THE SOUND ENGINEERING MAGAZINE

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THE SOUND ENGINEERING MAGAZINE
JUNE 1981 VOLUME 15, NUMBER 6

ABOUT THE COVER

• The PZM (Pressure Zone Microphone) has quickly evolved from Ken Wahrenbrock's hand-built prototypes into an expanding series of production models. Our thanks to Crown International for this month's cover, showing yet another view of the PZM.

db is listed in **Current Contents: Engineering and Technology**

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db Letters

WOOFERS VERSUS HONKERS

To The Editor:

I am in full agreement with Michael Rettinger's premise* that we need better low bass in control rooms. However, there has been a revolution in loudspeaker theory since 1961 which Rettinger seems to have missed. Most of the good papers from Thiele, Small, and Benson in Sydney, Australia, have been published in the Journal of the Audio Engineering Society during the past decade. Raymond Cooke (of KEF loudspeaker fame) has edited these into a *Loudspeaker Anthology* which is available from the Audio Engineering Society. This Anthology is must reading for the speaker designer, although some of the most important papers might be a bit mathematically mature for most speaker users.

The first studio monitor speaker to use the Australian loudspeaker theory is the Electro-Voice Sentry III designed by Ray Newman. It still goes way below its California competition. A recently designed studio monitor to use this theory as well as the latest in Colorado Crossover Theory is the Harrison Systems monitor designed by John Hoge. This speaker will produce the low A at 27.5 Hz and has the sweetest sound through the voice range of any studio monitor I have heard. At a lower price tag, the Koss CM 1030 has a lower 3 dB frequency of 30 Hz and will nearly equal the Harrison Systems monitor *except* that the acoustic power output for tolerable distortion is about 15 dB lower.

All these speakers use correctly aligned vented box woofers rather than the horns advocated by Rettinger. In an AES preprint, Don Keele (now with JBL) compared vented boxes with horns for woofers and concluded that for frequency response and efficiency, vented boxes win. ("An Efficiency Constant Comparison Between Low-frequency Horns and Direct Radiators." AES Preprint 1127—Ed.) Professor Marshal Leach of Georgia Tech has applied the concepts of electro-acoustic woofer specification to horn woofer design. If one is going to build a horn woofer for the control room, the best starting point would be to scale up the folded corner horn originally designed by Paul Klipsch in 1941. It is still the best horn woofer on the market. My belief is that the acoustic power needed in a control room

Index of Advertisers

Coming Next Month

• In July, our three-part series on The Signal Path continues, as we look in on the latest in Signal Processing. In other words, anything that may happen between Microphones (June) and Monitor Systems (August).

does not require a horn woofer. With the availability of Australian loudspeaker theory for doing vented box woofers, the real problems for state-of-the-art studio monitors are back in the mid range—where most of the music is.

J. Robert Ashley
Professor, Electrical and
Computer Engineering
University of Colo.

*Michael Rettinger, "Reproducing Electronic Music in the Control Room," **db,** March 1981.

Mr. Rettinger Replies

I should like to say that, given a sufficient number of vented box woofers, sufficient bass can be provided in a conventional control without noticeable distortion. For large enclosures, however, the acoustic transformer represented by a horn appears to be superior. Such horns become a "must" for motion picture theatres equipped with Sensurround reproduction.

To The Editor:

I noticed some discussion about the use of PZMs for recording (October Letters to Editor) and I, too, share in the wonderment of the lack of experimentation and reporting of the use of PZMs. Without the kind of feedback that is present in magazines such as **db,** we are all losing valuable cross-information.

I have used PZMs on two projects and find them to be the perfect answer in some situations. During the recent 1980 Chicago Jazz Festival, which was carried live in stereo over NPR's satellite, we used PZMs on the string bass (clipped to the f-hole) and taped to the lid over the high and low end of the 9-ft. Steinway. Since each night would feature four to five groups and artists, and with only 10 minutes for set change (remember we're live), we needed to cut the "on-stage labor" down to a minimum. The PZM proved to be small, unobtrusive and easy to change without a sacrifice in quality for the participating stations.

During a recent taping of a B. J. Thomas television special, I experimented with PZMs on the piano due to the high level of sound leaking into the piano, which was to play a very prominent role according to the charts. Because the PZMs can be placed directly over the strings with the lid closed, we substantially cut down the leakage. However, in the final few minutes before the show, I decided to spin the piano around and use the short lid stick because I couldn't get the PZMs to match the stereo image of the AKG C-24. I just felt for the purposes of simulcasting, the image of B. J.'s music was more important.

I plan to continue using the PZM in location recording, and hopefully good studio engineers will begin using them once someone else has done the experimenting for them.

DAVID MCNUTT
Freelance Audio Engineer

db replies:
For more on PZMs, see our PZM feature story in this issue.

db Calendar

JULY

6-17 **Audio Recording Seminar.** University of Iowa. For more information contact: Prof. Lowell Cross, University of Iowa School of Music, Iowa City, IA 52242. Tel: (319) 353-5976.

AUGUST

3-7
10-14
17-21 **Digital Sound Synthesis and Sound Processing Workshop.** 18 Oliver St., Boston, MA. For more information contact: Digital Music Systems, Inc., P.O. Box 1632, Boston, MA 02110. Tel: (617) 542-3042.

10-14
17-28 **MIT Summer Session Programs on Media Technology.** Cambridge, MA. For more information contact: Director of the Summer Sessions, Room E19-356, Massachusetts Institute of Technology, Cambridge, MA 02139. Tel: (617) 253-5960.

SEPTEMBER

4-13 **International Audio and Video Fair Berlin, 1981.** Exhibition Grove, Berlin, West Germany. For more information contact: Professional Travel Management Inc., The New ASAE Building, 1575 Eye St., N.W. Suite 1250, Washington, D.C. 20005. Tel: (202) 223-6415.

13-16 **NRBA 1981 Convention.** Fountainebleau Hilton, Miami Beach, FL. For more information contact: Lisa Friede, National Radio Broadcasters Association, 1705 De Sales St. N.W., Suite 500, Washington, D.C. 20036. Tel: (202) 466-2030.

29 **Sixth Sound Broadcasting Equipment Show.** Albany Hotel, Birmingham, England. For more information contact: Carol Pottinger, Audio & Design (Recording) Ltd., 16 North Street, Reading, RG1 4DA Berks, England. Tel: (0734) 53411.

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BARRY BLESSER

db Digital Audio

Floating Point Design

• The theoretical basis for lower-quality floating point A/D and D/A converters was presented in the previous column. Now, let's look at a specific implementation of such a system in order to understand the inherent trade-offs in the design.

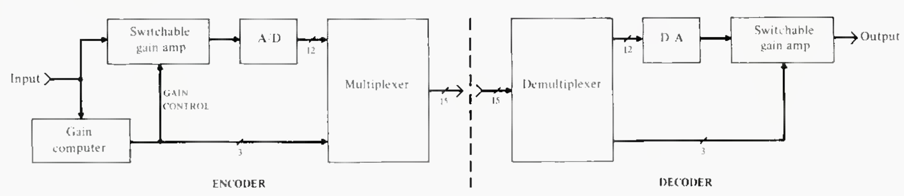
We know that the noise level from a quantizer is fixed and determined by the size of the LSB. Thus, if we can keep the signal at the converter as large as possible, then the signal-to-noise ratio can approach the dynamic range. In the usual straight PCM, we need a large number of bits because the signal could have a large dynamic range. In the floating-point implementation, the dynamic range is not as great because the signal is kept large artificially.

The system's operation depends on pre- and post-conditioning (i.e., compression and expansion), as shown in FIGURE 1. The blocks labeled "gain computer" and "switchable gain amplifier" manipulate the signal so that the converters operate at full signal level even when the input signal becomes small. The illustration does not include the input low-pass filter, input sample-and-hold, output de-glitch sample-and-hold and output low pass filter, all of which are assumed to be there.

SYSTEM BEHAVIOR

We will consider specific circuits in our discussion of the system behavior. Let's begin with a switchable gain amplifier, as shown in FIGURE 2. This amplifier is a high-speed operational

Figure 1. The floating-point encoder/decoder. Pre- and post-(A/D) conditioning keeps the converters operating at full signal level.



amplifier with a gain-bandwidth product on the order of 10-to-100 MHz. The FETs are switches which can change the value of the gain. The number next to each FET indicates the gain when that switch is closed. With no switches closed, the gain is 1. Therefore, the amplifier may have a gain of 1, 2, 4, or 8. A different choice of resistors would create other gains, while additional switches would allow more than four possible gain states. One might have gains of say, 1, 3, 9, and 27; or perhaps 1, 5, 10, and 20. There is no theoretical reason to design for a specific set of gains, although there are some practical reasons for preferring one set over another.

The gain computer is a set of comparators which compare the input signal to certain present thresholds. These thresholds must be matched to the gain states of the amplifier. To illustrate the operation, consider the gain states of 1, 2, 4, and 8: consider that the maximum A/D input voltage is 8 volts. When the input signal is between 4 and 8 volts, all the comparators are off, and all of the switches are open. The pre-conditioning has no effect since the gain is 1. Now consider the case of an input signal between 2 and 4 volts. The first comparator (A) is ON (1), and the logic results in the first switch (2) being closed. Since the gain of the amplifier is now 2, the output signal is again between 4 and 8 volts. When the input signal is between 1 and 2 volts, the upper two comparators (A, B) are ON. Therefore, the logic opens the first switch and closes the second, to give a gain of 4. Again, the signal at the amplifier output is between 4 and 8 volts. In the limit, with a large number of switches, the signal at the A/D will always be between 4 and 8 volts. This is equivalent to a compressor with an infinite compression ratio.

The A/D digitizes the signal and transmits it to the output D/A using a digital transmission or storage process. The output DAC regenerates the voltage which appeared at the input A/D system. But we must note that the correct output signal can only be determined when the gain control information is also transmitted. This may be done by incorporating a multiplex system, which is also seen in FIGURE 1.

The output switchable gain amplifier has gain values which are the reciprocal of the input gain values. For example, an input gain of 8 has a corresponding output gain of ⅛; an input gain of 4 has a corresponding output gain of ¼, etc. When we consider an input of 1.5 volts, it is amplified by 4, digitized and transmitted. At the output, digital recording gives us 6 volts and the switchable gain amplifier multiplies that signal by ¼ to give a value of 1.5 volts.

The major advantage of such an approach can be seen for low level signals. The output switchable gain amplifier will have a low gain, and this decreases the quantization noise from the output D/A converter. When the output amplifier has a gain of ¼, the quantization noise is reduced by a factor of 4. This is equivalent to an extra 2 bits.

FIGURE 3 shows an exaggerated picture of the quantization of the top half of two sinewaves. Notice that the quantization error increases with the signal level. This gives us the notion of a constant signal-to-noise ratio which is different from dynamic range. We define signal-to-noise as the noise which appears with the signal. This noise level follows the signal level to make a constant S/N. On the other hand, the dynamic range is just the ratio of the maximum signal to the noise in the absence of signal. The dynamic range of our floating point system is thus a function of the number of switches. We could have 10 switches with the last switch providing gains of 100 and 1/100 respectively. This would reduce the noise by 40 dB.

Increasing the gain of the last switch state increases the dynamic range. However, it does not effect the S/N, which is determined by the number of bits in the A/D and D/A. Because the floating point system separates these two issues, it offers a design flexibility.

still severe. Accuracies of 0.1% are typical. And these amplifiers must do their switching in less than 10 μsecs.

Another audible consequence of this algorithm is the noise during a click input. The click can be a very large peak which forces all of the switches to a low-gain state at the input (and a correspondingly high gain state at the output). Since the click has very little loudness, we hear the sudden increase in noise level. In fact, this becomes the definitive test of a digital system of the floating point type. A linear PCM shows no equivalent behavior. The noise can, however, be of relatively low amplitude depending on the number of bits in the A/D and D/A. Historically, 10 bits was considered adequate; today, 12 bits is preferred. This is not a significant economic issue, since 12-bit devices are much less expensive than they once were.

Because the floating-point system is really a compander with a linear PCM converter, the user should test it the same way that he would test a Dolby or dbx type compander. The major difference is that the floating point system does not have any problem recovering the gain control information, since this is sent with the digital bits of the A/D. An analog compander has a problem of recomputing the required gain at the output, but the floating point system knows the correct gain. Errors are only a result of component and circuit limitations. The analog compander must recompute the gain from the received signal which has been subject to corruption by phase and frequency response errors. The digital information is exact.

SPECS-MANSHIP

The published performance of a floating point system includes a very impressive dynamic range specification. One could have numbers on the order of 80, 90 or even 100 dB. In one sense, this is trivial to achieve. Consider a 12-bit A/D with an inherent 72 dB S/N combined with a 4-state switchable gain amplifier. However, unlike the previous discussion, assume that the gain states are 1, 3.3, 10, and 33. Since the no-signal case forces the gain to the highest level of 33, the output gain switch is 1/33. This corresponds to a 30 dB improvement in S/N. The published dynamic range could be as much as 100 dB. One could even create a very bad system with a 10 bit A/D and an 8-state gain switch. The last switch state could be a gain of 200. This would give a dynamic range of 106 dB. The system would sound terrible under certain kinds of program.

One manufacturer introduced a product with an 8-bit A/D with good published specifications. The product failed in the market and was withdrawn. The evaluation of a floating point system requires that attention be paid to the quality of the actual A/D in the system. This will often be published only as fine print. Alternatively, ask the manufacturer about the "true" S/N. If he does not understand the question, then ask the following: If I place a full level 1 kHz sinewave into the system and place a 1 kHz notch filter at the output of the system, what noise will I measure from the output of the notch filter? Notice that the presence of the signal forces the gain to the worst case. The notch filter then removes the signal, leaving only the noise.

Finally, there is the problem of the syllabic switching algorithm and the quality of the gain switch. This determines the size of the glitches and their frequency. I know of no manufacturer who deals with this in terms of a specification. Hence, the user must always try out such a system with real program. The best test signal is a low-frequency pulse. Such a pulse should have a slow rise time and sound like a thump. Any discontinuities, of the type shown in FIGURE 4, will be audible as tiny clicks. These pulses must be at a low enough frequency to allow the syllabic time constant to place the gain into the no-signal case.

Good Luck. Remember, one buys a system with a floating point encoder because it is much cheaper, and not because it is just as good as a straight PCM of the same dynamic range. ■

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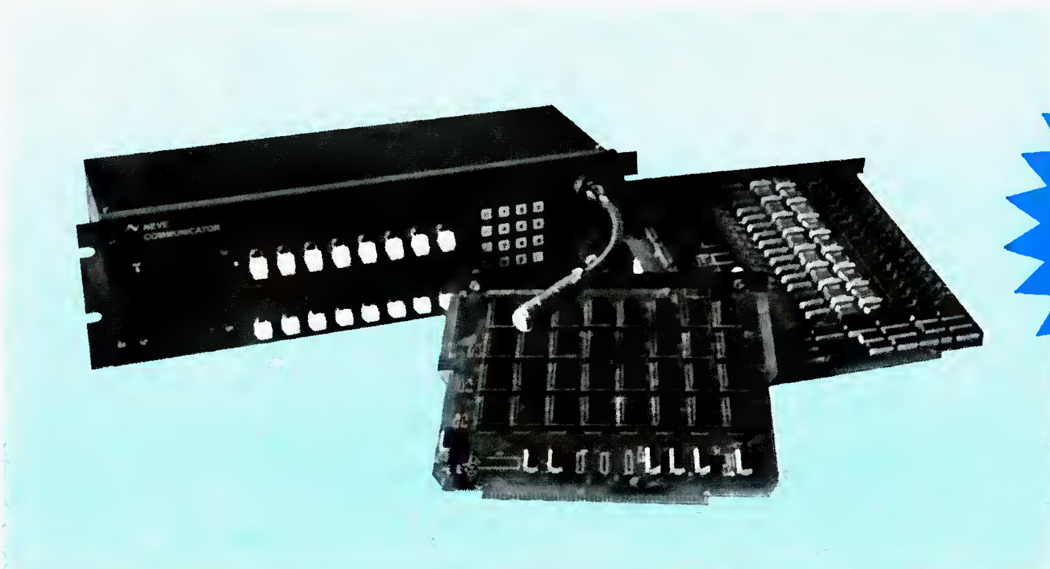

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NORMAN H. CROWHURST

A Third Alternative

• Too often in life, we try to make a decision between two alternatives, when in reality there is a third. This fact can often help us solve our problems, both in the technological realm and elsewhere. At a recent meeting, where I was conducting a discussion on the nature of human hearing, a psychologist present asked whether various hearing capabilities, such as the ability to locate the direction of a sound source, are innate or acquired.

When you get that kind of question, sometimes it helps to ask if the alternatives offered are mutually exclusive. In this particular example, it should be obvious that they are not. For it would seem that, while the *capability* is innate (that is, one is born with it), the *ability* to make that determination is acquired: one has to learn to use the capability one is born with.

PERFECT PITCH

A related question—leading to some discussion—is on the possession of perfect pitch. This led to some interesting observations that might be worth passing on here. One participant related the experience of a school child, born with perfect pitch, who suffered agonies if a musical instrument was "off" from that pitch. The instrument may have been in perfect tune, but its whole pitch was "off," and that bothered the person with perfect pitch no end.

Given a tone with a frequency of 445 hertz, in the complete absence of any other tone with which to compare it, the person with perfect pitch knows it is sharp of Concert A: if the frequency is 435 hertz, it is flat of Concert A. Only if the frequency is 440 hertz "on the nose," will it sound right to him, as a perfect Concert A.

That capability is fairly rare, but it does exist in a statistical proportion of the population. To the rest of us, the very capability seems incredible, yet it has been unquestionably verified by tests on those who possess it.

In answer to the question, "Is a person born with it, or is it acquired?", it seems to be generally conceded that some

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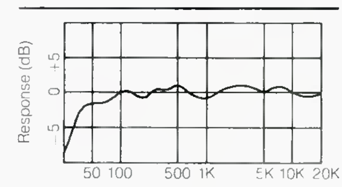

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people are born with it. But now comes the question pertinent to what we are talking about here.

Is perfect pitch a property that everyone either has or does not have, or is it really a matter of degree? Think about this: with musical training, many people can tell what key a particular composition is played in. For example, the same piece of music, played in C, in D, or in E-flat, sounds quite different.

Played in C, it sounds sort of neutral. In D, it sounds sharp, while in E flat it has a flat sound. Now, if the classical (just tempered), instead of the modern equal-tempered scale were used, this could be explained away by the slight changes in musical interval, between the various scales used by each key. But in the equal-tempered scale, the intervals are the same whichever scale is used. In any scale on any instrument tuned to equal temperament, any semitone throughout the scale represents a ratio of two to the one-twelfth power, or approximately 1.059463.

But now think about how our hearing mechanism identifies tones. Some fibers in the basilar membrane resonate, stimulating the appropriate nerve fiber, or fibers, to send a message to the hearing interpretive faculty of the brain. So even in a person without perfect pitch, changing the pitch does bring a different set of nerve fibers into play. There can be little doubt that the accurate recognition of any musical scale is an acquired skill, rather than an innate capability, although the biological "mechanism" that does it is definitely inborn.

When you think about it this way, perhaps one wonders why we don't all have perfect pitch! But the fact is, some people are definitely tone deaf, while others are at the opposite extreme, having perfect pitch. Most of us are somewhere in between.

This can lead to a whole set of questions about innate-versus-acquired capabilities. The seasoned conductor can listen to the orchestra and hear, quite discretely, the performance of any individual musician. To most other people, the orchestra is just an ensemble of sound. Is there a man-made analyzer that even approaches the conductor's capability? No. Yet who doubts that the accomplished conductor can do it?

Tests have shown that audio engineers develop a critical sense of hearing in a somewhat different way: they can tell the frequency range of the system, whether it drops off after 10 kilohertz or 20 kilohertz. They can hear various forms of intermodulation distortion that might not bother our musical conductor, because he doesn't listen to (or should that be "for"?) that: he's listening to individual musicians perform and could care less what the electronics does to it!

Another question in the realm of human hearing, concerns the effect of what many dump together under the heading "rock," by which they mean sounds with some semblance of musicality, played inordinately loud. Now that definition can get us into a whole bunch of arguments, but we'll try to get to the point of this discussion, hoping that I will not prompt someone to try and assassinate me.

We have been told that statistics have shown that continued exposure to such sound curtails hearing. And we have heard those claims denied. Presumably there are statistical data that would purport to support each conclusion. We will assume so. Now, along comes someone who says—and this too seems plausible—that whether or not such sounds have that effect on a person depends on that person's *reaction* to them: if the person doesn't want to hear them, is let's say nauseated by them, he is more likely to suffer damage than a person who derives pleasure from the same sounds. That's plausible but unproved. More significantly, when each set of data collections that led to these conflicting conclusions was going on, was any data collected on this aspect?

Probably not. The failure to look for a third alternative is often universal and is applicable in many contexts other than audio. In fact, it recurred throughout the discussion mentioned at the beginning.

The group was also interested in the discussion of digital audio that we have covered in earlier columns. That led to the question of the marvelous range of levels accommodated by human hearing. It staggers the imagination.

TECHNOLOGY VS NATURE

Can you imagine any man-made transducer with a level-handling range of 120 dB? Yet that, according to Fletcher and Munson and many other researchers, is the range of human hearing. What was then defined as the threshold of pain has a level relative to the threshold of hearing (a sound that the average person can just perceive). There is a difference of a million times the pressure level, or a trillion times the energy level.

Doesn't that stagger the imagination? A dynamic range of 60 dB, 80 dB, perhaps in some instances 100 dB: in electronics, we pat ourselves on the back, for having done well. But nature beats that, with the 120 dB range of human hearing. Discussing this brought back to my mind an incident that proves that this extraordinary capability is true, and raises some more questions.

It was back before World War II, which modern audio engineers probably regard as the dark ages. At Tannoy, we had designed a high-pressure horn unit, with a measured efficiency of slightly better than 50 percent, capable of taking 1600 watts of electrical audio power, and

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delivering some 800 watts of acoustic power. I had watched this baby being prototyped, but was on vacation when it was finally assembled for test.

At the time, I lived a little over a mile from the factory, to which I usually walked. I returned from vacation, and not yet having thought about work, I heard what I took to be a sound car, possibly electioneering. I couldn't see it looking out the window, so I went outside. I still couldn't see it, so I started walking in the direction of the sound (which was toward the Tannoy factory). I went block after block, and still the sound was coming from further away, although it did gradually seem a little louder.

Eventually I got to the main road where the trams (trolley cars) then ran. And there, facing me, from across the other side of the tram tracks, was our "baby" spouting forth. Someone was reading something from the newspaper as a test. As I got closer, the sound was still clear, and certainly not unbearably loud.

But then a tram came by. Usually, if you are close to a tram track and a tram comes by, any radio or sound system gets "drowned out" by the noise of the tram, an effect Fletcher and Munson defined as masking. You cannot hear the music for the noise of the tram. This brought back memories of trying to hold a conversation on a tram. The only way was to shout in one another's ears, to over-ride all that noise.

So imagine my surprise when a tram came by, apparently in complete silence! The voice from the loudspeaker continued, even when the tram passed between me and the horn! Incredible. That was the day I realized what the 120 dB range of human hearing really means. But would that sound be painful when you got up close? Actually, it wasn't. The only indication of it being really loud was that you couldn't hear trams go by, and it was impossible to hold a conversation in front of it. The person's lips moved, but you couldn't hear anything at all. Even shouting gave the impression he was just mouthing words, with no sound.

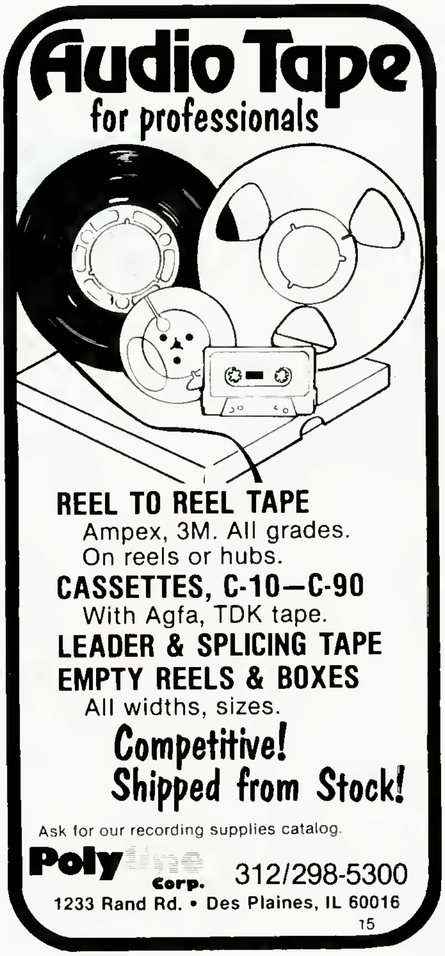
THE GREAT GRASSHOPPER EXPERIMENT

The aforementioned meeting covered a lot of ground—more than I'm going to attempt to cover here. It helped resolve some differences of opinion on some topics, such as how we really see and hear. Different disciplines develop different explanations that don't always agree, which reminds me of the story about the research into the biology of a grasshopper.

Equipment for the experiment consisted of a grasshopper, a pencil, and a lab notebook. First the experimenter placed the grasshopper by the pencil and said, "Jump." The grasshopper jumped. After the experimenter had retrieved the grasshopper from the other end of the lab, he ripped off one of its long hind legs and placed it by the pencil again. This time, he had to shout Jump somewhat louder before the grasshopper would jump, and he did not jump nearly as far. He retrieved the grasshopper again, ripped off the other hind leg, and placed it in position again. But this time, no matter how loudly he shouted, and however close he got to the grasshopper, the grasshopper did not jump. So he took up his pencil to write his conclusions in the lab notebook.

"The experiment proves that a grasshopper's hearing is in some way connected to his long hind legs. Removing one of them makes a grasshopper somewhat deaf. Removing both of them makes him completely stone deaf."! There you are. I said this story was related to audio. Perhaps you'd like to add a comment about the validity of conclusions?

While the absurdity of that one should be obvious to anyone, in principle it is not too different from many other conclusions one encounters in life that are solemnly accepted as "gospel." ■



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IF YOU NEED $5,000...20,000 EVEN UP TO 500,000 TO START A NEW BUSINESS OR TO EXPAND AN EXISTING FIRM—THEN READ WHY YOU TOO WILL CALL THIS INCREDIBLE MONEY RAISING MANUAL

"THE SMALL BUSINESS BORROWER'S BIBLE"

Practically prepares the loan application for you line-by-line...the "proper" way. All properly prepared applications are processed faster...no red tape! Guaranteed Loans...Direct Loans...and Immediate Loans are available now!

Most men and women seriously interested in starting their own business are eligible to apply — including those who already own a business and need capital fast for expansion...or to stay afloat...even if they've been flatly refused by banks and turned down elsewhere! Yet, too, many never qualify, simply because they do not know how to "properly" prepare the loan application...

In order to help those people applying for these guaranteed and direct loans fill out their loan applications the "right way," ICC Business Research, through its diligent compilation and research efforts, has successfully assembled and published a comprehensive, easy-to-follow seminar manual: The Money Raiser's Guaranteed and Direct Loans Manual, that will **quickly show you practically everything you'll need to know** to prepare a loan application to get federally Guaranteed and Direct Loans.

Here are just **some of the many important benefits** the Money Raiser's Guaranteed and Direct Loans Manual provides you with:

- **a completely filled in sample set of actual SBA loan application forms, all properly filled in for you to easily follow—aids you in quickly preparing your own loan application the right way. Each line on the sample application forms is explained and illustrated in easy-to-understand language.**
- **fast application preparation procedures for getting loans for both new start up business ventures and established firms.**
- **advises you on how to properly answer key questions necessary for loan approval and in order to help avoid having your application turned down—gives you advice on what you should not do under any circumstances.**
- **what simple steps you take to guarantee eligibility—no matter if you do not presently qualify.**
- **where you can file your application for fastest processing.**

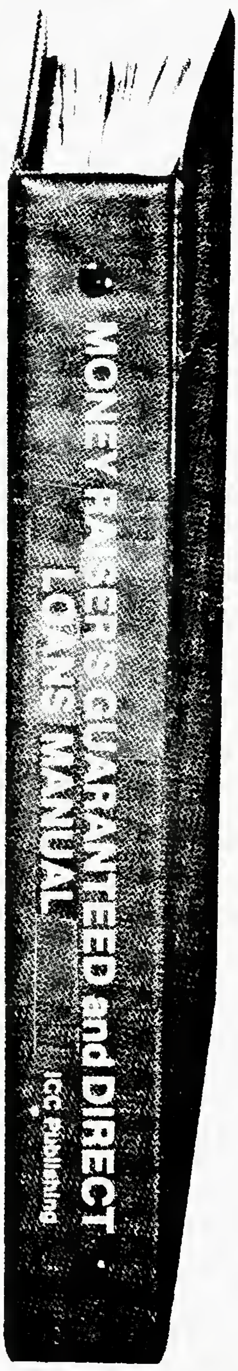
At this point the most important question you want answered is: Just where is all this loan money coming from? Incredible as it may sound—these Guaranteed Loans, ...Direct Loans...and Immediate Loans are indeed available right now — from the best, and yet, the most overlooked and frequently the most ignored and sometimes outright ridiculed, "made-fun-of" source of ready money...fast capital, in America — THE UNITED STATES GOVERNMENT.

Of course, there are those who upon hearing the words "UNITED STATES GOVERNMENT" will instantly freeze up and frown and say:

"...only minorities can get small business loan money from the government!"

Yet, on the other hand (and most puzzling) others will rant on and on and on that

"...don't even try, it's just impossible — all those Business Loans Programs are strictly for the Chryslers, the Lockheeds, the big corporations...not for the little guy or small companies," etc.



Still there are those who declare:

"...I need money right now...and small business government loans take too darn long. It's impossible to qualify. No one ever gets one of those loans."

Or you may hear these comments:

"...My accountant's junior assistant says he thinks it might be a waste of my time!" "Heck, there's too much worrisome paperwork and red tape to wade through!"

Frankly — such rantings and ravings are just a lot of "bull" without any real basis — and only serve to clearly show that lack of knowledge...misinformation...and and not quite fully understanding the UNITED STATES GOVERNMENT'S Small Business Administration's (SBA) Programs have unfortunately caused a lot of people to ignore what is without a doubt — not only the most important and generous source of financing for new business start ups and existing business expansions in this country — but of the entire world!

Now that you've heard the "bull" about the United States Government's SBA Loan Program — take a few more moments and read the following facts:

- **Only 9.6% of approved loans were actually made to minorities last year**
- **What SBA recognizes as a "small business" actually applies to 97% of all the companies in the nation**
- **Red tape comes about only when the loan application is sent back due to applicant not providing the requested information...or providing the wrong information**
- **The SBA is required by Congress to provide a minimum dollar amount in business loans each fiscal year in order to lawfully comply with strict quotas. (Almost 5 billion this year)**

Yet, despite the millions who miss out — there are still literally thousands of ambitious **men and women nationwide** who are properly applying — being approved — and **obtaining sufficient funds** to either start a new business, a franchise, or buy out or expand an existing one. Mostly, they are all just typical Americans with no fancy titles, who used essentially the **same effective know-how** to fill out their applications that you'll find in the Money Raiser's Guaranteed and Direct Loans Manual Manual.

So don't you dare be shy about applying for and accepting these guaranteed and direct government loans. Curiously enough, the government is actually very much interested in helping you start a business that will make a lot of money. It's to their advantage — the more money you make the more they stand to collect in taxes. In fiscal 1981, our nation's good old generous "uncle" will either lend directly or guarantee **billions of dollars** in loan requests, along with technical assistance and even sales procurement assistance. Remember, **if you don't** apply for these available SBA funds **somebody else certainly will.**

GUARANTEE #1

Simply — look over this most effective money raising loan preparation assistance manual for 15 days — and, then, if you are not convinced that it can actually help you obtain the Business Loan you need right away — just return it for a full and prompt refund.

GUARANTEE #2

Even after 15 days — here's how you are still strongly protected — if you decide to keep the manual — and you apply for an SBA Loan anytime within 1 year, your loan must be approved and you must actually receive the funds or your money will be refunded in full.

Don't lose out — now is the best time to place your order for this comprehensive manual. It is **not sold in stores.** Available only by mail through this ad, directly from **ICC Business Research, the exclusive publisher,** at just a small fraction of what it would cost for the services of a private loan advisor or to attend a seminar.

For example:

*Initially, this amazing Guaranteed and Direct Loans Manual was specially designed to be the basis of a Small Business Loan Seminar — where each registrant would pay an admission fee of $450. But our company felt that since the manual's quality instructions were so **exceptionally crystal-clear** that anyone who could read, could **successfully use its techniques** without having to attend a seminar or pay for costly private loan advisory assistance services.*

Therefore, for those purchasing the manual by mail, no 3 day class, no course and accommodations are required. And rather than $450 we could slash the price all the way down to just a mere $35 — a small portion of a typical seminar attendance fee — providing you promptly fill in and mail coupon below with fee while this special "seminar-in-print" manual offer is still available by mail at this relatively low price!

Remember, this most unique manual quickly provides you with actual sample copies of SBA Loan application and all other required forms—already properly filled in for you to easily use as reliably accurate step-by-step guides—thus offering you complete assurance that your application will be properly prepared, and thereby immediately putting you on the right road to obtaining fast, no red-tape loan approval.

YOU GET NOT 1 BUT 2 STRONG BINDING GUARANTEES!

YOUR LOAN MUST ACTUALLY BE APPROVED OR YOUR MONEY BACK

Of course, no one can guarantee that every request will be approved — but clearly we are firmly convinced that any sound business request properly prepared — showing a reasonable chance of repayment and submitted to SBA — will be approved. Only because we are so confident that this is a fact do we dare make such a strong binding seldom-heard-of Double Guarantee. **No stronger guarantee possible! It actually pays for you to order a copy of this remarkable manual — *100% tax deductible as a business expense* ...Don't delay—send for yours right now!**

THE EASY NO-NONSENSE WAY TO RAISE CAPITAL FAST!

NO RISK LOAN OPPORTUNITY FORM

Detach and rush for COMPLETE PREPARATION ASSISTANCE FOR LOAN APPROVAL

Please rush me ________ copies of the "Money Raiser's Guaranteed and Direct Loans Manual," each at a $35 fee plus $2.50 handling and shipping under your 2 strong binding Guarantees.

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Winston-Salem N C 27105

New Products & Services

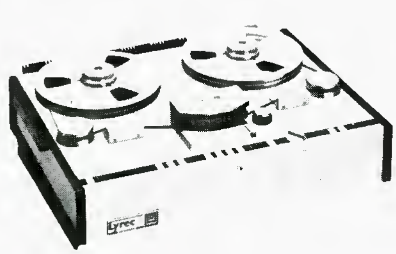
MICROPHONE SERIES

• The 671B series of dynamic cardioid microphones consists of the 671BL and 671BH models, which are low-impedance and high-impedance versions of the same microphone. The close-up bass boost and high-frequency characteristics of the 671B make it a useful vocal microphone, while a locking on/off switch makes it ideal for public address and sound reinforcement applications as well.
Mfr: Electro-Voice
Price: 671BH—$97.00, 671BL—$95.00
Circle 40 on Reader Service Card

POWER CONSOLE

• The 706 Series Console is a simple-to-operate power console. Among its features are a 200-watt amplifier; a 10-band combining-type graphic equalizer; a feedback finder circuit for identifying feedback frequencies; built-in simplex power supply for professional condenser microphones, and a Patch Block patch panel that is a combination block diagram of the console's internal circuitry together with eight patching jacks located at appropriate points right on the block diagram.
Mfr: Shure Brothers Inc.
Price: $1,190.00
Circle 41 on Reader Service Card

TAPE RECORDER

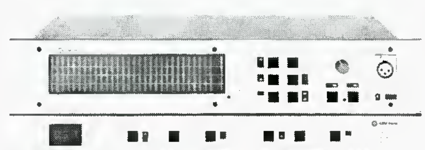


• Well suited for mastering, editing, and mobile applications, the Lyrec TR55 Tape Recorder contains a number of new approaches and features including a new servo-controlled DC discmotor that makes it possible for the TR55 to achieve a winding speed of 500 ips. The tape deck accepts any reel size from 3 to 14 inches in any combination. The reel motors operate in a double push-pull system such that tape tension is kept within close tolerance. To help locate edit point, there is a search system allowing search to either cue or zero. Continuously variable winding speed further helps the operator. The actual cutting is done by a tape cutter that is mounted in the head block and actually cuts right in front of the headgap.
Mfr: Lyrec Manufacturing
Circle 42 on Reader Service Card

MONITOR SPEAKER

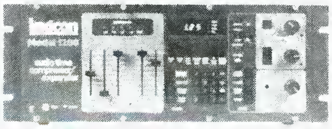
• The compact MS-100 combines a high 100 watt handling capacity with a high degree of flexibility. The MS-100 is capable of reproducing any signal from mixing boards or instrument amplifiers and has a frequency response of 100 Hz to 17 kHz. Input and Output jacks are provided, allowing the parallel connection of multiple units on the same line. Maximum flexibility is achieved through a six position switch which allows the MS-100 to operate at 16 ohms at full volume (0 dB) or attenuated to –6, –12, or –18 dB, to operate two units in parallel at 4 ohms without a drop in volume, or to switch the Monitor off when not in use. The unit is designed to stand on any flat surface, or to be mounted on a microphone stand with the adapter provided.
Mfr: RolandCorp US
Price: $165.00
Circle 43 on Reader Service Card

REVERB DECAY ANALYZER



• The new RT60 Reverb Decay Analyzer is a versatile measurement device to be used in conjunction with the Klark-Teknik DN60 Real Time Analyzer. The RT60 gives the user control over many of the parameters of decay analysis. A curser switch allows the choice of measurement using any single ISO 1/3 octave frequency or the total bandwidth. The RT60 also allows the user to select any portion of the time window from 0 dB to –30 dB in 2 dB increments. The RT60 will plot the decay curve, displaying the results on the DN60. The choice of 16, 64, or 208 milliseconds gives the user control over the horizontal resolution of the plotted curve. The RT60 also allows the user to accumulate up to 32 separate curves, enabling a true averaging of different point measurements.
Mfr: Klark-Teknik Electronics Inc.
Price: $1,395.00
Circle 44 on Reader Service Card

THERE'S A NEW PRODUCTION TOOL IN TOWN



lexicon

Model 1200 Audio Time Compressor automatically reduces or expands the play time of recorded material . . . gives you "time-tailored" programming at the push of a button . . . preserves original pitch and voice quality.

● With the remarkable new Lexicon Model 1200, you can now eliminate time-consuming retakes to fit commercials or other material into available time slots. For the first time, you can speed up tapes or slow them down — and get broadcast-quality sound free of distortion. ● You simply connect the 1200 with virtually any variable-speed tape recorder, set the timer for the on-air or play time you require, and you're in business. Material that runs too long can be compressed up to 25%. You can time edit to add tag lines or heighten the energy of the message. In TV applications, the Model 1200 can be teamed with a variable-speed film projector and/or a videotape recorder. ● The Model 1200 — a product of Lexicon's eight years of leadership in digital audio processing — marks a breakthrough in bringing time-processed audio to the level of quality necessary for radio and TV use. Based on sophisticated computer technology and proprietary intelligent digital processing techniques, the 1200 has been thoroughly field-tested in the production of nationally broadcast commercials by 19 of the top 20 advertising agencies. Write for detailed information and application notes.

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Export: Gotham Export Corporation, New York, NY 10014

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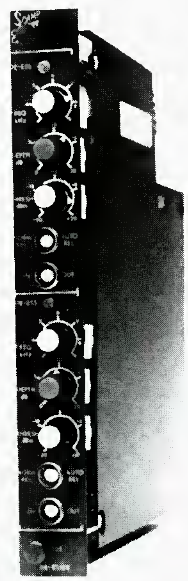
TAPE CARE AND MAINTENANCE PRODUCTS

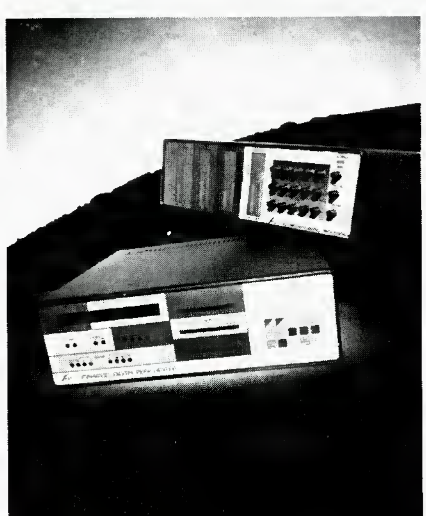
• Nortronics Company has introduced a line of professional tape care and maintenance products. Called the Proformance Series, the product line is aimed at the needs of the audio professional. All products in the Proformance Series meet or exceed the standards set by NAB, UL or other applicable professional testing organizations. Product categories include: head cleaners, demagnetizers, splicing accessories, and alignment tapes.
Mfr: Nortronics Company, Inc.
Circle 45 on Reader Service Card

DE—ESSER MODULE



• The S25 De-Esser Module, a dual mono or stereo processor, takes the incoming signal and splits it into two components, the main band and the ess band. The side chain has a sharply tuned variable filter which can be swept over the 5 kHz to 15 kHz range and which has characteristics which essentially sense the presence of "ess" iness. Three controls are provided: frequency, variable from 5 to 15 kHz; threshold, variable from −30 dBm to completely insensitive, and depth, variable from 0-20 dB. The Scamp S25 De-Esser Module attenuates only the ess frequencies which have been selected by the user, within certain parameters. The S25 has an In/Out switch per channel for A-B comparison. A Variable Neper Generator (VNG) is used to generate attenuation in a logarithmically controlled fashion.
Mfr: Audio & Design (Recording) Ltd.
Circle 46 on Reader Service Card

DIGITAL PEAK LIMITER



• According to its manufacturers, the Maximod is the industry's first and only digital peak limiter. The limiter incorporates a microprocessor which compares and samples the audio, prior to becoming the output. Thus, there is virtually no overshoot. What this means is on AM, where the FCC allows 125 percent, you may set the controls for 125 percent. Through command via the front keypad, various functions are made available. Positive and negative modulation limits are set, as are symmetrical or asymmetrical modulation, phase reversal, amount of limiting, high and low pass filtering, preemphasis for FM operation, gain and output, and variable release time. The keypad is locked by entering a private code.
Mfr: Applied Technology Corporation
Circle 47 on Reader Service Card

LIMITER/COMPRESSOR

• The LC-2 features input and output level controls, as well as Attack, Release, and Compression Ratio controls. An LED style meter displays gain over a 20 dB range. LEDs are also provided for overload and power indication. Front panel pushbuttons select between normal compression and de-essing or side-chain modes. These allow for the gain reduction signal to be processed through either an internal hi-pass filter or external equalizer, respectively. The side-chain input may also be used alone to allow a completely unrelated signal to control the gain. Additional features include: modular circuit-board construction for easy servicing; program-adjusted release time, and a three pole filter in the de-essing circuit that assures that only sibilance frequencies trigger gain reduction.
Mfr: Furman Sound, Inc.
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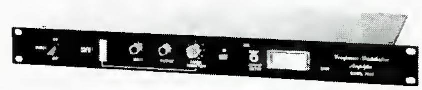
LEVELIZING DISTRIBUTION AMPLIFIER



• The Model 7833 contains a full-feature Model 3320 Gain Reduction Amplifier and a 1 x 8 Distribution Amplifier. The input to the 7833 is balanced, bridging, differential and the eight outputs are balanced 600 ohms to +20 dBm each. The dynamic range reduction characteristic of the 7833 is a "soft knee," gradually increasing the rate of gain reduction as more is required. This characteristic, along with a 10 microsecond attack time constant, an automatic variable release time, and a very wide operating range make the 7833 particularly suited for oaging or other systems that are left unattended and might expect a wide range of input levels. The front panel controls include all of those found on the 3320, an Input level control (threshold), an Output level control and a Range reduction control, to set the degree of dynamic range reduction to 30 dB while maintaining a constant peak output. An LED bar graph constantly indicates the amount of range reduction. An In switch with LED indicator permits the range reduction to be removed or inserted at will. The Distribution Amplifier portion of the 7833 is screwdriver adjustable from 0 to 10 db gain through the front panel. A VU meter is switch-selectable to monitor either the Input or Output levels.
Mfr: Modular Audio Products
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What the Shadow doesn't know ...it learns.

The BTX Shadow System:

- synchronizes and controls ATRs and VTRs to sprocket-lock, subframe accuracy
- learns each recorder's dynamic characteristics to optimize machine control
- maintains minimal offset during GO TO and CHASE functions
- runs independently, allowing hands-off synchronization in all control modes
- uses standard RS 232C interface to permit present and future computer automation, including electronic editing
- available with optional control console, which displays time and status data and allows keypad entries of destinations, GO TO and FOLLOW commands, and data storage/retrieval
- priced to permit each transport to have its own Shadow

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btx SHADOW

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db Editorial

Microphones and Milestones

SOONER OR LATER, someone will get around to writing a history of the recording industry, chronicling all of those milestones which we have passed over the last hundred years.

Surely, the list of milestones will include such things as the LP, stereo recording, multi-track, sel-sync, automation, etc. In historical perspective, will digital recording be regarded as a milestone or a millstone? We shall see.

And what about this month's featured subject: microphones? What microphonic milestones need to be remembered in our history book?

Certainly, the development of uni-directional microphones deserves a prominent place. In our August, 1978 issue, we reported on the early work of the late Benjamin B. Bauer, who in the mid 'thirties did some of the pioneering in this area ("A Backward Glance at Cardioid Microphones"). A page from one of his many patents graced the cover of that issue, and another of his patent illustrations faces this editorial page.

Blumlein, of course, deserves a prominent position, as does Dr. Harry Olson, whose early ribbon microphones are still much sought after by many studio owners.

Perhaps it's still a little too early to properly evaluate some of the more recent contributions to the state-of-the-(microphone)-art. For example, what will history have to say about the Sound Field microphone? the Ghent Microphone? And, coming closer to the immediate, what of the PZM system? Is it a revolutionary breakthrough in new technology, or just a better mousetrap?

We'd be willing to bet that once the smoke clears, PZM will be seen as a better mousetrap. Like the humble paper clip, it is an ingenious application of old—not new—technology. In fact, its simplicity puts it in the "Now why didn't *I* think of that?" category.

The technology has been staring us in the face for years. We've all read about what happens when sound waves hit boundary surfaces. Is there a studio engineer anywhere who has not thought about what happens when sounds are reflected off nearby walls? Probably not. But few of us have thought of the boundary as an integral part of the microphone itself.

Yet, the basic idea has been seen before. Electro-Voice's "mike mouse" (model 411) has been around for about ten years, and Shure Brothers are manufacturing "long distance" microphone stands (S53P and S55P). In both cases, the devices are used to place a microphone very close to a surface—usually, the floor.

But it took PZM developers Ed Long and Ron Wickersham to go the additional distance (less than one inch) and conceive of the microphone and the floor as being part of a unified system.

In 1978, Ken Wahrenbrock built some of the first PZM prototypes, followed shortly by a limited supply of production models. In 1980, Crown International gave the prototypes a face lift, and began full-scale production of several versions.

People have been reacting to PZM ever since. Some of the most impressionable types are proclaiming it as the one-and-only answer to just about every question on mikes and miking. Others tell us it'll never work. As usual, reality lurks somewhere in between.

What's encouraging to us is that PZM forces us to re-think our approaches to microphone placement. To the familiar cardioid, figure-8 and omni patterns, we must now add the hemispherical PZM pattern. But we should remember that it is an addition to—not a replacement of—our more-familiar microphone types.

As you may suspect by now, this issue features an article on PZM. And, we've included another excerpt from our soon-to-be published *Microphone Handbook*, written by John Eargle. And then, there's a little something on the mathematics behind the microphone. The equations are not the sort of thing to quicken the pulse of the recording engineer, but taken in small doses, they may help give a better understanding of what happens when more than one microphone is plugged in. ■

"Our Auditronics 720 combines recording studio quality with live broadcast flexibility,"

says Graham Simmons, Chief Engineer at Miami's WPBT — Channel 2. "Auditronics developed this 36 in — 16 out audio mixing console to give us all the EQ, reverb and signal processing we need for studio quality multi-track recording of our productions."

"When you're a national production facility like we are, you've got to have an audio signal path that's strictly state-of-the-art. For example, our Auditronics 720 preamp design is the latest generation and gives us the best signal-to-noise performance available. It allows us to do multiple generation dubbing and mixing without noise build up."

"In addition to its multi-track recording advantages, our 720 has the flexibility to do all the necessary mixing and signal processing in real time for a mono mix for TV, a stereo mix for FM simulcast, including network satelite feeds, as well as a scratch mix on videotape for later synchronization in post-production editing."

"Whether it's multi-track recording, live broadcast or post-production, our Auditronics 720 does everything we want it to do, and does it very nicely."

If you'd like to know what WPBT-Miami and over 300 other satisfied users know about Auditronics broadcast consoles, circle reader service number or contact:

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CHRISTINE KOFOED

Billy Bob's Texas: Billy Bob Buys Bull Bar and Bistro

Or... Blasting Boxer Bass Horns Baffle Buckaroos.

THE IDEA WAS BARELY on paper before the rumors began to fly. Biggest Nightclub in the World! Seven acre parking lot! A real indoor rodeo with *live* bulls! Forty-two bar stations with 2,100 gallons of beer on tap at any given moment, a 14,000 square foot dance floor, a quarter-million dollar sound system, all the biggest names in country entertainment booked to play within a year! Why, they've even got their own zip code!... Where? Why in Texas of course! Billy Bob's Texas, 76106!

The best part about it all is that the rumors were true. The biggest nightclub ever conceived opened its doors April 3 in the Fort Worth stockyards, and while the club is still sharing its zip code with some of the other folks in the area, there's no question that Billy Bob's Texas is the Club of the Decade.

The sound system, which is our concern, is the responsibility of Mr. Randy Vaughn, owner of Ambassador Enterprises of Norfolk, Virginia. Mr. Vaughn, a two-time Syn-Aud-Con graduate, has been designing and installing sound systems for over ten years and has completed over 400 installations around the world. But even with all this experience, Billy Bob's is the largest and most complex job he has tackled to date. Besides its amazing size (over 100,000 square feet of dancing, eating, drinking and listening space, documented and approved by those tireless people at the Guinness Book of World Records), the club had many other built-in idiosyncrasies guaranteed to endear it to the professional sound designer. Originally built in the early 1900s as a cattle holding area, the building has concrete floors, a ceiling with many different heights in just as many places (the highest and lowest points are located just over the stage), and both a ceiling *and* a floor which slope thirty feet from the northwest corner to the southwest corner.

"When I first saw the club," Randy said, "it was early December, and I had to rely a great deal on what everybody said it was going to be, because although it had walls, and beams to hold the ceiling up, there wasn't much else. Bulldozers were sort of pushing the floor around, pick-up trucks were running through... it was difficult to visualize at first. We spent about six hours walking through the building, discussing what was where and how it would operate. In advance of our coming down, Billy Bob's people had sent us some floor plans and the preliminary specs they'd received from other sound contractors who were bidding on the job. When I looked at those plans, I saw some major holes, because while the bidders had for the most part specified quality equipment, no one had even mentioned a delay line—and just looking at the size of the place, you knew there was no way that it could be done without a delay of some sort. In fact, my first quote was for *just* the delay system. At the time, before I met the owners, I thought that they might want to book National acts occasionally, but wouldn't do it too often, and on those occasions they would simply rent a large concert system. But as soon as I found out that they would have at least 110 performers of National stature in the first year alone, we showed them in about five minutes that it was more economical to buy and install the entire system."

"We were first contacted on December 15, 1980," Randy continued. "We actually got the criteria for designing the system a month later. We signed the quote on February 20th and had the system totally assembled and on a truck for Ft. Worth by March 13th.

"The way I usually design a sound system is to first design it the way I would do it if there were no limitations, no restriction on the amount of money available to make it a first class system. Then I ease it back to fit the budget that is actually available. Well, in this case I designed what I wanted to use and they said, 'Fine. Go ahead.' So this is it. A no-compromise system in the biggest club in the world."

The most unique feature of the club and its sound system is its modularity. The main floor has been broken into several staging areas: The National area, which encompasses the entire club, the Regional area, which covers two thirds of the club, and the Local area, which involves only the back third of the main room and accommodates 2,000 people along with 27 pool tables and over 30 electronic games. In the Regional mode of operation there is plenty of room to seat another 1,000 people. The Regional area also encompasses one of the dance floors. The National mode involves seating for 4,000 people, two dance floors and the 24 x 60 ft. main stage.

The sound system had to cover not only the 100,000-foot club floor when it was being used in the National mode but also each of the other two areas when they were being operated independently. The main stacks, splayed to left and right of the stage, are four-way, stereo, and totally horn-loaded. Two rows of delay clusters are used to augment the main stacks. The first delay cluster, which is 132 feet from the concert stacks, consists of four bi-amplified clusters flown from the ceiling. The Regional stage is placed at this point, so that the delay speakers are the main sound system when the Regional-sized room is in operation. If a more sophisticated system is needed in the Regional mode, half of the main stacks can be rolled out to the Regional stage and patched in. The second delay cluster, about 72 feet from the first delay, is capable of being used independently when the lobby and main bar area are being used

Christine Kofoed is the executive vice president of Community Light & Sound, Inc.

Welcome to Billy Bob's Texas!

as a game room and cocktail lounge. Of course in Texas they don't call it a cocktail lounge, they call it a bar. In fact, the owners insist on calling the entire building a bar.

BILLY BOB'S: THE PERFORMER'S CLUB

Prior to an engagement at Billy Bob's Texas, each act will be sent a three-ring binder containing comprehensive information on the sound system and on the club itself. The binder includes a complete description of equipment available to the performers and their road crew, a map of the staging area and mixing locations, a list of microphones and stands, and a flow chart of the system. Following that are manufacturer's specification sheets on each of the components used in the system.

HARDWARE

The main mixing console is the Yamaha PM2000-32, a 32-input board with eight sub-master channels, eight matrix outputs, four foldback channels and two channels of effects. The Yamaha was chosen not only because it could handle the operation with aplomb, but because it is a very easy board to understand. Many times a date at Billy Bob's will be part of an extended tour, and it will be necessary for the system to be easily mastered by those who are used to their own equipment.

Signal processing at Billy Bob's is provided by two UREI 537 one-third octave graphic equalizers, two LA4 compressor/limiters, and two UREI 525 five-way electronic crossovers.

The eight large concert reinforcement stacks, four per side, each consist of one of Community Light & Sound's new Boxer Bass Horn loaded with JBL 2240 18-inch bass woofers. There are two straight-axis bass horns loaded with JBL 2220H 15-inch woofers, a Community Light & Sound S90/365 90-degree radial horn with a 2441 driver, and a Community SQ90 radial horn loaded with JBL's new 2421 compression driver.

The eight concert stacks are powered by four power amp racks, each containing one Crown PSA-2, one DC300-A and one D150-A. A combination of permanently installed and portable cabling is provided to allow the mixing location to be positioned anywhere a group desires between 50 and 250 feet from the stage. Connectors on the speaker stacks are all twist-lock. All of the AC power for the amplifier racks and the signal processing rack is totally isolated from anything else coming into the building.

The large portable effects rack contains a Crown RTA-2, the compressor/limiters, graphic equalizers, crossovers and the delay lines. The delay lines are both Delta Labs DL-1 digital delay systems. The first delay system consists of four bi-amplified clusters each made up of two straight-axis bass horns powered by two JBL E140 15-inch speakers, two Community SRH90 radials stacked and splayed, each with a JBL 2470 compression driver on the back, and four piezoelectric tweeters. The second delay system (the four bar area speakers) is exactly half the size of the first delay system—one bass horn, one radial, two piezos. The delay systems are powered by one Crown PSA-2 power amplifier and two Crown DC300-As.

Microphones available to the performer include four Crown PZM Pressure Zone Microphones, six Shure SM58s, fourteen Shure SM57s, six Shure SM77s, four Shure SM78s, and six UREI 315 passive direct boxes.

MONITOR SYSTEM

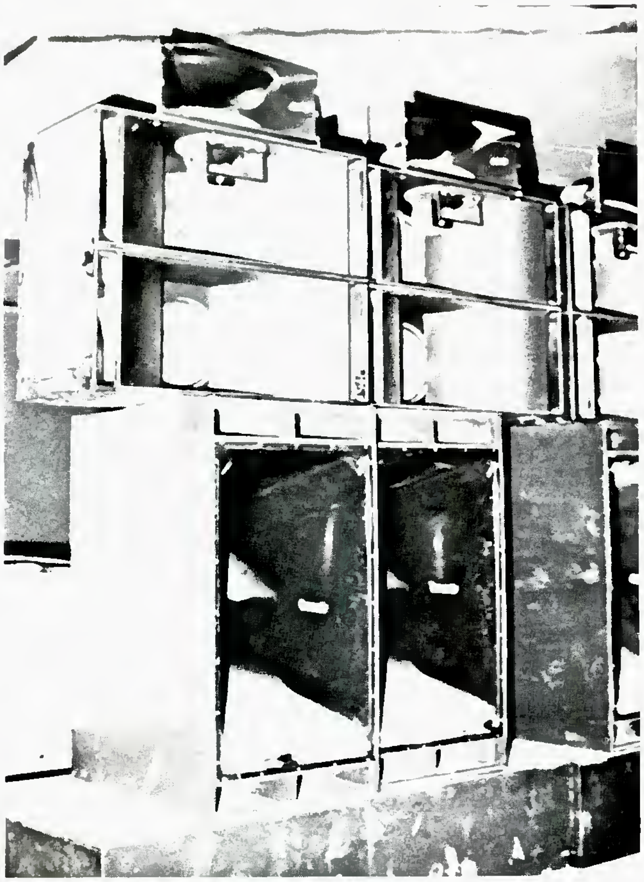
The stage monitor mixing console is a Stevenson Interface 32x8. The eight stage monitors (each containing a Community EW500 straight horn and a JBL 2470 driver, a JBL E140 15-inch woofer, the JBL 3105 passive crossover and a 2405 tweeter) are two-way active, three-way passive. The system also contains four UREI 567 PA processors and two PSA-2s and DC300-As.

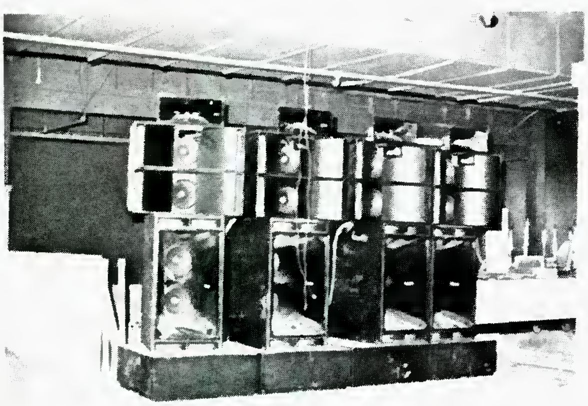
Currently a four-way monitor mix, the monitor system will be expanded to eight-way in the near future by the addition of four more PA processors and additional amplification.

"There was considerably more than the usual cooperation from all of the manufacturers represented in this system because of our time schedule," Randy Vaughn stated, "and we sincerely appreciate all of the help that we got. We used a great deal of either new or prototype equipment. We had help from Mark Gander, the engineer at JBL who designed the new 18-inch loudspeaker, help from Community in getting the new SQ90 out to us and rushing out plans for the Boxer system. JBL provided us with prototypes of their new 2421 high-frequency driver with the diamond-type surround. And at the time of putting this sytem together, we weren't a PM2000 dealer, but Yamaha gave us a great deal of help in acquiring the board and teaching us to interface it with all the other equipment. UREI was kind enough to rush the electronic crossover—we got great cooperation from everybody. And then there's Sam Helms. Back home Sam is my Rep for Crown and Community, but here he's the Chief Engineer for the job. He's amazing."

The effects console.

Some of the speaker stacks employed at Billy Bob's.





Two months after it was designed, Billy Bob's quarter million dollar sound system was installed, amid 300 tons of air conditioning, ninety ceiling fans and thirty-odd smoke eaters. The eight four-way concert systems were stacked and the delay clusters hung, while down on the floor an enormous construction crew labored frantically to complete what seemed like acres of dance floor and hundreds of feet of rough-hewn railing. The beautifully finished effects console and the dazzling PM2000 were rolled out onto the floor to a temporary mixing location. All connections were completed and the system turned on.

Two hundred Stetsoned carpenters, floor layers and welders stopped mid-drawl and listened as a tape recording of Eddie Rabbit emanated loudly and muddily from the black stacks at the front of the room, rolled out along the air conditioning ducts and up the sloping concrete floor, hit the rear wall and rolled back again. The reverberation in the room was astounding, and a phasing problem was very much in evidence. The system was shut down and eventually Eddie Rabbit faded away. "Rock and ROLL!", someone yelled, and the hammering and welding resumed. After a short, animated discussion among the sound crew on the subject of JBL, black terminals and positive voltages, some plugs were pulled, phases reversed, and the system re-fired. The hope was expressed that the addition of the soon-to-be-installed carpet and six thousand warm bodies would deaden the room a bit, and there was general agreement that *something* would have to be done about the proximity of the duct work to the mouth of the concert stack at stage left.

The rest of the evening was spent in a downpour of pink noise as the room was analyzed and the system equalized. The construction crew went home for dinner. The sound crew stayed on. Sam Helms enthused over the RTA-2 mounted in the effects rack.

"The real time analyzer is the most significant tool ever introduced in the audio industry. *Finally,* an instant answer to the 'Sound Authority.' Now, when some fool walks in off the street and says, for instance, that you don't have any bottom in the system, all you have to do is point to the straight line that goes to 31.5 and tell him to take a hike."

Around ten o'clock, the construction foreman lumbered through carrying a hammer and dragging some 2x4s, and proceeded to nail the giant barn-like doors shut against an impending dust storm. Outside, across the gravel parking lot in the Cowtown Coliseum, horses and riders wheeled in a dance formation, practicing for the Rodeo opening scheduled for the following Saturday night.

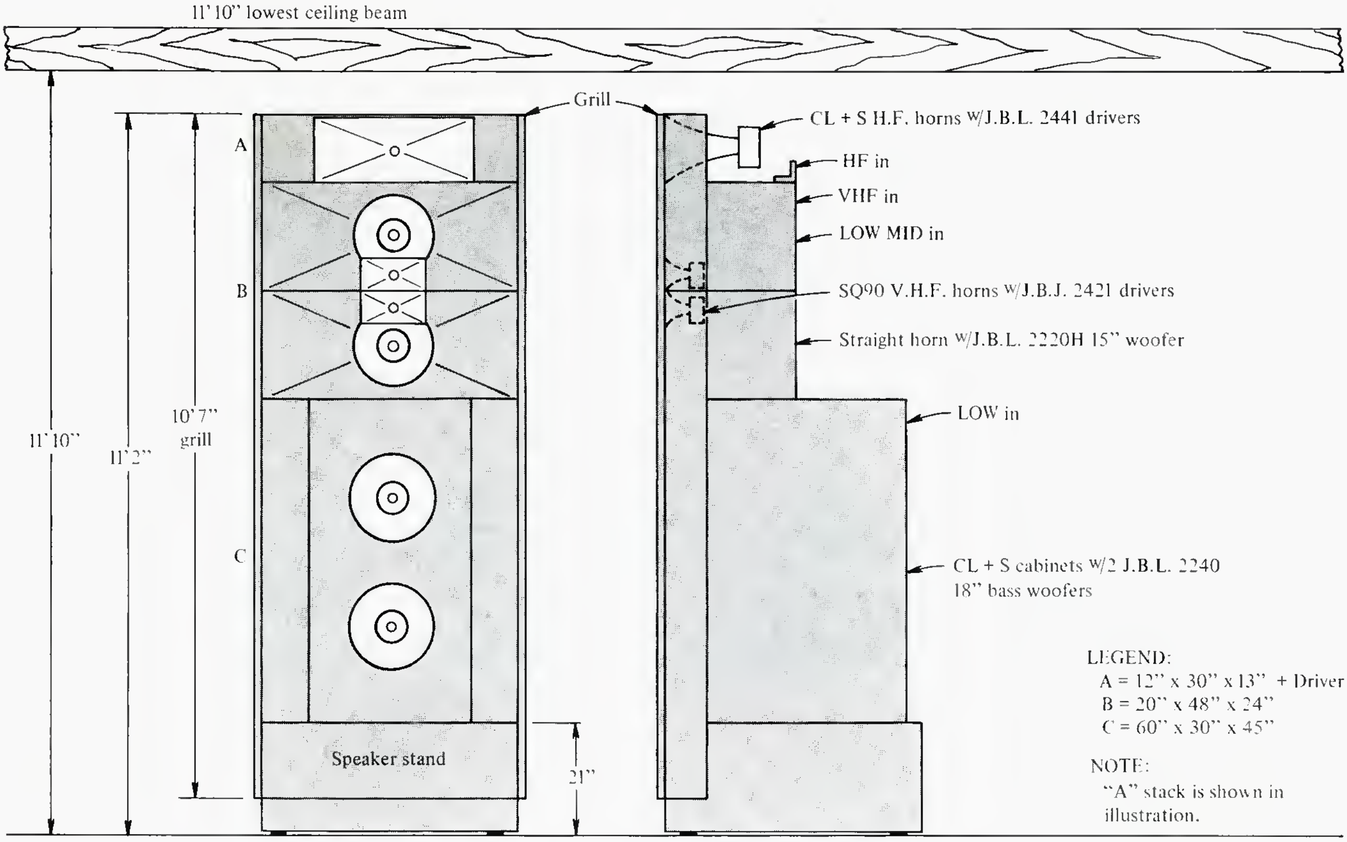
The next morning the delay lines were brought in. "What I'm looking for in a delay," Sam said, "is total intelligibility for the vocals—the frequencies between 400 and 6000 Hz. The other is that I want the level back there past the delay clusters not to be any higher than it would be perceived in traveling the distance from the stage."

Levels were matched left and right and then front-to-rear, so that the first delay line was down 6 dB from the main stacks, with the second delay down 3 dB from the first. Timing was set at 56 and 72 milliseconds respectively.

SOUND CHECK

Next evening's schedule called for a live sound check, with the "live" provided by the new house band headed by a local favorite named Jerry Max Lane. When the sound crew returned from dinner, the band was onstage and unpacked. Jerry Max, outfitted in a bright blue Billy Bob satin jacket and the obligatory Stetson, tapped on the mic as Sam took his position behind the Yamaha and Dave Coats, of Ambassador's crew, manned the Stevenson at side stage.

Jerry Max and the boys swung into an upbeat Willie Nelson number and were immediately overcome by a deafening howl which resulted in frantic dial twisting at the boards and steely-eyed glares from the band. Another attempt at Willie was followed by a unified leap back from the screeching monitors. The 27-foot cavity over the stage was trapping the monitor



Concert stage loudspeaker stack, designed by Disco Scene Consultants & Designers.

output, creating what was described as a "400 Hz tunnel effect." The next day all of the stages were carpeted, and an acoustical curtain of heavy burgundy velour was ordered, the pleats 9-inches deep, 3-inches apart, and hung in front of the rear stage wall.

"We had a real serious problem on stage with the monitors, Sam said. "Because the stage was done in three different sections for portability, all plywood with angle stress beams under it, it wasn't dampened enough. We needed to get some type of trap in the rear of the stage to keep everything from going back down in the monitors again. When we carpeted the stage we got another 6-to-8 dB of gain out of the system before feedback. By the time the curtain was up, the band members asked us to turn the monitors down."

The sound check was ended by doing two equalizations, one for playback and one for PA vocals—mostly determined by the characteristics of the music that would be played through the system.

"Sometimes a room may not characterize the frequencies as well as the ear does," Sam said. "Many times a person's ears are tuned to bumps in the response that are pleasing to him. In some cases, when the majority of the people listening to a system *like* that particular bump, the system should be adjusted to it."

The "bump" for live music was from 60-100 Hz, then the system was tuned flat to 4 kHz and dropped 6 dB-per-octave from there. Playback got a "bump" at 70-80 Hz, a reduction in mids from 400-4 kHz, another bump at 6-8 kHz, and then a 3 dB-per-octave roll.

The problem with the air conditioning duct was temporarily solved by removing one of the Boxer stacks from its riser and placing it directly on the floor, allowing some amount of clearance between the high frequency horn and the ductwork, effectively de-coupling the air conditioning system.

"The only thing that this system lacks is some kind of flanging or digital reverb on the vocals," Sam said. "And they'll probably add it before too long. The owners here are tops. They didn't cut any corners on the initial investment. They made this place a showplace—a showplace for Country and Western Music and Fort Worth and everybody who is involved in Billy Bob's Texas. And I think they're going to keep it that way. It sounds terrific from every seat in the house. We've got super bandwidth all over this place. I think they're going to be very happy."

They were. Two nights before opening week—a week that included performances by the Gatlin Brothers, Janie Fricke, Waylon Jennings and two nights of Willie Nelson—Billy Bob, who had been closely observing the proceedings for the past two weeks, appeared with a leather case of tape cassettes.

"Wanna try it?", Sam asked. Billy Bob was ready. Sam gave the 6-ft., 5-in. rancher an introduction to the PM2000 level controls and the tape machine. Billy Bob brought out a tape and snapped it into place.

"How far can I turn it up?", he wanted to know. "As far as you want," Sam said. "You can't hurt it." The opening bars of Jimi Hendrix' Purple Haze filled the room. "As far as I want?" "Further!"

Billy Bob pushed the gain up and took a leisurely stroll around the club, nodding in satisfaction as the crew stared open-mouthed. "Now that," he said, "that sounds good." ■

The following changes were made to the system two weeks after opening:

The existing stage monitors were replaced by eight larger ones built by Showco of Dallas, Texas, one of the early bidders on the system. Due to the great amount of noise generated by crowds around the long bar area at the second delay point, six Showco S4 cabinets were hung from the ceiling over the bar and added to the delay system. Stay tuned....

DR. CLAY BARCLAY

PZM; Theory and Practice of a New Microphone Design

Author Barclay offers us a close look at the theories behind, and the capabilities of, the PZMicrophone.

THE CROWN PZMicrophone™[1], besides being a very new product for recording and sound reinforcement, has brought with it the need to re-evaluate recording and reinforcement techniques that have become almost sacred. The PZM™ makes new miking techniques possible, and in some cases necessary.

In order to take full advantage of the capabilities of this new mike, it is important to understand just how it works, and how that process makes those new techniques work.

We need to start explaining the PZM by restating some well-known truths about sound, since they are important to understanding the Pressure Recording Process™[2].

SOUND

Sound, as we all know, is composed of waves of kinetic energy which acts on particles of matter, displacing them from their original, at-rest position. The movement of each particle is dependent on a number of factors such as the velocity of the wave at any instant in time (which is a measure of the loudness of the sound), the material of which the particle is a part (air, wall, diaphragm, etc.), and the angle at which the wave encounters the particle.

In the standard microphone design, this wave encounters a diaphragm (or ribbon) which has been specially constructed to enable it to respond to the kinetic energy of the sound wave. The sound wave and the transducer in the microphone are directly coupled. That direct coupling of wave and sensor, which has made microphony possible for one hundred years, also introduces several problems.

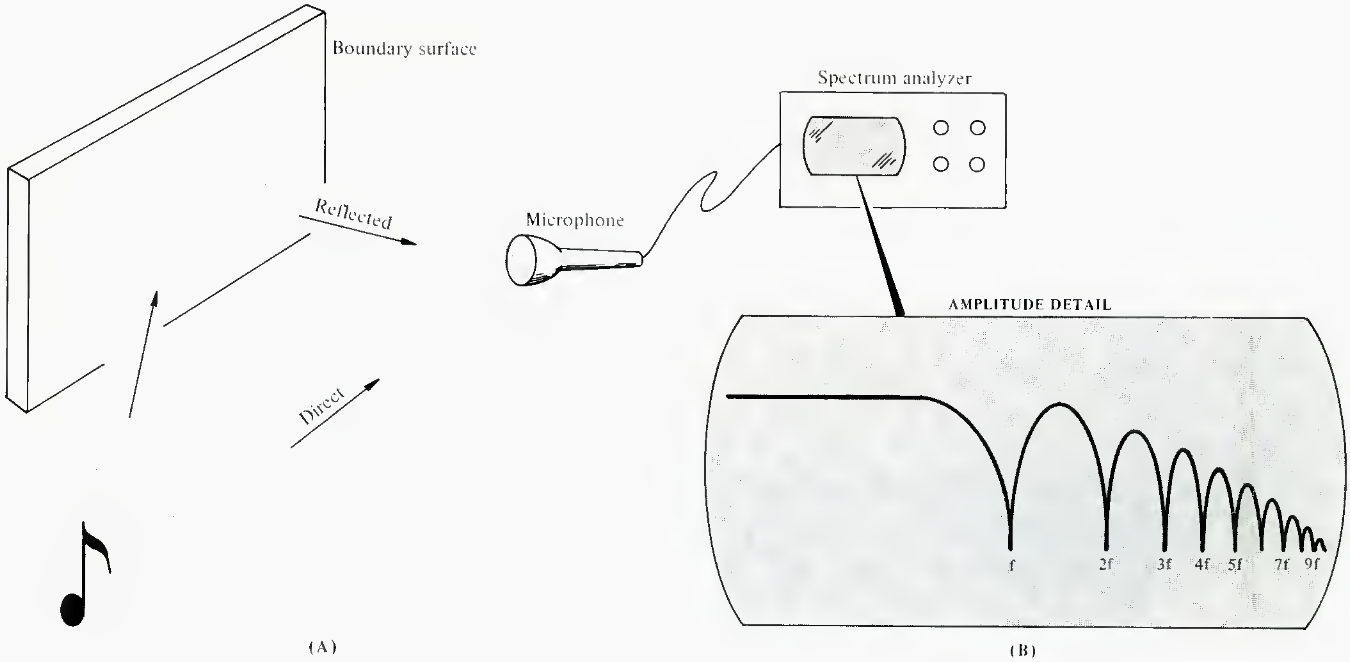
For one thing, the microphone is sensitive to the angle between its own axis and a line drawn from the sound source to the microphone. When this angle is something other than 0 or 180 degrees, the microphone will sense the wave differently, and will react less vigorously, creating attenuation of the sound. Or, it will react differently to different frequencies, thus creating distortion in the transduced signal.

Some variations in traditional microphone design have been developed that are actually more comfortable with off-axis sound, but until recently, no microphone design was comfortable with sound sources originating at *any* point around the mike. Indeed, most standard mikes will create easily audible differences in the transduced signal as the sound source is moved around the mike. Performers have had to learn to stay on the mark, or to keep the mike in one position relative to their lips, for as long as they are producing a signal.

Secondly, conventional designs are subject to comb filtering, which is a result of the combination of the direct signal and the signal as it is reflected to the mike from a primary boundary (floor, table, nearest wall, etc.). The different paths traveled by these two signals create a time difference which produces phase interference. Depending on the distance and angles, certain frequencies are reinforced, while others are attenuated, creating comb-like patterns in the frequency response curve.

The primary boundary is sometimes the microphone itself, or at least the part which covers and protects the diaphragm. Reflections from this will create comb-filtering effect which

Dr. Clay Barclay is the promotion manager for Crown International, Elkhart, Indiana.



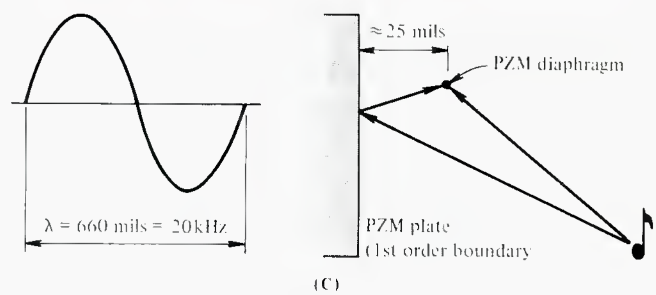


Figure 1. (A) As direct and reflected sounds are combined at the microphone, the spectrum analyzer display (B) shows the comb-filter effect. As the microphone is brought closer to the boundary, the null frequencies move beyond the audible range. (C) With the PZM diaphragm at about 25 mils from the boundary plate, any cancellation would occur far above the audio frequency bandwidth.

Figure 2. With the PZM capsule mounted very close to the boundary, the system response is in effect a hemisphere. Adding the vertical barrier reduces the response to a quarter-sphere, with a "polar pattern" as indicated by the shaded area on the polar response chart. For comparison, a conventional cardioid pattern is also shown.

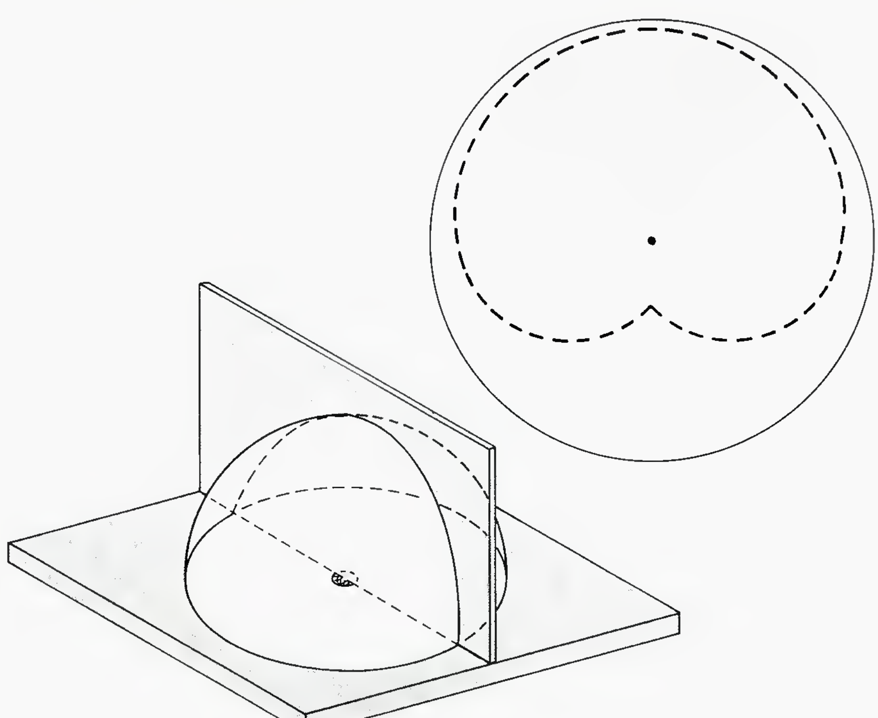
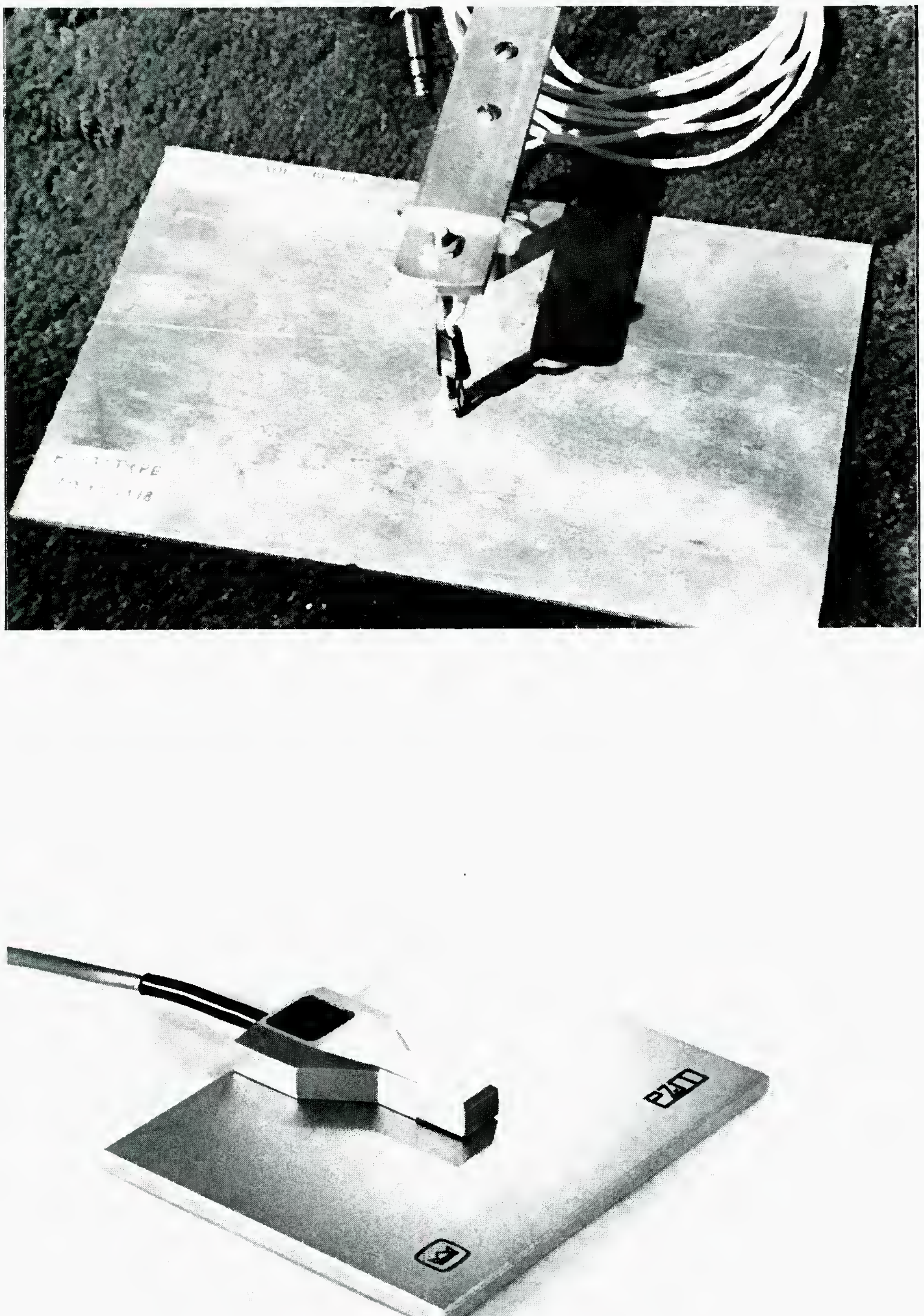


Figure 3. Two PZMs mounted on the lid of a grand piano.

Figure 4. PZMs old and new. (A) An early prototype constructed by Ken Wahrenbrock. (B) The PZM 6LP, manufactured by Crown International.



becomes a characteristic of that microphone, and is identified as part of its "personality."

Most of the problems associated with standard design microphones originate with these two phenomena, and the miking techniques which have been developed over the last hundred years have largely been attempts to compensate for them. They are responsible for colorations which certain mikes introduce, and also for creating the muddy, unlovely sounds so characteristic of "off-axis" problems. This is sort of strange, because our ears do not seem to be subject to these problems. Off-axis hearing does not introduce distortion into our hearing, but it does lend a sense of direction. Primary boundary reflections are not a problem for our heads, since we seem to sense the sound in a different manner than do the traditional microphones.

THE DEVELOPMENT OF THE PZM

The development of the PZMicrophone actually began with the question, "What is happening at the primary boundary?" Ron Wickersham and Ed Long were the first to seriously examine this phenomenon with a view towards putting their findings to work.

What they found was that, as the sound wave approaches a boundary, a significant change in the character of the wave takes place. At the boundary, the velocity of the wave becomes zero, and its kinetic energy is converted to a potential, which is expressed as pressure uniformly distributed on the boundary.

THE PRESSURE ZONE

This "pressure zone" is generally only a few mils deep, with its actual depth being frequency dependent. The minimum depth of the boundary has been determined by Long and Wickersham to be 1/6th of the wave length of the frequency.

Within the pressure zone, there is no direction of propagation, and a sensor would be unable to determine the direction of the source of the sound wave. For such a sensor, then, there would be no off-axis position.

The elimination of comb filtering in the pressure zone can be demonstrated by a simple experiment. In FIGURE 1, a time delay spectrometer displays the frequency response curve of the mike as it reacts to the sweep source. As the distance from the moveable plate to the microphone is changed, different null frequencies will be created. The scope display will change as the spatial relationships between the mike, the source and the plate are varied.

If the microphone diaphragm is brought to within a few mils of the boundary, the scope display will be flat, with no interference anomalies. However, comb filtering has not entirely disappeared. If the test setup were capable of reporting events in the ultrasonic range, we would find that the first interference null had simply moved above the audible range. It is, therefore, no longer of interest to the recording engineer, and within the pressure zone, direct and reflected audio-frequency signals now act to reinforce each other.

In their early experiments, Long and Wickersham attempted to take advantage of the boundary phenomenon by flush mounting a conventional mike so that the diaphragm would form part of the boundary, with the mike facing out from the boundary. This particular solution did generate some improvement in the sound, but the configuration was found to still be subject to off-axis differences.

The PZMicrophone, as finally developed by Ken Wahrenbrock and as now manufactured by Crown, represents an ideal configuration to take advantage of the physics of the pressure zone. In the PZM, a small precision-calibrated pressure capsule is mounted within the pressure zone, but not in contact with the boundary. It is also mounted in such a way that it is shielded from the direct signal. The PZM capsule is thus only indirectly coupled with the signal, since it is sensing only the pressure changes created within the pressure zone by the sound waves.

Figure 5. At Golden West College in Huntington Beach, California, Evan Williams illustrates several PZM placements.

THE BENEFITS OF THE PZM DESIGN

In the opinion of its developers, the PZM system offers several advantages over traditional design. First, there is no axis to the mike. Indeed, if you were to carefully move a sound source over the surface of an imaginary hemisphere with the PZM at the locus, a signal analyzer tracking the PZM output would be unable to detect any motion of the source. This is indicative of the fact that, within the pressure zone, there is no direction of propagation and the pressure is uniformly distributed. As long as the PZMicrophone can "see" the sound source, it will accurately pick up the sound signal.

Secondly, the PZM introduces no comb filters into the signal, and is completely free of any phase anomalies from the primary boundary. We stress "primary," since phase anomalies resulting from secondary boundaries will still be detectable. This is desirable, since it is precisely these anomalies in the recorded sound which provide the spatial information needed to recreate the ambience of the space in which the original sound was created.

Phase-interference from secondary boundaries such as walls or ceilings help us to understand where the musicians were when they created our music. Without those anomalies, the sound would seem to be coming from an anechoic chamber, which makes for lifeless listening.

A third advantage is that miking in higher SPL environments is possible, since the sensing mechanism used in the PZM is not a kinetic device. The pressure capsule in the PZM is capable of accurately detecting sound signals in a 150 dB level.

The sound detected by the PZMicrophone has the same spectral balance as the sound which is detected by our own ears. Differences generated by the time delays between direct and reflected sound are compensated in exactly the same way in which our ears hear the sound. Zero-degree incidence response and random-energy-response are perceived as equivalent.

Finally, the reach of the PZM mike is significantly better than a standard mike in an equivalent position. Whispered conversations thirty feet away have been clearly reproduced by PZM mikes. As the mike is moved back from the source of the sound, the frequency anomalies which are regarded as usual in standard microphone designs do not happen in PZM.

How has this happened? The simple answer is that the PZM design faithfully follows the concepts embodied in the Pressure Recording Process. Not so apparent are some of the more significant engineering aspects of the PZMicrophone. The PZM system is a unique configuration of sound detector and transducer, functioning in a manner which may not be immediately obvious. In fact, a clearly evident problem to early PZM users was that it needed new "rules" for microphone placement in order to take best advantage of its unique characteristics.

The PZM is the first microphone in which a precision-calibrated pressure capsule has been placed in an intimate relationship with a major boundary. It had been perceived by those early users of the PZM that its placement had to be consonant with an analysis of the position of the major boundary surface. Although the mike itself contains a boundary plate which is significant in improving its spectral characteristics, it has been shown in many applications that the performance of the mike can be enhanced by extending that boundary. This may be done by simply placing the mike on a larger surface (floor, wall, table, etc.).

THE DIFFERENCES IN PZM MICROPHONY

What are the differences in PZM microphony? One sound contractor with considerable PZM experience has suggested that the best technique for determining an optimum positioning is to act as if the PZMs were your ears: simply move around the sound source until you find the "sweet spot," and place the mike there. To many sound engineers who for years have been schooled in different techniques of mike placement, this sounds so simple as to amount to heresy. But it works. Yet it is difficult to accept the fact that the PZM does detect and reproduce sound in much the same way as our ears do, and that therefore we would do well to treat them as another pair of ears when working out our mike placement charts.

One highly experienced engineer, responsible for mike placement for an instrumental group, was reminded of the early days of microphony when entire orchestras were miked with one transducer on a high stand. "We used to do it all the time," he said, "so why do we find it so unusual to be able to do it with the latest in mike technology?"

Another major consideration in PZM placement is for the recording or reinforcement engineer to ask; what *is* the major boundary? Is it the floor? The wall? Is it a table top? Or perhaps, the ceiling? In each case, the answer will be determined by the knd of sound which the engineer is attempting to reproduce or reinforce.

For example, for small combos or stage productions, the major boundary could very well be the floor. A great deal of experience with PZM mikes indicates that such events can best be picked up by putting PZM mikes on the floor in front of the performers. On the other hand, the wall could be the best major boundary for solo vocalists or for drum kits which have been isolated by goboes. Or the ceiling, possibly represented by suspended panels, could be the best way to mike orchestras or large choirs with PZM.

The special characteristics of the PZM also make it possible to use them in some unique ways, such as inserted inside certain instruments. String bass or kick drums are ideal situations for such use of the PZM, and it has proven to be ideal for miking pianos, even where a leakage problem from nearby instruments has required the piano to be closed and covered with heavy blankets.

In conference settings, whether around a large table, or wall-mounted to pick up audience remarks, the PZM reach has proven most helpful to engineers. Lavalier versions, especially in redundant designs, have been especially valuable in broadcasting because of their low microphones, and almost total reliability.

It has proven most rewarding to men like Ed Long, Ron Wickersham and Ken Wahrenbrock to observe the successful application of the Pressure Recording Process. Engineers and listeners may well add their own approval. ■

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FOOTNOTES

1. PZM, PZMicrophone and Pressure Zone Microphone are trademarks of Crown International, Inc.

2. PRP and Pressure Recording Process are trademarks of E. M. Long Associates.

JOHN M. EARGLE

Stereo Microphone Techniques

The following discussion on various ensembles of microphones is excerpted from the author's forthcoming Microphone Handbook.

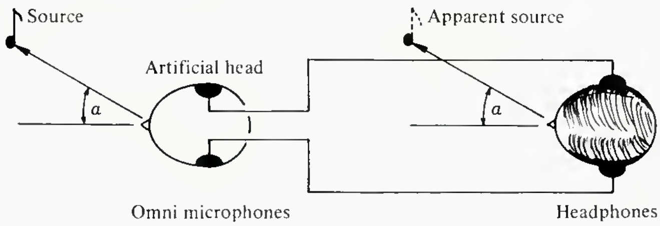
THE EARLIEST EXPERIMENTS in spatial sound reproduction date from the late 1800s, when telephone receivers and earphones were used in conveying a sense of auditory perspective through the use of an *artificial head*. The technique is shown in FIGURE 1. Here, microphones are placed inside the artificial head at ear position. At some remote location, a listener wears headphones, each corresponding to one of the microphones. In this method of sound transmission, referred to as *binaural sound*, the ears of the listener are effectively transported to the performance environment itself. Despite the shortcomings of those earliest transducers, it is reported that this system conveyed a remarkable sense of realism.

COINCIDENT MICROPHONE TECHNIQUE

Stereophonic sound refers to the pick-up of sound with an array of at least two microphones and subsequent replay over a pair of loudspeakers in such a manner as to convey some sense of the original left-right, fore-aft spatial relationships of the original performance. The earliest experiments in stereo were probably carried out by Alan Dower Blumlein in England in the early thirties. Those experiments made use of what has become generally known as the *coincident* microphone technique, an approach where the microphones in the pick-up array are clustered closely together.

Just a few years later, Bell Telephone Laboratories in the United States embarked upon an ambitious program which incorporated auditory perspective and multi-channel sound transmission, developed along with significant improvements in amplifiers, loudspeakers and microphones.

Figure 1. By means of microphones placed in an artificial head, directional cues may be transmitted to a listener wearing headphones.



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The art remained fairly constant for the next 20 years. Only during the post-World War II period did the magnetic tape recorder provide the impetus for the record industry to experiment with stereo. Recorded tapes were introduced in the early fifties, and the stereo disc was introduced in the late fifties. That decade saw a rapid expansion of both the technical and esthetic bases of stereo recording, and the art was extended beyond its role of merely documenting acoustical events, into one of creating new and original sounds in the studio.

In recent years, the introduction of 16- and 24-track recorders has broadened that base further. But too often, the simple lessons of Blumlein have been forgotten, and so it is with a review of his work that we begin our study.

BLUMLEIN'S TECHNIQUES

One of Blumlein's basic stereophonic microphone arrays is shown in FIGURE 2. A pair of crossed bi-directional microphones is oriented so that the axis common to the two positive lobes is aimed toward the center of an ensemble of performers. Because of the precise nature of the cosine polar response of the two bi-directional microphones, a sound source at position 1 will be picked up only by the left-oriented microphone. A similar relationship occurs with the right microphone and a sound source at position 2. For a sound source at position 3, there will be equal response at the two microphones, and the sound source will appear in the playback loudspeaker array to be directly in the center. Reverberant energy, since it comes from all directions, will be reproduced by the pair of microphones as coming from the space between the loudspeakers quite evenly.

In this way, a pair of bi-directional microphones crossed at an angle of 90 degrees behaves exactly like a panoramic potentiometer (panpot). The panpot is a signal processing device which routes a single input into two outputs so that the apparent position between loudspeakers becomes a function of the physical position of the panpot itself. The gradual cross-fading of the input between one output and the other provides a smooth transition of the reproduced image between the two loudspeakers in the playback array.

There are no interference problems when the outputs of both microphones are mixed. Since they are coincident—occupying virtually the same position—their summation represents an accurate monophonic signal.

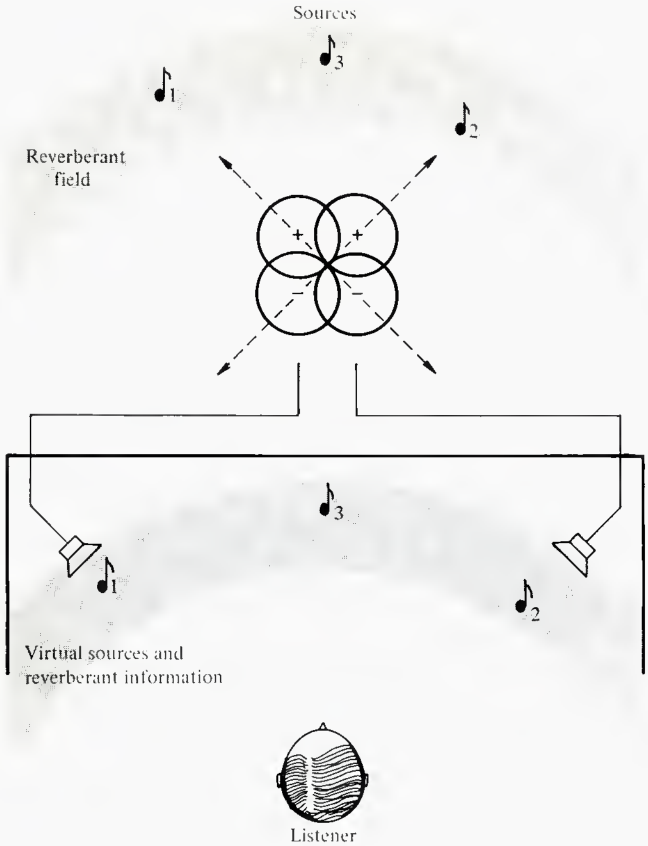


Figure 2. The Blumlein stereophonic technique. Crossed figure-8 microphones respond equally to sounds arriving from all directions.

This property of the crossed Blumlein-pair leads us to the concept of *sum* and *difference* transmission of the two signals, as shown in FIGURE 3. Through the use of what are called *matrixing* transformers, the left and right microphone outputs are added and subtracted to derive sum and difference signals. An additional pair of transformers undoes the process, and the left and right signals are recovered.

The process shown in FIGURE 3 forms the basis of stereophonic FM broadcasting, wherein a mono receiver picks up only the sum signal. The same condition exists in the playback of a stereophonic disc with a monophonic cartridge responding only to the lateral, or sum, signal.

Fortunately, those recording engineers wishing to make use of Blumlein's technique have access to so-called *stereo microphones*. These are available from a number of manufacturers, and they contain within one housing a pair of capacitor elements which are physically, as well as electrically, adjustable. The engineer can adjust both the precise polar response of either element, as well as the angular relationship between them.

VARIATIONS ON BLUMLEIN: THE M-S TECHNIQUE

The M-S (Middle-Side) technique uses a single forward-oriented microphone (M) as the sum signal, and a bi-directional microphone oriented at 90 degrees (S) as the difference signal. In Blumlein's early work, both microphones were bi-directional; today it is common practice to use a uni-directional (or even an omni-directional) microphone for the sum (M) signal.

When the outputs of the M-S pair are resolved by means of a sum-and-difference matrixing transformer, the result is a response equivalent to a pair of left-right oriented patterns, each of which resembles a super-cardioid response. No input matrix is required, since the M-S pair performs this function itself.

Further elaborations of the M-S technique are *width* and *position* controls. The width control acts by varying the amount of difference signal in the program.

As the difference signal is attenuated, the reproduced program will appear to the listener to be more monophonic, or center-oriented. When the difference signal predominates, its out-of-phase components effectively broaden the signal. The effect of the position control is to shift the entire reproduced array from left to right, as required by the program requirements at hand.

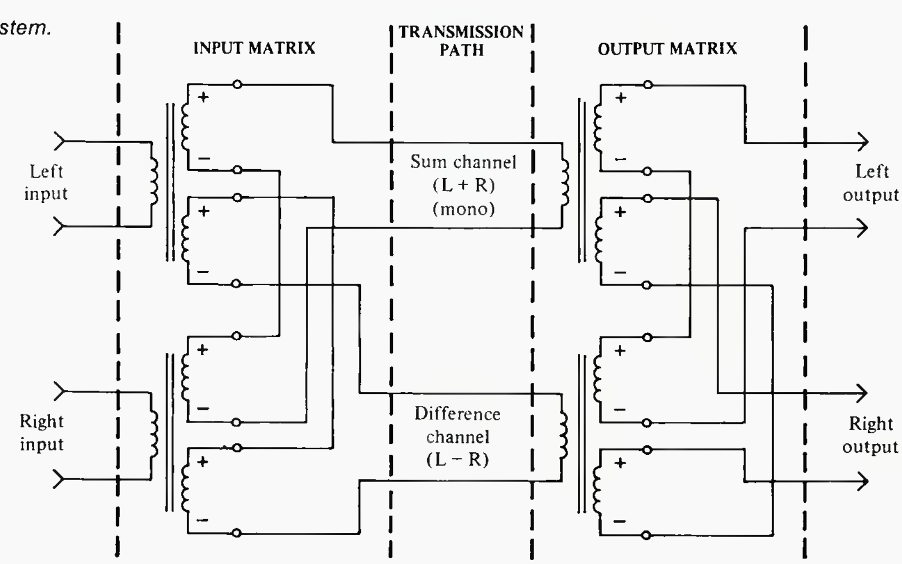
Typical usage of multiple M-S pairs of microphones is shown in FIGURE 4. Here, the main pickup pair is centered and its width adjusted to occupy the entire stereophonic playback stage. Sectional pairs would be positioned as required, and their width controls set for the appropriate effect in the overall picture.

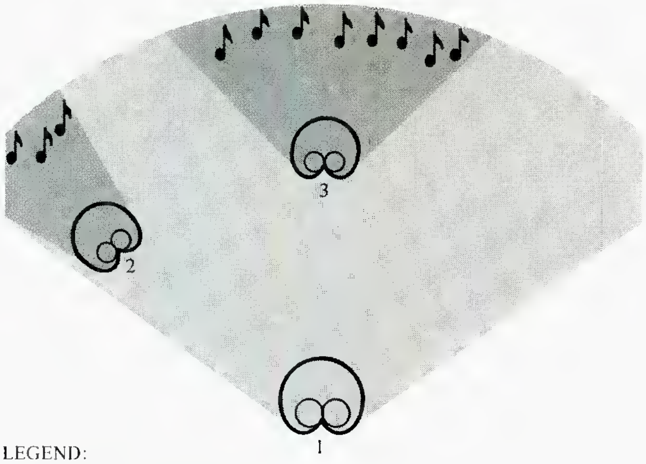
A pair of sum-and-difference microphones resolved into left-right signals is often referred to as an *X-Y* pair. FIGURE 5 shows equivalent X-Y pairs forms for a variety of sum-and-difference combinations.

SUBJECTIVE CHARACTERISTICS OF COINCIDENT TECHNIQUES

The chief virtue of any coincident technique is the absolute integrity of the panning or positioning function which it provides; particular instruments seem to originate in the playback loudspeaker array at their proper points. And as we

Figure 3. A sum-and-difference matrix system.





LEGEND:
1 = Main pickup: Wide spacing; center position
2 = Sectional pickup: Narrow spacing; left position
3 = Sectional pickup: Moderate spacing; center position

Figure 4. A typical application of multiple M-S microphones.

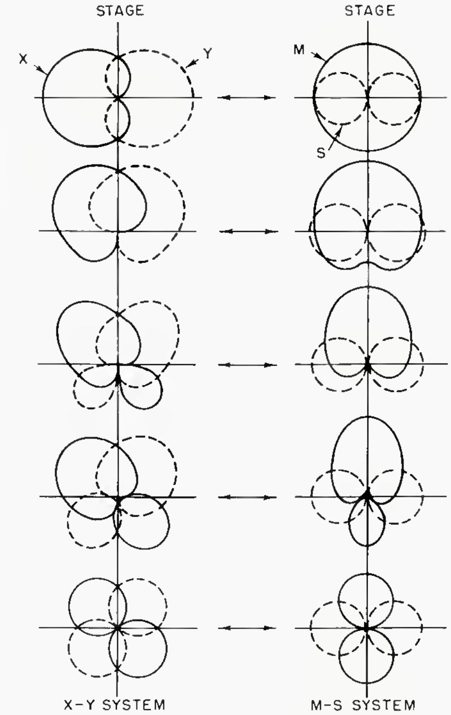


Figure 5. Various combinations of M and S microphones will produce a variety of X-Y patterns.

have stated, reverberant information seems to be evenly spread between the two playback loudspeakers.

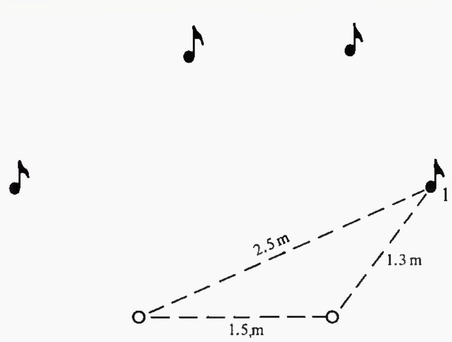
Yet in spite of this high degree of spatial accuracy, coincident stereo microphone techniques are not widely used today. One reason is that the approach is highly dependent upon precise distance relationships between the performers and their relation to the acoustical environment. The right combination is a rarity. Even when the right conditions do exist, many recording engineers and producers feel that the coincident technique lacks the degree of warmth and depth which even a slight spacing apart of the microphones provides.

SPACED-APART TECHNIQUES

Experiments by the French National Broadcasting Organization (ORTF-*Office de Radio diffusion-Television Francaise),* have led to an X-Y microphone array in which the capsules are spaced 17 cm apart, and at an angle of 110 degrees. This technique may be best described as *quasi-coincident;* at low frequencies (long wave lengths) the two microphones effectively occupy the same location. At frequencies above about 500 Hz, the microphone array responds to time delays due to the finite spacing of the pair: the listener senses not only these time-delay cues, but also the intensity difference, which is primarily a function of each microphone's polar response. Most listeners sense this as a feeling of space and "airiness" in the reproduced sound. A similar technique has been adopted by the Dutch Broadcasting Organization (NOS-Nederlandsche Omroep Stichting), making use of a pair of X-Y cardioids spaced 30 cm apart at an angle of 90 degrees.

As we have observed in the discussion of coincident microphones, a pair of directional microphones can pick up stereo information based upon intensity relationships alone. Of course, a coincident pair of omni-directional microphones would pick up no stereo information; the outputs of both microphones would be virtually identical. In order to work satisfactorily, omni-directional microphones must be spaced apart by some distance, usually suggested by the musical resources. In FIGURE 6, we show two such spacings: a typical setup for recording a chamber group is shown, as well as a typical setup for orchestral recording. In both examples, the

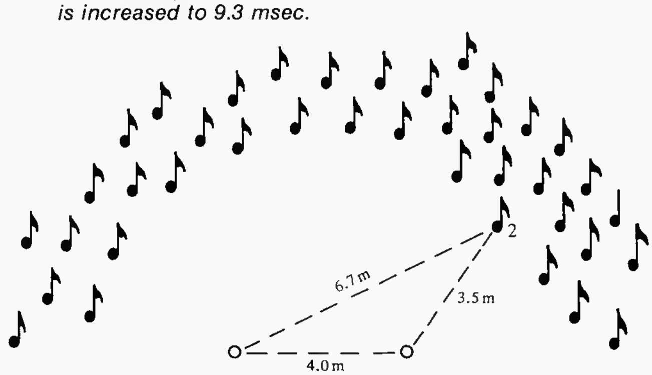
Figure 6. Delay and amplitude cues at a pair of microphones. (A) For a small ensemble, the sound at the left-hand microphone is attenuated by 5.7 dB, and arrives after a delay of 3.5 msec, as compared to the right-hand microphone. (B) For a larger ensemble, the delay is increased to 9.3 msec.



Inverse square ratio = $\left(\frac{1.3}{2.5}\right)^2 = 0.27$ (5.7 dB)

Time delay = $\frac{(2.5-1.3)}{344} = 3.5$ msec

(A) SMALL ENSEMBLE



Inverse square ratio = $\left(\frac{3.5}{6.7}\right)^2 = 0.27$ (5.7 dB)

Time delay = $\frac{(6.7-3.5)}{344} = 9.3$ msec

(B) LARGE ENSEMBLE

microphone pair is shown responding to a specific instrument in the ensemble. In each case, the distance ratio—and hence the amplitude ratio—is the same at the microphones. However, because of the wider spacing in the orchestral setup, the time delay cues will be greater, thus causing the sound source to be localized farther toward the right loudspeaker.

For wider spacings, there is a tendency for the elements of the ensemble to cluster at the two stereo loudspeakers. Where the spacing exceeds, say three or four meters, many engineers will add a center microphone to the array, panning its output equally into the two stereo signals. This microphone tends to fill in the "hole in the middle" and stabilize localization in the playback array.

USE OF ACCENT MICROPHONES

For the most part, commercial recordings of classical music make use of many microphones. Usually, the overall pickup is provided by a spaced-apart left-center-right group, or by a coincident or quasi-coincident pair. So-called accent microphones are then used sparingly to highlight individual sections or instrumental soloists. The various accent microphones are panned into the stereo array at positions approximating their actual location in the orchestra.

A particular problem with the use of accent microphones arises from the fact that their response *anticipates* in time the response of the main pickup array. In many typical setups, the shorter delay path from the accent microphones is not taken into account, and even a small amount of an early signal from an accent microphone can disturb natural perspectives. Heavy-handed usage of too many accent microphones has a tendency to render the perceived sound stage flat and two-dimensional, since it can swamp out the delicate balance of direct and early reflections picked up by the main array.

Figure 7. Stereo microphones. (A) Schoeps CMTS 301. (B) Neumann SM 69. (C) With a retail price of $135, Sony's ECM-949T brings the M-S sysem within reach of hobbyists, as well as the studio on a budget.

One way to alleviate the problem is to delay the accent microphones by an amount such that the sound reaching the listener from those microphones arrrives no sooner than that from the main array, or even a few milliseconds later. When this is done, the Haas effect no longer brings the signals from the accent microphones to the fore, and the balance engineer experiences less difficulty in establishing a pleasing musical balance.

SUBJECTIVE EVALUATION OF SPACED-APART MICROPHONE TECHNIQUES

Just as some engineers are avid devotees of coincident or quasi-coincident techniques in stereo microphone placement, others favor the spaced-apart approach. As a general observation, the coincident approach preserves geometric relationships in the performing group better than a spaced-apart array does; the listener is better able to pinpoint the positions of individual instruments. By comparison, the spaced-apart approach often conveys a heightened sense of the dimension and characteristics of the recording environment itself. Many listeners favor what they call a "warmer," less analytical, sound through the use of spaced-apart microphones.

In ideal environments, there is little doubt that coincident or quasi-coincident techniques offer a high degree of realism. In less-than-ideal environments, the spaced-apart technique often helps, through exaggeration of time delay cues, to create a sense of larger space.

Finally, we note that the spaced-apart approach results in a somewhat wider listening area than the coincident techniques do, due to presence of time delay cues in the recording process which counteract the amplitude offset which occurs when a listener moves off-axis. ■

JOHN M. WORAM

The Mathematics of the Microphone

Included in this article is everything *you've ever wanted (and some things you probably didn't want) to know about the mathematics of the microphone.*

WHEN THE CONVERSATION turns to microphones, most recording engineers are reasonably comfortable with terms such as cardioid, figure-8 and omni-directional. However, when some designer-types start expounding on the fine points of *limacon* responses, or cosine elements, most of us tune out and move on to more important things, like trying to get a better drum sound.

To get that better sound, most of us probably reach for a cardioid microphone, though some prefer omnis for close-up placements, thereby avoiding the problems of proximity effect. Very few think about using a figure-8. And, as we go about our selection process, it's almost a sure bet that none of us give much thought at all to the mathematics of the microphone.

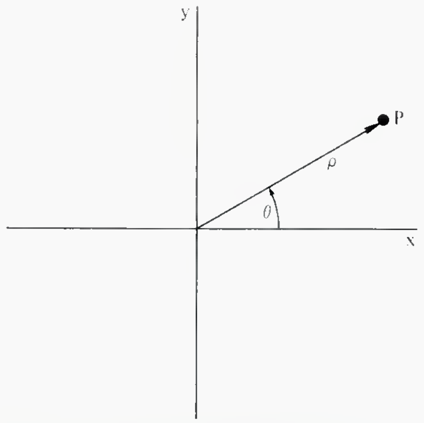
After all, why should we? Learning that the polar equation for a *limacon*—oops, cardioid—microphone may be written as $\rho = .5 + .5\ cos\theta$ is not exactly the sort of information that is desperately needed to save the session. (ρ = Greek "rho.")

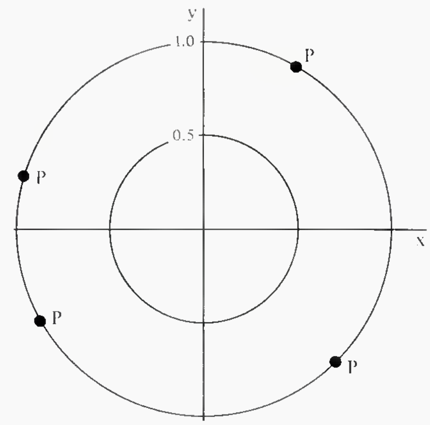
Or is it? In most cases, probably not. But every now and then, you might wish that you could "create" a polar pattern that would help out in an unexpected situation. An example might be a figure-8 pattern: suddenly you need one on a remote, and all you have along is a collection of cardioids and omnis. What to do?

This one's easy. A pair of back-to-back cardioids will do it, provided you get the polarity of one of them reversed. (In effect, that's how the figure-8 pattern is derived in many multi-pattern capacitor microphones.) This particular combination may be recognized almost intuitively by anyone who goes to the trouble of drawing (or at least, visualizing) the two cardioid patterns. However, what happens when the same two cardioids are placed at some other angle?

Provided the polarity of one microphone is reversed, the resultant pattern is *always* a figure-8, regardless of the angle between them. But it takes a lot of visualizing, or many hours at the drawing board, to see why this is so. And, if you're very busy (or, trying to be very busy) doing sessions, you probably have neither the time nor the inclination for this sort of thing. However, if you do have the time and the inclination to investigate some of the theory behind the practice, a little "mike math" may be of interest, and possibly, of some practical value. Once some of the basics are better understood, it becomes quite easy to predict what will happen when two microphone outputs are combined, intentionally or otherwise. But first...

John Woram is the editor of **db** *Magazine.*





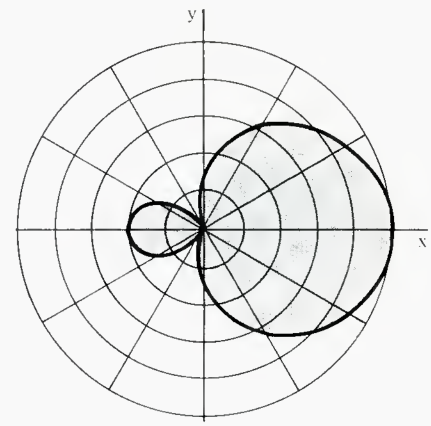


Figure 1. In the polar coordinate system, the point, P, is at a distance, ρ, and angle, θ, from the origin.

Figure 2. An omni-directional polar response. The microphone is equally sensitive to sounds at any of the points, P.

Figure 3. The polar response for a microphone with a sensitivity of 0.3 × 0.7cosθ. The rear lobe (to the left of the Y axis) has a polarity reversal.

POLAR PATTERNS

Everybody looks at the polar pattern as a pictorial representation of a microphone's sensitivity at various angles. For example, a certain polar pattern may show us that a microphone is down by 6 dB at 90 degrees, and so on. The mathematicians in the crowd may note that the cardioid pattern resembles a *limacon of Pascal*—a well-known (at least, to Pascal) mathematical figure. In fact, all polar patterns may be derived mathematically using the same general equation, which we'll get to in a minute.

First, let's go back to the high school math book, and review polar coordinates. The polar coordinate system is merely a convenient way of representing any point, P, on a graph. The point is at a certain distance, ρ, from the origin (i.e., where $X = 0$ and $Y = 0$), and is at a certain angle from the X axis, as shown in FIGURE 1.

Now, if we plot a series of such points at various angles from 0 to 360 degrees, we shall have our polar pattern. As an easy example, if the distance, ρ, remains constant (say, $\rho = 1$) at all angles, our polar pattern will be a circle with a radius of 1. If we assume the radius represents the relative sensitivity of a microphone, then our circle is the polar response of a microphone whose sensitivity, or output voltage, remains the same, regardless of the angle from which the sound arrives.

In FIGURE 2, we see such a polar response. Of course, it's for the familiar omni-directional mike, and it's clear that the microphone responds equally to sounds arriving from any of the points, P, along the circumference labeled 1. However, if any of those Ps had been drawn on the smaller (dashed-line) circle labeled 0.5, the microphone's output would be understood to be one-half that of the omni-directional type.

THE POLAR EQUATION

As noted earlier, a single equation is used to draw the polar pattern of any microphone. The equation is
$\rho = A + B\,cos\theta$,

where

ρ = the microphone's sensitivity, at any angle, .
A & B = constants which determine the shape of the polar pattern, and
θ = the angle from which the sound arrives (θ = Greek *"theta."*)

Before listing the specific values for A and B, let's assume that all microphones should have the same sensitivity when the sound arrives from straight ahead. In other words, for any microphone, the on-axis ($\theta = 0$) sensitivity is $\rho = A + B\,cos0 = A + B = 1$.

From the above, we see that the sum of A and B must always be 1, in order to satisfy our requirement that any microphone type will have the same on-axis sensitivity.

OMNI-DIRECTIONAL PATTERN

Now, let's plug in some values for A and B and see what happens. We can start simply, with $A = 1$, $B = 0$. In this case, $\rho = 1$ at all angles, since the $B\,cos\theta$ element of the equation is always zero. In short, we have our omni-directional pattern: a perfect circle with a radius of 1.

BI-DIRECTIONAL PATTERN

If $A = 0$, then $B = 1$, and the equation becomes $\rho = cos\theta$. We have just created the cosine element, and when we plot the equation for various values of θ, we see the familiar bi-directional "figure-8" pattern.

UNI—DIRECTIONAL PATTERN

In between these two extremes, A and B may take on a multitude of values. Here, let's take just one: $A = 0.5$, $B = 0.5$. When $\rho = 0.5 + 0.5\,cos\theta$, mathematicians call the pattern a *limacon of Pascal.* More pragmatic types call it a cardioid polar pattern.

COMBINING POLAR PATTERNS

Now, let's ask the question again: what happens when two cardioid microphones, M_1 and M_2, are combined? Well, it depends on the angle between them. In any case, $M_1 = 0.5 + 0.5\,cos\theta_1$ *and* $M_2 = 0.5 + 0.5\,cos\theta_2$. If the mics are pointing in the same direction, then, of course, $\theta_1 = \theta_2$, and we have

$$\begin{array}{l} M_1 \quad 0.5 + 0.5cos\theta \\ +M_2 + \underline{(0.5 + 0.5cos\theta)} \\ M_t = 1 + 1cos\theta \end{array}$$

The resultant polar pattern has twice the sensitivity ($A = 1$, $B = 1$) of a single cardioid pattern, which is just what we'd expect when the outputs of two cardioids are combined.

But now, suppose the mics are placed back-to-back. $\theta_2 = \theta_1 + 180°$, which means that at any angle, $cos\theta_2 = -cos\theta_1$. Therefore,

$$\begin{array}{l} M_1 \quad 0.5 + 0.5\,cos\ _1 \\ M_2 + \underline{(0.5 + 0.5\,cos[\theta_1 + 180°])} \\ M_t = \quad 1 + 0. \end{array}$$

Once again, we have an omni-directional polar pattern. Finally (!), let's subtract the outputs of the two cardioid mics by reversing the polarity of M_2. For the moment, we won't concern ourselves with whether the microphones are back-to-back, pointed in the same direction, or at some angle between 0 to 180 degrees. In any case,

$$\begin{array}{rl} M_1 & 0.5 + 0.5\,cos\theta_1 \\ -M_2 & -(0.5 + 0.5\,cos\theta_2) \\ \hline M_t = & 0 + 0.5\,[cos\theta_1 - cos\theta_2]. \end{array}$$

Note that we are left with simply a cosine element, since A–A=0. In other words, the resultant polar pattern is *always* a figure-8 (i.e., a "cosine" pattern), regardless of the angle between the microphones. The specific angle determines the size of the figure-8 pattern. For example, when the angle is 180 degrees, then the on-axis sensitivity is $\rho = 0.5[cos0 - cos180] = 0.5[1 - (-1)] = 1$. As the angle diminishes, the resultant output also diminishes, until at zero degrees the output of our figure-8 microphone combination is zero.

At first glance, this "mike math" may appear to be more trouble than it's worth. However, it's really not that tedious, since in most cases we needn't even bother ourselves with the actual values of $cos\theta$. Rather, we should be on the lookout for the values of A and B that result when two patterns are combined. Generally speaking, as A approaches 1, the polar pattern approaches omni-directional. As A approaches 0, the pattern approaches bi-directional. And when A = B, we have a perfect cardioid pattern. (By the way, what is the pattern when the mics are at 90 degrees, and there is *no* polarity reversal? See part II [next month] for the answer.)

HYPER- AND SUPER-CARDIOID PATTERNS

The so-called hyper-cardioid pattern is midway between the cardioid and the figure-8 pattern. That is, A = 0.25 and B = 0.75. The super-cardioid pattern is midway between cardioid and hyper-cardioid, with A = 0.375 and B = 0.625. As always, A + B = 1.

If we were to observe a polar pattern making a smooth transition from omni, through the various cardioid patterns, to bi-directional, we would see the 180-degree point gradually move inwards towards the origin. At A = 0.5 and B = 0.5, the pattern would indicate a sensitivity of zero at 180 degrees; that is, infinite attenuation. (Needless to say, this is the theory; in practice, the output may be down by some 20 or so dB.)

As we continue beyond this point, our equation will yield negative values, once B is greater than A. For example, $\rho = 0.3 + 0.7\,cos180 = 0.3 + 0.7(-1) = -0.4$. Graphically, this is represented by the appearance of a rear lobe, as seen in Figure 3. The rear lobe gets progressively larger, until at A = 0 and B = 1, it is the mirror image of the front lobe. Now, we have the two circles of the figure-pattern. The negative values signify that a sound source picked up by the rear lobe will produce an output signal whose polarity is reversed, as compared to the same source picked up by the front lobe of this—or any other—microphone. This is why it's important to keep the back-end of a bi-directional microphone away from the front-end of other nearby microphones. If not, combining the outputs, either on-session or during the mixdown later on, will produce some strange cancellations.

RANDOM ENERGY EFFICIENCY

The random energy efficiency of a microphone may be expressed by the ratio

$REE = E_1/E_2$,

where

REE = random energy efficiency,

E_1 = total energy picked up by the microphone in a diffuse sound field, and

E_2 = total energy picked up by an omni-directional microphone in the same diffuse sound field.

The hyper-cardioid microphone offers the minimum random energy efficiency (REE = 0.25).

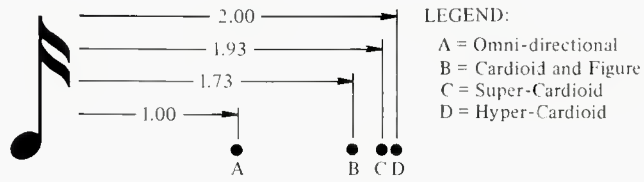


Figure 4. The distance factor indicates the relative working distance of various microphones.

DISTANCE FACTOR

The distance factor indicates the relative working distance of various microphones, as compared to an omni-directional mic (see Figure 4). At these distances, each microphone's signal-to-noise ratio is the same as that of the omni, assuming that the signal is on-axis, and the noise comes from the diffuse sound field. The distance factor is found from the equation

$DF = 1/\sqrt{REE}$.

Of course, the more directional the microphone, the greater the distance factor, which makes it a convenient index of the microphone's "directionality." It is interesting to note that the cardioid and figure-8 microphone have the same distance factor (DF = 1.73). This means that the cardioid is no more "directional" than the figure-8; in a diffuse sound field, these microphones simply pick up their energy from different areas.

HIGHER-ORDER MICROPHONES AND THEIR EQUATIONS

The microphones discussed so far are all known as "first-order" patterns, since there is only one cosine term in the equation. A second-order set of equations may be written by multiplying the first-order equation by $cos\theta$. Thus, the equation for a

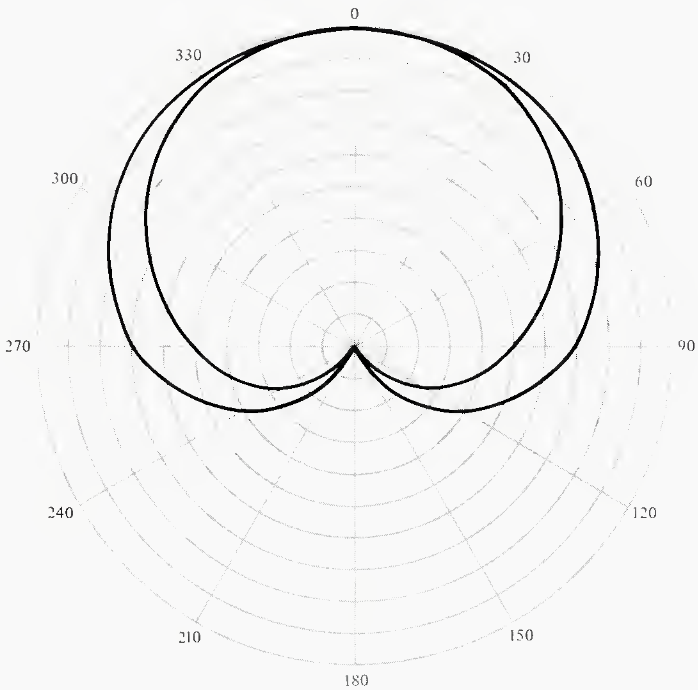


Figure 5. Two polar responses for the same microphone. The inner pattern is a graph of sensitivity; the outer one is of dB attenuation.

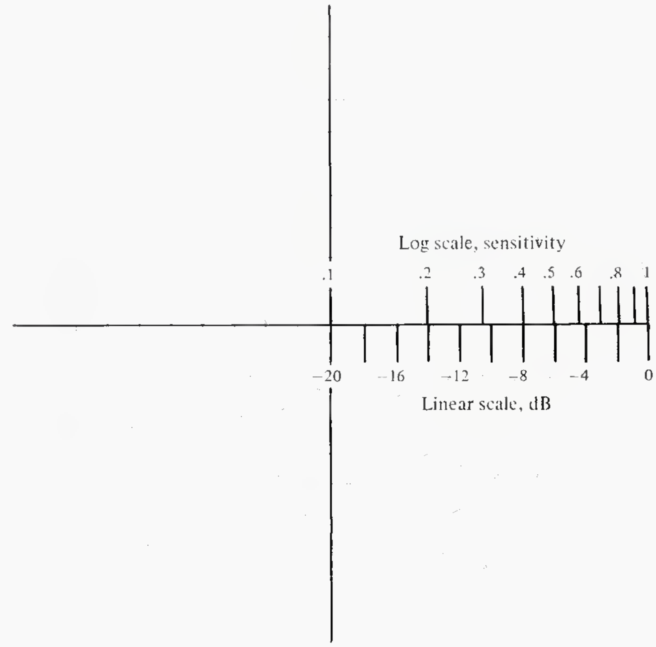


Figure 6. The concentric circles are drawn in 2 dB increments. The superimposed log scale may be used to graph sensitivity versus angle.

second-order polar response is $\rho = (A + B\cos\theta)(\cos\theta)$. The general equation for *any* microphone, of order, n, may be written as $\rho = (A + B\cos\theta)(\cos\theta)^{n-1}$. For first-order patterns, n = 1, and so $(\cos\theta)^{n-1} = 1$. In other words, we are left with simply, $\rho = A + B\cos\theta$, as before.

Most studio microphones are first-order types, although a few second-order patterns have recently been introduced.

A second-order cardioid microphone will have a much narrower "beam," thus making it—theoretically, at least—very attractive for its highly-directional capabilities. For example, a sixth-order cardioid microphone has a distance factor of 4.9. However, just as the higher-order equations get more complex, so do the practical design problems. For the moment at least, high-quality, high-order microphones are not commercially available.

POLAR EQUATION SUMMARY

A	B	REE	DF	Polar Response TYPE	SHAPE	Simplified Equation
1.0	0	1.0	1.0	omni-	circle	1
0.5	0.5	0.33	1.73	uni-	cardioid	n/a
0.375	0.625	0.268	1.93	uni-	super-	n/a
0.25	0.75	0.25	2.0	uni-	hyper-	n/a
0	1.0	0.33	1.73	bi-	figure-8	$\cos\theta$

Note that the calculated values of ρ, and the resultant polar patterns, represent the microphone's output sensitivity, and not the amount of attenuation in dB. To convert sensitivity to dB, simply use the equation

$N_{dB} = 20 \log \rho$.

For example: At 90 degrees, what is the attenuation of a cardioid microphone?

$\rho = 0.5 + 0.5 \cos 90$
$= 0.5 + 0 = 0.5$. Therefore,
$N_{dB} = 20 \log 0.5 = 20(-0.3) = -6$ dB.

SENSITIVITY VERSUS ATTENUATION

Polar responses are sometimes drawn according to the sensitivity values calculated as described above. However, the polar patterns on most spec sheets show microphone response in terms of attenuation in dB at various angles. As shown in FIGURE 5, the actual shape of the pattern will differ, depending on whether the pattern is either a graph of sensitivity or attenuation, versus angle of incidence.

When combining polar patterns graphically, make sure that the patterns are graphs of sensitivity, and *not* of dB attenuation. To draw a single pattern which may represent either value, FIGURE 6 may be used as an example. Note that the equidistant concentric circles represent increments of 2 dB, as shown by the linear horizontal scale.

Sensitivity values are given on the logarithmic scale printed immediately above the dB values. The dual scale may be used as a nomograph to calculate the output in either dB or sensitivity when two microphones are combined.

Example 1. What is the output in dB when microphones with sensitivities of 0.2 and 0.4 are combined?
0.2 + 0.4 = 0.6 = -4.4 dB.

Example 2. The outputs of two microphones are both 8 dB down. What is the combined output, in dB?
-8 dB = 0.4 sensitivity.
0.4 + 0.4 = 0.8 sensitivity.
0.8 sensitivity = -2 dB.

PROGRAMMING POLAR PATTERNS

A reasonably simple computer program will tabulate and print the sensitivity values for any polar pattern. For example, to calculate the values for a cardioid pattern:

```
100 FOR D = 0 TO 360 STEP 15
110 TH = .01745*D
120 S = .5 + .5*COS(TH)
130 PRINT D, S
140 NEXT D
150 END
```

```
10 T1 = 1: T2 = 10
20 HOME:VTAB 5
30 PRINT "        - - - POLAR RESPONSE - - - "
40 PRINT:PRINT:PRINT
50 PRINT "PROGRAM CALCULATES POLAR SENSITIVITY, IN"
60 PRINT "15-DEGREE INCREMENTS."
70 PRINT:PRINT
80 PRINT "IN THE EQUATION, A + B COS(TH),"
90 PRINT
100 PRINT "WHAT IS THE VALUE OF";
110 FLASH
120 PRINT " A ";
130 NORMAL
140 INPUT A
150 B = 1 - A
160 PRINT
170 PRINT "WHAT IS THE ORDER NUMBER? ";N
180 HOME:VTAB 5
190 PRINT "A = ";A,"B = ";B
200 PRINT
210 PRINT "ANGLE";
220 PRINT TAB(10)"SENS.";
230 PRINT TAB(20)"ANGLE";
240 PRINT TAB(30)"SENS."
250 PRINT
260 FOR D = 0 TO 360 STEP 15
270 TH = .01745*D
280 S = (A + B*COS(TH))*COS(TH)↑(N - 1)
290 S = INT(S*100 + .5)/100
300 IF D = 180 THEN T1 = 20:T2 = 30:VTAB 9
310 HTAB (T1)
320 PRINT D;
330 HTAB (T2)
340 PRINT S
350 NEXT D
360 END
```

Figure 7. POLAR RESPONSE program. This BASIC program calculates the polar sensitivity in 15-degree increments.

Line 100 selects angles in 15-degree increments, and 110 converts these into radians, as required by most computers. Line 120 does the calculation and 130 prints the angle, D, and the calculated sensitivity, S.

Since this program prints some 26 lines of data, the first few lines disappear off the top of the CRT screen as the last few lines are printed. (The actual number of lines displayed on the CRT depends on the particular computer. This, and the following programs, were written on an Apple II computer. Minor variations may be required for other systems.)

Some additional lines of instruction will allow the program to compute the sensitivity for any pattern, and keep all the values on the screen. In addition, the program may be expanded to calculate values for any order microphone. The complete program is given in FIGURE 7.

COMPUTER CALCULUS

Readers who get turned-on by calculus textbooks probably don't need the next program. However, for those who find the calculus about as much fun as an IRS audit, the following may help ease the pain.

The equation for calculating a microphone's random energy efficiency (REE) is

$$REE = 0.5\int_0^{\pi} f^2 \cos\theta \cdot \sin\theta \cdot \partial\theta .$$

In reasonably plain English, the value of ρ in the equation, $\rho = A + B\cos\theta$, is a function of the value of θ, which of course varies from 0 to 360 degrees. Therefore, the value of ρ varies as a *function* (f) of $\cos\theta$. In the REE equation above, the term $f^2\cos\theta$ instructs us to square this function. Next, we must multiply this by $\sin\theta$. We must do this repeatedly, at very small increments, $\partial\theta$.

```
10 HOME: VTAB 10
20 PRINT "     RANDOM ENERGY EFFICIENCY (REE)"
30 FOR T = 0 TO 1500: NEXT T
40 HOME: VTAB 5
50 PRINT "1. PROGRAM WILL CALCULATE REE FOR VALUES"
60 PRINT "   OF 'A' FROM 0 TO 1,  IN INCREMENTS OF"
70 PRINT "   0.05."
80 PRINT:PRINT
90 PRINT "                - - - OR - - -"
100 PRINT:PRINT
110 PRINT "2. PROGRAM WILL  CALCULATE  REE FOR  THE"
120 PRINT "   VALUE OF 'A' THAT YOU SELECT."
130 PRINT:PRINT
140 PRINT "   TO BEGIN, PLEASE DEPRESS";
150 FLASH
160 PRINT " 1";
170 NORMAL
180 PRINT " OR";
190 FLASH
200 PRINT " 2."
210 NORMAL
220 INPUT K
230 HOME: VTAB 10
240 INPUT "WHAT ORDER MICROPHONE? ";N
250 PRINT:PRINT
260 IF K = 1 THEN 340
270 PRINT "IN THE EQUATION, A + B COS(TH),"
280 PRINT:PRINT
290 "WHAT IS THE VALUE OF";
300 FLASH
310 PRINT " 'A'";
320 NORMAL
330 INPUT A
340 HOME: VTAB 10
350 PRINT "A";
360 PRINT TAB(10)"B";
370 PRINT TAB(20)"REE"
380 IF K <> 1 THEN 400
390 FOR A = 0 TO 1 STEP .05
400 B = 1 - A
410 DEF FN F(TH) = (A + B*COS(TH))*(COS(TH))↑(N - 1)
420 FOR TH = 0 TO 3.14 STEP .01745
430 S = (FN F(TH))↑2*SIN(TH)*.01745
440 S = S + M
450 M = S
460 NEXT TH
470 RE = M/2
480 PRINT A;
490 PRINT TAB(10)B;
500 RE = INT(RE*1000 + .5)/1000
510 PRINT TAB(20)RE
520 IF K <> 1 THEN 550
530 M = 0
540 NEXT A
550 END
```

Figure 8. RANDOM ENERGY EFFICIENCY. This program calculates the random energy efficiency for various values of A in the equation, A + B cosθ. Or, it will calculate a single REE for a specific value of A.

How small? Well, the smaller the increment, the greater our accuracy will be. In the program given in FIGURE 8, we will make calculations in one-degree increments, although for all practical purposes, ten-degree increments would give a satisfactory answer that is accurate to two decimal places. However, as long as the computer is going to do all the real work, we may as well demand precision.

Returning to the equation, the term $0.5\int_0^{\pi}$ tells us we must make our calculations over the range from 0 to π radians (180 degrees), and then take the summation ($\int$) of these calculations and divide by 2 (i.e., $0.5\int$).

In other words, this innocuous looking equation directs us to calculate $(A + B\cos\theta)^2(\sin\theta)$ 180 times (!), add them up and divide by 2. Of course, mathematicians just love little chores like this, and those familiar with the calculus can whip out answers in no time. For the rest of us, the computer will tell us what we want to know in about 30 seconds, or in less than 10 seconds if ten-degree increments are used. The program given in FIGURE 8 will also calculate REE for values of A from 0 to 1, in increments of 0.05. This takes about ten minutes. ■

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db People/Places/Happenings

• Based on a supply contract concluded between **Studer International AG** Regensdorf/Switzerland and **Meltron** Maharashtra in India, an opening order for 300 professional tape machines was received in Regendsdorf, to be supplied in the course of this year. As the sole distributor in Europe and elsewhere overseas for the professional range of audio products of **Willy Studer** Regensdorf, Studer International will supply 565 console type and 515 reproduction-only-machines as well as 312 portable tape machines to India during the forthcoming 5 years. First-phase delivery covers complete equipment, for operation at **All India Radio (AIR)**, National Broadcasting of India.

• **John F. Phelan** has joined **Sony Corporation** as western regional manager, according to **Nick Morris,** general manager of Sony's Professional Audio Division. Mr. Phelan brings to his new position nine years experience in sales and management of professional audio products. Prior to joining Sony, he was vice-president of sales at **Filmways Audio Services, Inc.,** where he had previously served as general manager. Previous to that, he was manager of professional sound products for **Shure Brothers, Inc.**

• **HM Electronics, Inc.** has appointed **Robert W. Carr** as marketing manager. With over thirty years background in engineering and marketing functions at **Shure Bros., Inc.,** Carr will be responsible for the expansion of marketing and product planning activities at HME, according to President **H. Y. Miyahira. John F. Kenyon** has been promoted to assistant sales manager.

• **Tom Horton,** President of **Pentagon Industries, Inc.,** the world's leading manufacturer of cassette tape duplicators, announced that **Harry Sheaffer** has been appointed to the position of director of sales and customer service. Mr. Sheaffer joined Pentagon five years ago. Prior to that, Sheaffer was national service manager for **Simpson Electric Company.**

• **John Edward Pritchett,** president of **Quantum Audio Labs** in Glendale, CA, died of cancer on January 29, 1981. He was 41. Pritchett was the president of Quantum Audio Labs at the time of his death. Quantum Audio Labs is a manufacturer of professional mixing consoles for the recording and broadcast industries. Mr. Pritchett had a controlling interest in the company and had been with them since 1975. Prior to that he worked for **Fender Music Co.,** Anaheim, CA; **Altec Lansing,** Anaheim, CA; and **James B. Lansing Co.** in Northridge, CA.

• **Syn-Aud-Con** has announced the availability of subscriptions to their highly acclaimed "Newsletters" and "Tech Topics" to non-Syn-Aud-Con graduates for the first time. A non-graduate one year subscription is $50.00. The subscriber may apply the $50.00 to a Syn-Aud-Con Sound Engineering Seminar any time during the year of the subscription. The subscriber receives four quarterly mailings consisting of one Newsletter per mailing plus a minimum of two Tech Topics per mailing. To obtain information, write Don Davis, Synergetic Audio Concepts, P.O. Box 1115, San Juan Capistrano, CA 92693.

• **Bonneville Productions** upgraded its 24-track Studio C facility with the installation of **UREI** 813 time-aligned monitor speakers, each powered with 800 watts of **Crest** amplification. Studio C, the only 24-track studio in Salt Lake City, features **Ampex** MM1200 24-track and MM1100 16-track, ATR102 two-tracks and full **Dolby** and **dbx** noise reduction. Bonneville's Studio B has been remodeled to create a more acoustically pure environment and has added **Big Red** Monitors. Studio B's control room also has a new Ampex ATR-104 four-track recorder. The studio is equipped for looping (lip-sync) operation. Bonneville Productions' Media Studios A and B, and Music Studio C are the largest audio recording facility in Utah.

• **Michael Fusaro** was recently named chief engineer at **The Automatt** by studio owner/producer **David Rubinson.** Fusaro was assistant supervisor at **CBS'** San Francisco studios from 1971-78 and has worked in engineering and maintenance with both **Fantasy Studios** and **Coast Recording.**

• **CBS Records** has announced the introduction of the CX™ system, a major audio process which virtually eliminates surface noise from phonograph records and dramatically extends dynamic range. CX-encoded records can be played on conventional stereo equipment and will cost the same as standard records. Furthermore, CX releases can be immediately manufactured on conventional equipment in any quantity wherever records are now being pressed. Developed by the CBS Technology Center in Stamford, Connecticut, the CX process increases the dynamic range of discs by 20 db to nearly 85 db, comparable to that of an actual concert hall environment. The unique compatibility feature of the CX process allows CX records to be played today on any stereo system with the sound quality of conventional stereo releases. This is not true for records encoded by other noise reduction techniques which can be played only on systems equipped with the appropriate decoders. CBS Records has already begun to press CX releases, and the company is fully equipped to immediately manufacture any commercial quantity necessary. The first CX records will be released in the near future and the company intends to steadily increase its CX release output until all CBS releases are encoded as a matter of course. CBS Records' mastering facilities are being equipped with CX encoders, and the company is working closely with outside mastering studios to expedite the installation of CX encoders on an industry-wide basis. **UREI**, a leading manufacturer of recording studio equipment, has been licensed by CBS to manufacture professional CX encoders. CBS Records is making the CX technology available to the entire record industry under royalty-free licensing agreements with other record companies.

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