustic chamber; Dolby A; Spectrosonics; Scully; Ampex. Free flight for out-of-towners. **Tewksbury. (415) 232-7933.**

MOBILE HOME recording studios! Complete set of plans, $1,250.00, or let us sell you turn-key with or without equipment. **Paul Westbrook Audio, (512) 578-4401, 578-8072.**

WANTED

WANTED: Recording equipment of all ages and variety; Neumann mics, EMT, etc. **Dan Alexander, 6026 Bernhard, Richmond, Ca. 94805. (415) 232-7933.**

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SOUND REINFORCEMENT SYSTEM CONSULTANT and designer required by acoustical consulting firm. College degree necessary, plus knowledge of sound system equipment and design techniques for large permanent installations. Some travel required.

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MAJOR New York City recording studio seeking top notch creative maintenance engineer. Excellent opportunity. All inquiries confidential. **C/O Studio, P.O. Box 275, Old Chelsea Station, New York, N.Y. 10011.**

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Major N.Y.C. studio. New automated 24-track. Send resume to **Dept. 83, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

DISC MASTERING ENGINEERS. The East Coast's largest independent disc mastering house is in need of two experienced cutters, preferably with a good technical background and customer following. Excellent salary and benefits. Apply with resume to: **Dept. 82, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

ENGINEER/MANAGER. A San Francisco music and commercial production studio seeks chief engineer/manager. Must be able and willing to sell ad agencies, etc. Studio is comfortable, solid, 16-track facility. Salary commensurate with qualifications. Replies held in confidence. Please send qualifications to **db Magazine, Dept. 91, 1120 Old Country Rd., Plainview, N.Y. 11803.**

MAINTENANCE/CUTTING ENGINEER. Large independent East Coast disc mastering facility seeks an experienced maintenance/chief engineer who is looking for a bright future and who is completely familiar and is able to maintain Neumann, Scully, Westrex, and Capps cutting room equipment. Other duties will include occasional cutting, R&D projects, and construction. Applicant must be able to work with little or no supervision and be of high calibre. Excellent company benefit programs. Salary commensurate with qualifications. Reply with resume to **Dept. 81, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

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db People/Places/Happenings

• Recognition from the **Audio Engineering Society,** in the form of an honorary membership, has been awarded **John M. Eargle,** vice president, product development, for **James B. Lansing Sound.** Mr. Eargle has not only been prominent in the audio industry, but has held a number of positions of responsibility with the AES. He is one of 50 people who have received this signal honor.

• Doubling of its Memphis, Tenn. facility has been announced by **Auditronics, Inc.** The new centralized location, at 3750 Old Getwell Rd. is designed to pull together many functions which had been housed in scattered quarters.

• U.S. contacts with British **IMF Electronics** are now possible through the **Mark Granby Company.** They're at 5606 Ostrom Ave., Encino, Ca. 91316.

• The promotion of **Karl Kramer** to the newly created post of vice president, manufacturing, at **Midwest Electronic Industries, Inc.** of Chicago, Ill. has been announced. Mr. Kramer has been the manager of manufacturing for six years.

• **Robert Kimball** has joined **Sanyo Electric, Inc.** of Compton, Ca. as national sales manager, television. Mr. Kimball had been with **Sony** and prior to that, with **RCA.**

• An independent recording production company, **Glen Kolotkin Productions,** has been formed by **Glen Kolotkin.** Mr. Kolotkin has an impressive history, having recorded many of the most prominent entertainers in the business. The address of the new facility is 24 Elda Dr., San Rafael, Ca. 94903. Phons: (415) 472-0345.

. . . in reverent quest of the last dB

John Joseph Boyle

December 20, 1947-August 9, 1978

For those of us who were lucky enough to have known John Boyle, his passing from this physical world is a time of tremendous sadness, yet it is also a time of great strength. John was able to see the future as it related to the music, the equipment and most importantly, the people involved in pro audio. It was his great foresight and energy that inspired so many to follow his visions and his ideas. His accomplishments, through his involvements with Altec, Teac/Tascam, Express Sound, Sound Workshop, and so many others are numerous, but they are far surpassed by his inspiration to those he touched. In a competitive industry such as ours, John was always able to convey the importance of the music and the people. The gear was John's vehicle to what he cherished most; bringing friends and music together. We will miss John dearly, but he will live on in everything we do.

From John's friends in the industry who loved him dearly.

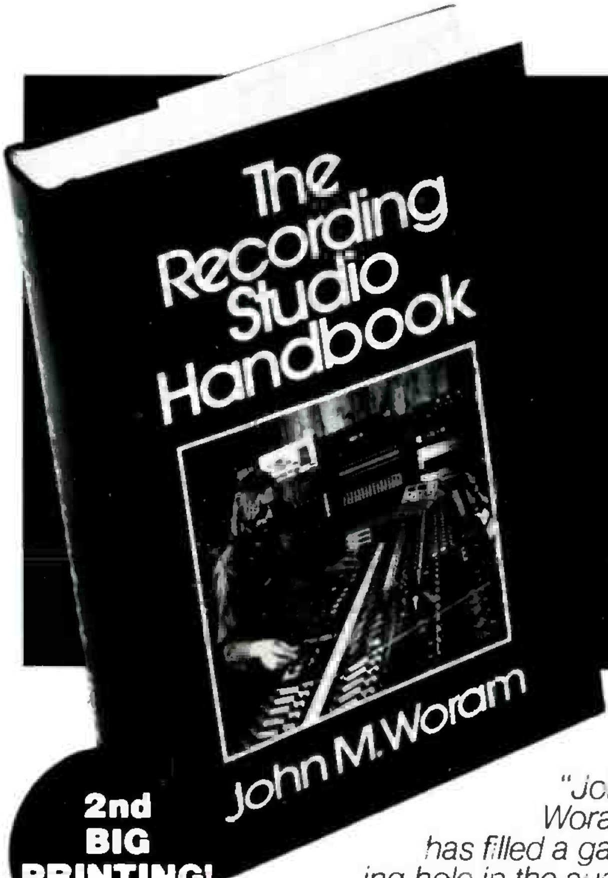
• **Stanford University,** Stanford, Ca. plans to build a prototype all-digital multitrack recording system that will incorporate studio and location recording, overdubbing, editing, mixing, equalization, limiting-compressing-expanding, ADT and AMT, reverberation, delay, localization, pitch change, etc. all in real-time with all functions full automated. The multitrack capability will range from 30 to 150 tracks. Sponsoring the endeavor is the Center for Computer Research in Music and Acoustics at the University.

• Cited for community involvement, such as donating musical instruments to the handicapped and designing Christmas lights for a Veterans' Hospital, as well as for business success, **Michael B. Matthews,** president of **Electro Harmonix Co.** of New York City, was named the New York State Small Business Person of the year. The award was sponsored by the national **Small Business Administration** (SBA).

• Several additions in personnel have been made at **McMartin Industries,** of Omaha, Nebraska. **Robert J. Schneider,** based in Albert Lea, Minn., has been appointed district sales manager for the upper Midwest; **Robert Beattie** has joined the firm as a district sales manager for the South Central region based at Birmingham, Ala. International sales is in the hands of **Charles B. Patterson. Eric Somers** has been appointed to the post of advertising manager.

• The appointment of **Charles Toda** as assistant general manager, sales planning, consumer electronics group, has been announced at **Panasonic,** of Secaucus, N.J. Mr. Toda began his career with the company in 1964, working for the parent company, **Matsushita Electric Industrial Co.**

• The **NRBA** has gone on record endorsing the concept of the deregulation of radio, one of the points covered by the proposed new Communications Act (HR 13015). While not in accord with all the proopsals contained in the bill, such as the spectrum fee and limiting to five the number of facilities which can be owned by any one entity, the NRBA approves the modernization of communications legislation, heretofore governed by the Communications Act of 1934.



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- **Flanging and Phasing**
- **Tape and Tape Recorder Fundamentals**
- **Magnetic Recording Tape**
- **The Tape Recorder**
- **Tape Recorder Alignment**
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- **Studio Noise Reduction Systems**
- **The Modern Recording Studio Console**
- **The Recording Session**
- **The Mixdown Session**

This hard cover text has been selected by several universities for their audio training programs. With 496 pages and hundreds of illustrations, photographs and drawings, it is an absolutely indispensable tool for anyone interested in the current state of the recording art.

Use the coupon at the right to order your copy of *The Recording Studio Handbook*. The price is only $35.00, and there's a 15-day money-back guarantee.

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