THE SOUND ENGINEERING MAGAZINE

OCTOBER 1980
$1.95

db

Coming Next Month

• In November, we take a look at some Good Engineering Practices related to the care and feeding of master tapes. The data comes to us from a recent SPARS survey of member studios.

Also—a look at the preparation of test tapes, a quick-reference to what's available from four test tape manufacturers, an introduction to Otari's multi-track tape recorders, a tutorial on RMS, and ways to correct digital tape errors.

All this in the November issue of **db—the Sound Engineering Magazine.**

About The Cover

• BT Express, a rhythm-and-blues group, is shown here testing the 3M Digital Mastering Systems at New York's Sound Ideas Studios. Chief engineer Jim McCurdy is punching in the 32-track 3M digital recorder via its remote console.

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THE SOUND ENGINEERING MAGAZINE

OCTOBER 1980 VOLUME 14, NUMBER 10

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

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db Letters

To The Editor:

I would like to comment on John Diamond's presentation at the AES Convention concerning human response to digital recording and Nelson Morgan's review of Dr. Diamond's presentation.

No one is questioning the value of true scientific research. On the other hand, many of the most significant discoveries in science have come about as a result of a human experience which could not be explained by available knowledge at the time. As a professional in the field of music reproduction, I have experience with high quality analog recording systems of different types and have had the opportunity to work with some of the more advanced digital systems on the market. Even a person with "untrained" ears can easily hear that there are a number of analog recorders which are more accurate in reproducing the line level signals fed through them than any digital recorder available. It is true that most analog recorders are noisier than digital, but it is possible to make analog recorders which meet or exceed each meaningful parameter offered by digital recording techniques available at this time. The big advantage of the digital process is expediency, not performance, and as such it has tremendous commercial appeal. It is the opinion of many individuals, including myself, that digital technology has great promise but needs much further development in order to have its significant potential realized. One of the questions is, how do we tell when digital technology is good enough to accept and utilize?

A number of my associates and I have experienced headaches and other discomfort while working with program material recorded digitally in patterns that suggest problems in that technology. Furthermore, the dealers and distributors for my company's products throughout the world and their customers also report in many cases similar feelings of irritation and discomfort when listening to digitally recorded material. Certainly, this is not scientific evidence. However, the sum of information available to me

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at this time indicates that there is more to this than is commonly understood or accepted.

One of the problems in today's world is that music, which is essentially an art, has also become an industry loaded with economic objectives and financial parameters that are not always in harmony and in fact are frequently in direct opposition to the interests of musicians and music lovers. As a musician who also happens to be President of a $3 million company, I am aware of many of the dynamics of this problem. The question is, when we die and are gone, what will we have contributed to this world? My personal and professional experience has made it very clear to me that digital in its present form is a totally unacceptable compromise as a medium for recording music and that any person who cares to contribute to music reproduction must face this fact. It is only by facing the truth that we can really make progress.

Mark Levenson
President
Mark Levenson
Audio Systems, Ltd.

db replies:

For the moment, we do not all agree on a standard direction in which "truth" may be found. We suspect that many readers will not agree "that digital in its present form is a totally unacceptable compromise." What about it readers—What do you think? (For every letter we print in our upcoming digital audio issue, we'll send a complimentary subscription or renewal.)

To The Editor:

I noticed with a bit of wry humor that you carried two advertisements for "The Recording Studio Handbook." I'm not sure the "Third Big Printing" or the "Fourth Big Printing" will draw the most response but it got a chuckle from me. Perhaps the ad department can get its act together later on. (Free subscription?)

Mind you, your magazine is a breath of fresh air over here in Swaziland where anything of a technical nature is from the good old U.S.A. We are watching with interest for your coming articles on studio design for radio since with our expansion, more studios are probably in the works. (We presently have three control room/studios for combo production of radio programming.)

One parting question: I'm wondering why the big recording studios seen in DB never seem to be using the pressure zone microphones? Perhaps I missed the point of the past articles, but I gather they are limited to mostly stage usage. In recording African music in bush (i.e. no power etc.) situations, it would be nice if these pressure zone units would work out. Acoustics are usually cement floor and walls with steel roofs with all the reverb you want and then some. Close miking helps and then reverb is mixed afterward. If pressure zone units would help, it'll be worth a try.

On the point of African music recordings (their musical instruments), we are in contact with someone who is an expert with several records to his credit and if you are interested I'll send you the info later. This is not some amateur but part of a society specializing in the study of African musical instruments, etc.

Anyway, enough of your time.

CALVIN DONNER
Trans World Radio
Manzini, Swaziland

db replies:

Thanks for the good word about **db**. *While we'd like to say that the book is selling so fast that we went from the third to the fourth printing in one issue, it was in fact a mistake. As to your PZM question, we passed that onto Crown. You will find their reply printed below.*

I do not know why the big recording studios discussed in **db** magazine do not yet seem to be using the pressure zone microphones. The reports we receive from the field show that their use is growing every day. The use of PZM's to record albums is growing at an even greater rate. They are new and their appearance in these articles is just a matter of time.

Your second question concerning the recording of African music is a little more difficult to answer. At first, three suggestions come to mind. One is to place a PZM in the random incidence field and shape the pattern of the PZM with acoustical foam. This would help to remove the wave coming off the back wall and ceiling and also help to remove inverted reverb.

The second would be to place the PZM on a hard plastic boundary (2-ft. square plate, or something similar) and move it into the direct field for close miking. By adjusting the placement of the PZM you will get more or less reverb from the building. This may be the most useful to you and may make it easier to get the recording you want.

My third suggestion would be to try a new PZM configuration which mounts in a corner. These PZM's use more than one boundary. This will help to lower the reflection from either 2 or 3 surfaces, thus lowering the reverberation in your recording.

At the present the corner-mount PZM's are only available from Wahrenbrock Sound Association; all other PZM's are from Crown International.

If you have further questions, please feel free to contact me.

DENNIS L. BADKE
Sales Engineer/
Training Coordinator

TO THE EDITOR:

First my compliments to Dr. Blesser on his excellent articles about digital audio. There is, however, one area in the sampling installment I would like to comment upon, regarding his discussion of 'Nyquist Frequency' (in the August issue—**Ed.**). One could get the impression that the 'Nyquist' defines an ideal or optimum sampling rate, while in fact it defines a minimum sampling rate for a given bandwidth. The choice of 40kHz or 44kHz sampling rates was highly motivated by cost as higher sampling rates would be very expensive (today at least). A 44kHz rate will be acceptable because our ears don't recognize accurate reproduction of 20kHz sine waves, if they require any reproduction at all.

JOHN H. ROBERTS

db replies:

Quite right—the Nyquist rate is the lowest-possible sampling rate, and is equal to twice the highest audio fre-

quency which the system must be capable of passing. On the other hand, the optimum sampling rate is a function of many variables, including cost, compatibility, the competition, etc. In a future column, Dr. Blesser will discuss the choice of a sampling rate in greater detail.

TO THE EDITOR:

Remember those wonderful Wollensacks? Back at WGBH we used them by the handful for auditioning and office listening. They were rugged, dependable and delivered damn good quality. They were also built to last.

At one point 3M bought the company, but I have not been able to find parts for a T-1500 anywhere. Perhaps you know of a place or maybe your readers can help. There must be hundreds of these machines sitting around waiting to go back into some kind of service for want of a small part or two.

ROBERT D. CAREY

db replies:

3M recommends contacting Chicago Tape Recorders Specialists, 4226 W. 26th St., Chicago, Ill. 60623. Tel: 312-522-0500.

db Calendar

OCTOBER

30 **Society of Professional Audio Recording Studios — Audio Recording Conference III.** Doral Inn, New York, NY. For more information contact: M. Rosenberg, SPARS Administrator, 215 South Broad Street, Philadelphia, PA 19107. Tel: (215) 735-9666.

29-30 **Kentuckiana 6th Biennial Sound & Communications Seminar.** Ramada Inn Northwest, Indianapolis, Indiana. For more information contact: Tony Monfort, P.O. Box 40905, Indianapolis, Indiana 46240. Tel: (317) 849-5726.

31-Nov. 3 **AES 67th Convention.** Waldorf-Astoria Hotel, New York, NY. For more information contact: Audio Engineering Society, Inc., 60 E. 42nd St., Rm. 2520, New York, NY 10165. Tel: (212) 661-8528.

NOVEMBER

9-14 **Society of Motion Picture & Television Engineers—122nd Conference Equipment Exhibit.** Hilton Hotel, New York, NY. For more information contact: SMPTE, 862 Scarsdale Ave., Scarsdale, NY 10583. Tel: (914) 472-6606.

11-13 **Synergetic Audio Concepts Seminar—Sound Engineering and Acoustics.** Dana Point Marina Inn, Dana Point, CA. For more information contact: Synergetic Audio Concepts, P.O. Box 1115, San Juan Capistrano, CA 92693. Tel: (714) 496-9599.

20-23 **Billboard's 2nd International Video-Music Conference.** Sheraton-Universal Hotel, Los Angeles, CA. For more information contact: Nancy Falk, Billboard, 9000 Sunset Blvd., Los Angeles, CA 90069.

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BARRY BLESSER

db Digital Audio

• For the next few articles in this series, we will turn our attention to the actual circuits which are used to perform digitization of audio. Some of these circuits are extremely complex. Nevertheless, we need to understand their inner workings in order to properly select and evaluate them. A design engineer will *always* buy circuit modules from a specialty manufacturer rather than build them from discrete parts. The single most critical module is the DAC (digital-to-analog) converter, which accepts a digital word as input and produces a corresponding analog voltage or current. Interestingly enough, the same DAC module forms the heart of the A/D converter as well as the D/A converter.

THE DAC MODULE

There are many different circuit configurations for producing the DAC function, but they all consist of precision voltage or current sources which are switched on or off by the input digital word. For an N-bit DAC, there will be N sources; each having a ratio of 2:1 to the neighboring source. FIGURE 1 shows how a 5-bit unipolar DAC is made using 5 resistors with values R, 2R, 4R, 8R and 16R. When a switch is closed, the voltage

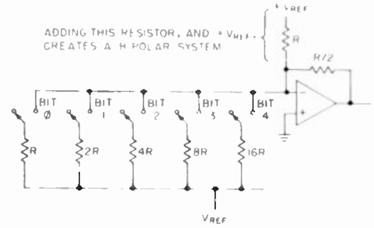


Figure 1. Current mode DAC with binary-weighted resistors for controlling the current into a summing junction of an op amp.

across the corresponding resistor will be V_{ref}, since the summing junction of the OP amp is at 0 volts. Adjacent resistors—and therefore, currents—have a binary ratio. For example, the least significant switch (bit 4) produces an output of V/32 since the effective gain is the ratio of the feedback resistor, R/2, to the input resistor, 16R. The next switch produces twice the output, V/16, since its resistor is 8R. We can represent the full output by the following equation:

$$
\begin{aligned}
V_{out} = {} & (\text{bit } 0) \times V/2 + \\
& (\text{bit } 1) \times V/4 + \\
& (\text{bit } 2) \times V/8 + \\
& (\text{bit } 3) \times V/16 + \\
& (\text{bit } 4) \times V/32
\end{aligned}
$$

The table in FIGURE 2 shows the voltage output for the corresponding digital word input (switch states).

Digital Input	Analog Output (Unipolar)	(Bipolar)
00000	0/32 V	−1/2 V
00001	1/32	−15/32
00010	2/32	14/32
00011	3/32	−13/32
00100	4/32	−12/32
00101	5/32	−11/32
00110	6/32	−10/32
00111	7/32	−9/32
01000	8/32	−8/32
01001	9/21	−7/32
01010	10/32	−6/32
01011	11/32	−5/32
01100	12/32	−4 43
01101	13/32	−3/32
01110	14/32	−2/32
01111	15/32	−1/32
10000	16/32	0 V
10001	17/32	1/32
10010	18/32	2/32
10011	19/32	3/32
10100	20/32	4/32
10101	21/32	5/32
10110	22/32	6/32
10111	23/32	7/32
11000	24/32	8/32
11001	25/32	9/32
11010	26/32	10/32
11011	27/32	11/32
11100	28/32	12/32
11101	29/32	13/32
11110	30/32	14/32
11111	31/32 V	15/32 V

Notice that this DAC is unipolar. To make it bipolar for audio, the range is offset by V/2, as shown in the upper part of FIGURE 1, where an offset resistor of value R is connected to $+V_{ref}$. This subtracts a voltage V/2, producing the bipolar outputs also given in FIGURE 2. Note that the range is not quite symmetric, since there are an even number of states, but 0 volts is one of the positive voltages. There are 16 negative non-zero states but only 15 positive non-zero states. The range could be made symmetric if the offset were −31/64 V. Then the most negative output would be −31/64 V and the most positive would be +31/64 V, but there would be no state corresponding to exactly 0 volts. Since audio is an AC signal, this difference is irrelevant.

For large numbers of bits, there is a serious difficulty with this DAC. Consider the extreme case of a 16-bit device: the ratio of the largest-to-smallest resistor is 2^{16} = 65,536. If the smallest

resistor is 1k, then the largest is over 65 M ohms. Therefore, if the largest current is 2 ma., the smallest current is 30 nanoamps.

This would be almost impossible to implement because of leakage currents, distributed capacitance, switch errors, and other defects. Another type of circuit configuration is thus preferred for high quality DACs. This uses a special resistive network called an R-2R ladder, as shown in FIGURE 3. This has the property that the voltage at each node is twice that of the neighboring node. We can demonstrate this without extensive computations by making some equivalent circuits of the basic structure. In the top of FIGURE 3, we have the basic 5-node ladder, with the last section shown in the dotted box. The equivalent resistance from node 1-to-ground is R, since it is the parallel combination of 2R and 2R. Thus, we have a 2:1 voltage divider between node 2 and node 1, since there is a series resistor R between these nodes, and an equivalent resistor R from node 1-to-ground. Hence, $V_1 = \frac{1}{2} V_2$.

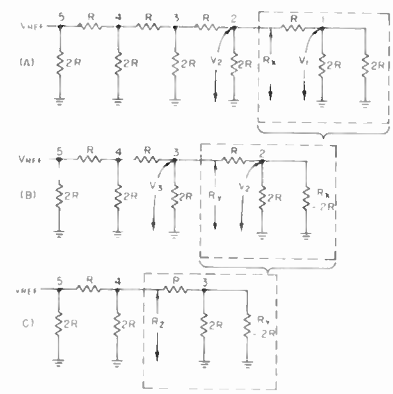


Figure 3. Ladder of resistors, R-2R, used to create binary currents. (a) basic 5-bit ladder. (b) dotted box of above is replaced with equivalent resistor Rx, of 2R. (c) again, the dotted box of above is replaced by equivalent resistor Ry, of 2R. Voltage at any node is half that of the node to the left.

This voltage divider has a composite resistance, $R_x = 2R$. Thus, it may be replaced (as in FIGURE 3B) by a new resistor $R_x = 2R$. Now we see that there is an effective voltage divider with a 2:1 ratio between node 3 and node 2. Hence, $V_2 = \frac{1}{2} V_3$ and $V_1 = \frac{1}{4} V_3$. This analysis can again be carried out for the next node by replacing the dotted box of FIGURE 3B with an equivalent resistor $R_Y = 2R$ in FIGURE 3C. The analysis repeats.

In each case, the node-to-ground current in each leg of the voltage divider is equal to the node voltage divided by 2R. Thus, the currents have a binary ratio to the preceding one. This allows us to create all the required currents using only two values of resistor, R and 2R, rather

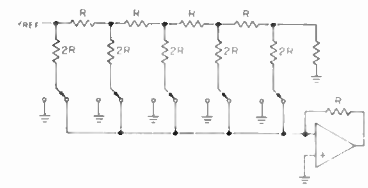


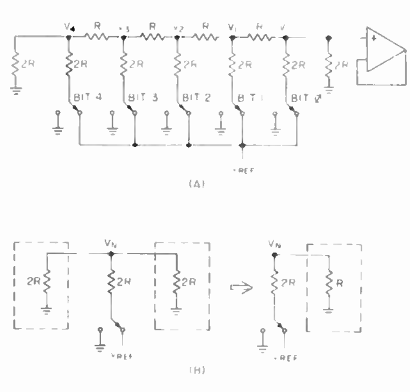
Figure 4. Use of R-2R ladder to create binary switched currents.

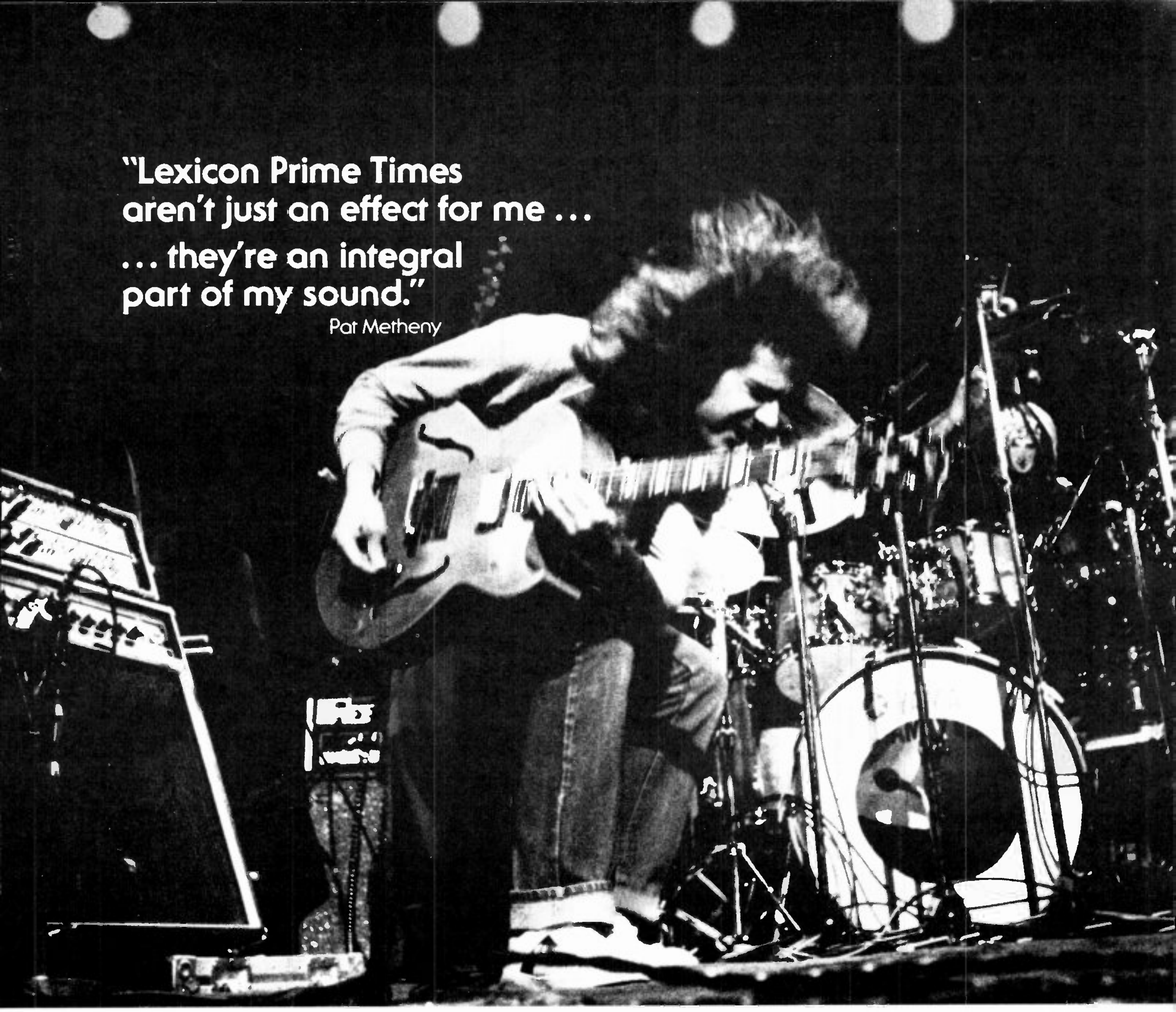
than binary resistors. In the complete circuit configuration (FIGURE 4), the switches are SPDT, since the current in the resistors flows either to the op amp or to ground. This is required since the state of a switch must not be allowed to effect the voltages at the other nodes. The only difference between the circuits in FIGURES 1 and 4 is the way the binary currents are created.

This approach solves the problem of having a wide range of resistor values, but it does not solve the problem of the wide range of currents in the switches. Another variation of the R-2R ladder is shown in FIGURE 5. This is like the previous ladder except that the voltages of the 2R ladder elements are switched, and the output is a voltage directly. To understand the operation, consider any single node, which we will call V_n (FIGURE 5B). It has an effective load to the left of 2R and an effective load to the right of 2R, as shown in the dotted boxes. This creates a voltage divider between the switched resistor of 2R and a composite load of R (2R in parallel with 2R). Hence, V_n will have a contribution of either 0 volts or V/3, depending on the switch state. The next node's voltage will be influenced by this switch to produce either 0 or V/6, since the 2:1 divider ratio is still valid, as we showed previously.

In other words, the switch for bit 0 will contribute either 0 or V/3 to the output at node 0. The switch for bit 1 will contribute either 0 or V/3 to node 1, which contributes 0 or V/6 to node 0. Similarly, the switch for bit 2 will contribute either 0 or V/3 to node 2, 0 or V/6 to node 1, and 0 or V/12 to node 0. This is a voltage ladder in which

Figure 5. Voltage switching in an R-2R ladder to create binary voltages at output.





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the switches control voltages to the 2R resistors. Notice that all of the switches are switching between the same voltages and they all have the same currents in them. In this configuration, the output is a voltage which is buffered by a unity gain op amp.

ACCURACY

It is not easy to build a quality DAC with a large number of bits because of the accuracy problem. To demonstrate the extreme difficulty, let's continue to discuss our 16-bit DAC with an output range of ±10 volts. The distance between two neighboring quantization levels is 20/65,536 = 0.00305 volts. This means that the output produced by the digital word 1000000000000000 is 3 millivolts larger than that produced by the digital word 0111111111111111, meaning that the sum of the lower 15 bits must be very accurate compared to the value of the highest bit.

Notice that the accuracy requirement is on the individual bits as well as on the sum of the bits. For the MSB (most significant bit), the requirement of an accuracy of one-half of a quantization level means a relative error of 0.0015 percent. Similarly, the sum of the lower 15 bits must also have the same percentage accuracy. This is the worst case since the sum of the 15 currents (or voltages) must be very carefully controlled. This is called the major carry, which, unfortunately, corresponds to the center of the DAC range. In audio terms, this is the region of small signal level. The above percentages refer to full scale, but the relative error to the audio signal is much higher when the audio is low level.

The problem of accuracy is made even worse by the fact that the accuracy must be achieved in a very short amount of time. We speak in terms of a few microseconds for the DAC to achieve this kind of accuracy. It is for these reasons that the design and manufacture of DACs of high precision is left to specialty manufacturers rather than to an audio engineer. Even so, the requirements of quality audio are almost beyond the limits of today's technology.

FORMAT

The digital word format in the previous discussions showed a maximum voltage for the word 111111111 and a minimum for 000000000 even when offset for symmetric range. Since the first bit is usually referred to as the sign bit, the MSB is often complemented at the input to the DAC. This gives the most positive digital word as 011111111 and the most negative as 100000000. Some DACs have an auxiliary input for the MSB to achieve the complementing without the requirement for an external inverter. The use of this so called 2s complement format will be discussed later. ■

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Sound Separation

• In this discussion of basics, we have found that sound can be separated or analyzed in different ways. A wave analyzer does it simply by frequency and amplitude. Our hearing obviously does it in a more complicated way: we separate various composite sounds within the overall composite sound we hear. It may be different musical instruments in an orchestral performance, different human voices in a crowded room, or whatever.

The question we want to address this month is, how do we do it? What is critical, in the reproduction of sound, to this capability, and what is uncritical? When stereo was first being introduced, a lot of work went into determining this, and it has not altogether ceased since.

One thing that investigators found, before even stereo made the general scene, was that spurious frequencies present could destroy our capability of separating sounds. The presence of distortion, in particular, had this effect. The presence of a strong hum could do so too, but not with such small amounts as distortion would do it.

The presence of other spurious sounds such as people talking, foot scuffing, paper rustling, applause, and so forth, has more the effect of being distracting, than of preventing separation. You wish they would be quiet (with the exception of applause, maybe) so you could hear better, or more comfortably. But you can still hear, unless they become very loud in their interference.

If you play early recordings of bands, where the distortion level was higher than it is today, you.will find that you cannot separate, or perhaps even tell which is which, on solo parts, between brass, woodwind and even other instruments.

The distortion makes all the sounds blend together, so individual parts are inseparable to the ear. It may also prevent identification of what instrument is playing a solo part. We've come a long way from that stage today: you'll have to find a really old record to demonstrate the effect—like 30 or 40 years old.

When stereo was being introduced, particularly on forms of disc (before the 45/45 cutting system became the standard), some were arguing whether stereo could be any better than mono. Why? Because the early stereo records had more distortion on them than mono recordings of the time, with the result that stereo was often demonstrated for the so-called ping-pong effect, rather than for improved separation of instruments, or parts.

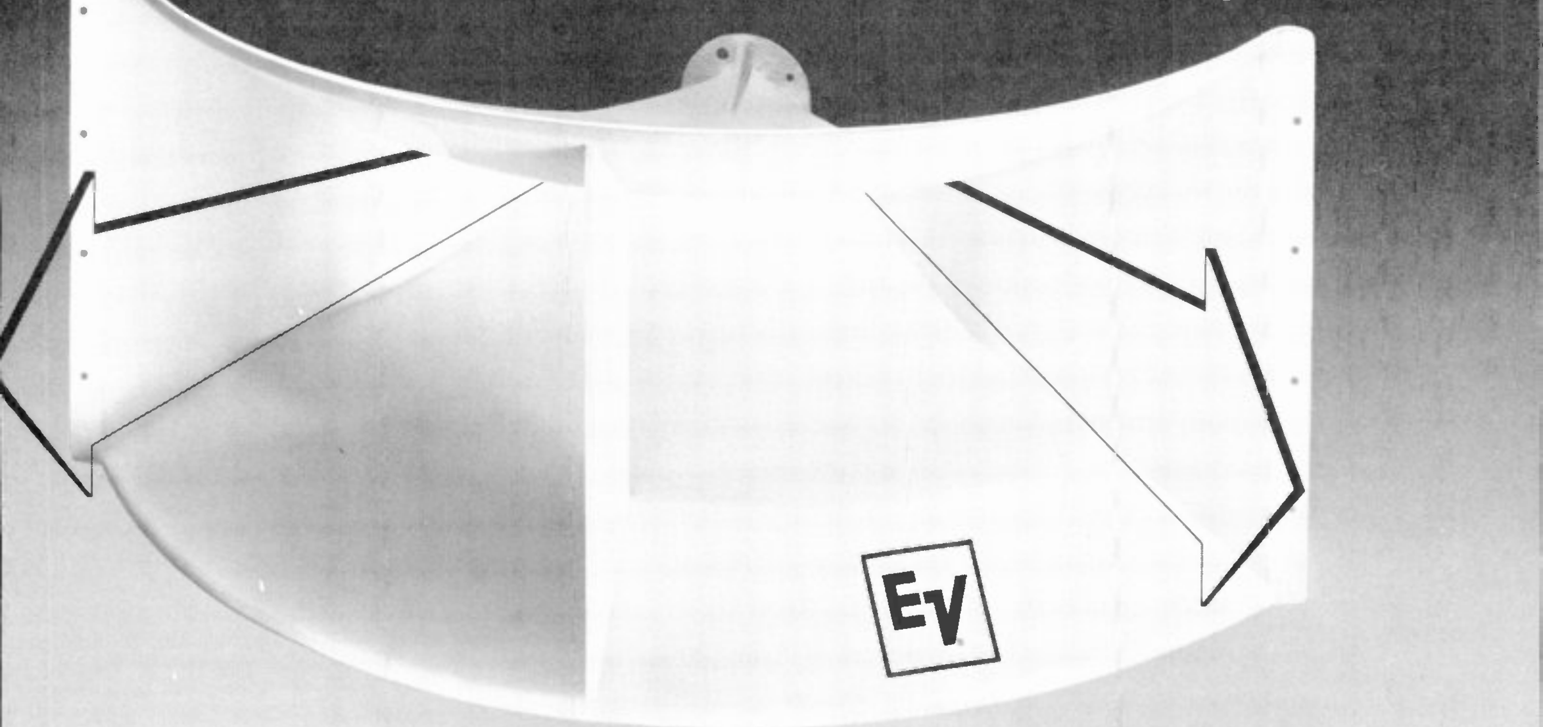
So, freedom from distortion is essential to good capability at separation, even more so than the complete absence of spurious sounds unrelated to what we want to separate.

As the quality of stereo improved, to the point where distortion is now at lower levels than it ever was on pre-stereo mono, clarity and separation improved. We came to realize that separation means more than just being able to tell that brass is on the right, woodwinds on the left: physical separation, in the sense of recognizing that the instruments occupy different positions, is not so important as being able to tell that both instruments are there, regardless of where they may be.

During the early period of stereo, work was directed at determining what it was that enables our hearing to provide the illusion of separation. In those days, separation meant the physical sense: telling that different sounds came from different physical locations. While some researchers may have placed undue emphasis on this—after all, when you listen to an orchestral performance in an auditorium, you do not bother locating each instrument or group of instruments (probably couldn't if you tried)—some useful results came from this work.

It was generally concluded that such physical separation was prevalently due to phase relationships at the lower frequencies, and to intensity relationships (between the sound heard by each ear) at the higher frequencies, with a range of frequencies in mid-range where both effects contributed to the illusion. This led to a valid conclusion: that separation occurs better in mid-range, and that our capability for "placing" sources of sound relies mainly on the mid-range frequencies.

con′stănt di·rec·tiv′i·ty



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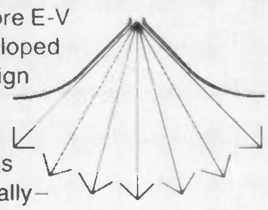

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[1] U.S. Patent Number 4071112

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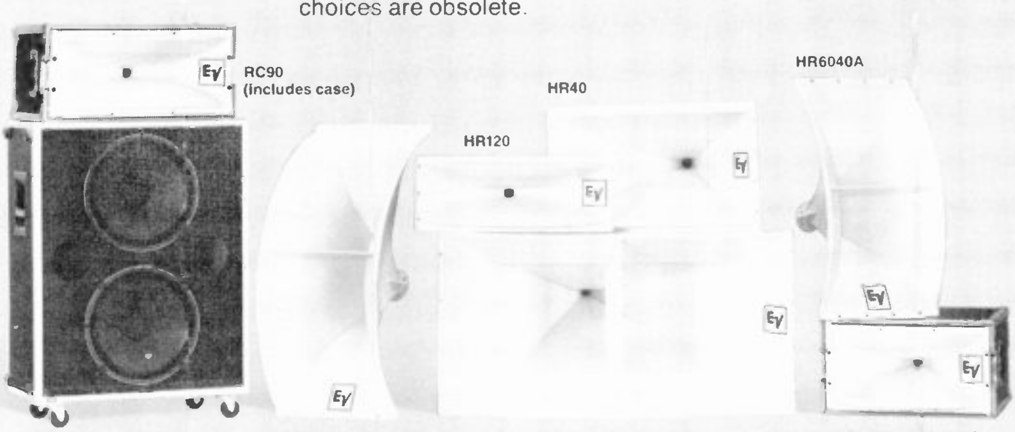

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A closer examination of these results showed something else: our hearing faculty can separate original, direct sounds, but it has difficulty with reverberant sound. In an auditorium, the original sound (without reinforcement to confuse things) comes very definitely from on-stage, but the reverberant echo comes from all around. It has to be a very bad piece of acoustic design that enables you to notice a "lump" reflection from a big piece of wall somewhere, separate from all the rest.

Lesser degrees of reverberation may produce confusion of sound, but your hearing does not seem to bother with where the confusion comes from: it only tries to locate the direct source. How can it tell the difference?

In most situations, the direct sound is the first to reach your ears; the reverberant sound comes later. So it became apparent that somehow our hearing pays more critical attention to the first sound to reach our ears than it does to the "follow through" elements of the sound. That is, it pays more critical attention to the "leading edges," so to speak.

One problem for all of us in trying to research this, is the fact that we learned to use our hearing very early in life, and thus take it for granted. We think we know what we hear, so we don't really bother to analyze how we hear it. Ordinary reverberation we tend to ignore, or if we are conscious of it, pay little attention to it. But the sound of a single pistol shot, bouncing round the walls, we may notice, because we don't hear anything quite that well separated in reverberant sound everyday.

There is another reason why we notice that: each reflection possesses the same property that enables our hearing to focus on the direction of original sounds: a short, sharp, impact, or transient sound. Most reverberation lacks that property.

CBS Labs, particularly the late Ben Bauer, who contributed a lot to audio development, did some work on acoustical fields of sound. Rather than go through the theory that Ben did—many of our readers shy away from theory—we could go back to a practical form of loudspeaker, and to problems in phasing, that lead us to the same thing.

Before we had loudspeakers mounted in cabinets, they were commonly mounted on open baffles, so that sound could emanate from both front and back of the unit. We usually listened in front. But if you got round to an edge-on position, you noticed some peculiar effects. For one thing, all that lovely bass—which was why we used a baffle (the bigger the better)—disappeared when you listened from an edge-on position, and all the other frequencies lost their sense of source. From the back, or from the front, you could tell they were coming from this unit, but edge-on, the sound

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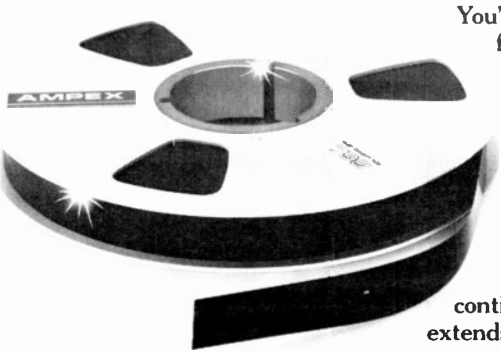

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seemed to come from "nowhere."

As multiple loudspeakers came into use, particularly for covering large auditoriums, those who installed them found that phasing was important. And one way to verify phasing consisted of placing a couple of loudspeakers fairly close together, side by side, and reversing the connections to one of them, while you listen directly in front, at a position equidistant from the two.

Making the connection one way, in phase, resulted in sound that seemed to come from midway between the two units. Making it the other way, out of phase, resulted in sound that no longer seemed to come from those loudspeakers at all. This effect was given the name the "dissociation effect."

Later, when stereo was beginning to take hold, but some were wanting to simulate it using their old mono program as source, one of the most successful ways utilized a short tape delay to simulate reverberation. The direct sound, undelayed, was played into both speakers in phase, while the delayed sound was mixed with it, reversing the phase of one feed. Compared to some of the early true stereo material, this was almost the same thing: it added the sound of an auditorium to sound recorded without it.

The CBS experiments went further. It used two observations of the time, that are still true, to produce a very good stereo illusion. We wondered why it was not more popular than it ever became. I think the only reason was that people do not like to admit they are being fooled!

The system used three loudspeakers. Frequencies below a crossover that could be in the range from 100 to about 250 hertz, were fed to a single woofer that could be concealed under a sofa, or behind a wall cabinet. Frequencies above this crossover were fed, stereo, to a couple of small, open back units, that CBS called "dipoles," or "isophonic."

What they had proved was that two such dipoles, fed with two-channel (stereo) sound in appropriate phase and/or intensity relationships, can simulate a stereo sound field throughout the room, over a far wider listening area than two conventional, closed back units can. Further, although the body of sound from such instruments as a string bass, or the deeper organ tones, really came from the concealed unit, the transient sounds, with which those tones started, came from the little dipoles, and fooled our ears into believing that the whole sound came from there.

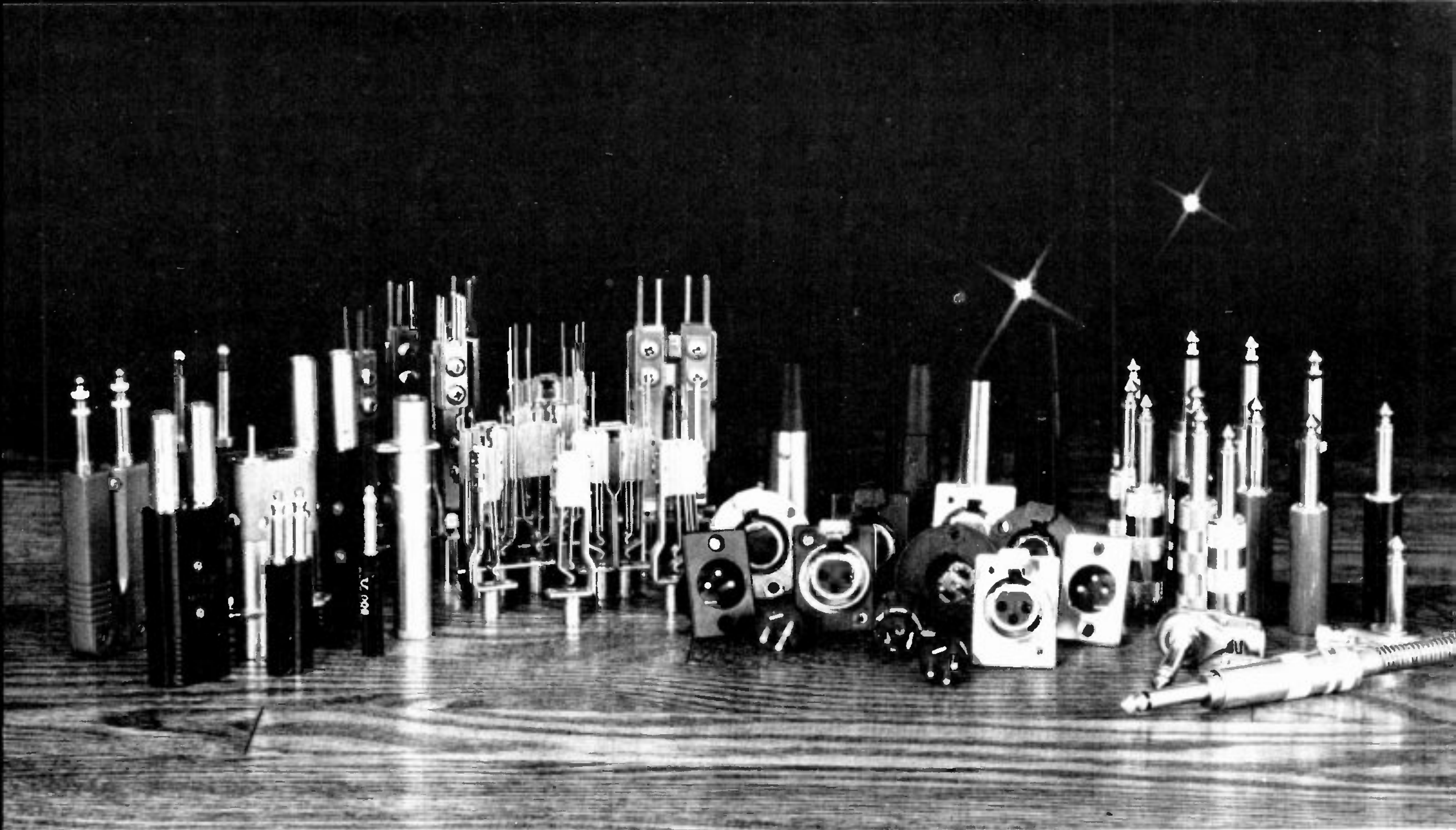
It was a case of "it's not nice to fool mother nature!" And to those who always believed that accuracy in reproduction, in every sense, was the secret of audio realism, this was a real blow, one they were not ready to accept.

Effects can by synthesized. Since then, we have gone to digital audio, with wonderful results. In the field of optics, whether motion picture film or video technology is used—or even combining the advantages of both, wonderful things are done. Will we reach that stage with sound and audio? Audio always seems to have been the "poor relation." It has a lot of catching up to do.

What can we expect? Can selective time delays be used to focus in a certain direction, relative to a pair of stereo mikes, like human hearing can? It would also need to be programmed so it gives particular attention to the transient beginnings of new sounds. A way would need to be found to ignore sounds in which we are not interested, like human hearing can—unless the intensity level is high enough to produce masking.

Mixing is old hat—it's been done ever since we've had audio. But mixing in audio is like lap dissolves in video or photography, and equally old hat. Video has brought in techniques of superimposition that are far more intricate than dissolves, enabling some fantastic illusions to be achieved. Can the equivalent be done with audio?

Undoubtedly, digital audio is a step toward that, but we still have a long way to go. Each step begins by understanding what we already have, and then formulating where we want to go from there. By knowing what human hearing will do, we take the first step toward finding a way to duplicate it. ■



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MARTIN DICKSTEIN

db Sound With Images

One of the most distressing occurrences possible has probably happened to most of us when putting on an AV show including slides with sync sound and/or film; the sound during the presentation was horrible. If it has not happened to you, you may have been in the audience during such a show. There was this super slide show, or film, and the sound was lousy.

Let's look at a few simple considerations. At the beginning, when the show first goes into production, a decision is made on what has to be seen, and anything that has to be said is put into the script. Music is decided upon to fit the mood of the presentation, to fill the gaps between sections of the show, to open and close the modules or segments, and to provide and keep a rhythm going for the movement of the slides. Music can be used for bridges, stings, background, dramatic openings and closings, or for many other purposes. The voice used to read the script is usually that of a professional, chosen for the quality of his or her voice, the mood put across during the reading, the ability to hype an audience into action, and/or the skill to be a "quick study," who reads the script over in advance, and then reads precisely as desired in just one or two recordings. (This last quality can sometimes save money in a recording studio.)

Several other decisions also have to be made on the music. Are there recordings available (records, tape, music files and libraries, etc.), or should the music be written specially for the show? Is there a problem with license and clearance for the use of existing records and tapes? If the music is written for the show, how big a group should the orchestration include? How long will it take to write, score, record, mix down, etc. and how much will it cost? The music might also include special effects. Then, the voice and music have to be mixed. Equalization and reverberation might have to be added. It all can turn out to be a lengthy, tedious, and expensive proposition. Or, it can be as simple as taking a record off a shelf, recording the script into a microphone plugged into a home tape recorder, and then getting the two elements mixed. (The latter method leaves a lot to be desired compared to the more professional way, but it's been done, consideration of music clearance aside.) It still would be a shame if either method sounded badly during playback at the show. The more expensive the recording process, the greater the crime when the audience can't understand the words, or is turned off by the poor quality of the sound distribution.

If the show is produced by a professional AV production company, there is usually someone from the organization that will be at the show to be sure all their work is not in vain. If the presentation is produced and then handed over completely to the client, it becomes his or her responsibility to see that the sound quality at the show is *great*, or at the very least, *good*. There should be no settling for anything less.

The final step is taken by the company providing the sound equipment to be used during the showing. In some instances, the show is sent to the location with a specified or recommended sound system. In others, equipment is rented locally, and installed at the site by the rental house. In other situations, the in-house system is used. While this can sometimes be the least expensive method, it may end up being a most costly mistake. The speakers could be located on the ceiling, if it is very high (like a ballroom), or at the side walls, which can

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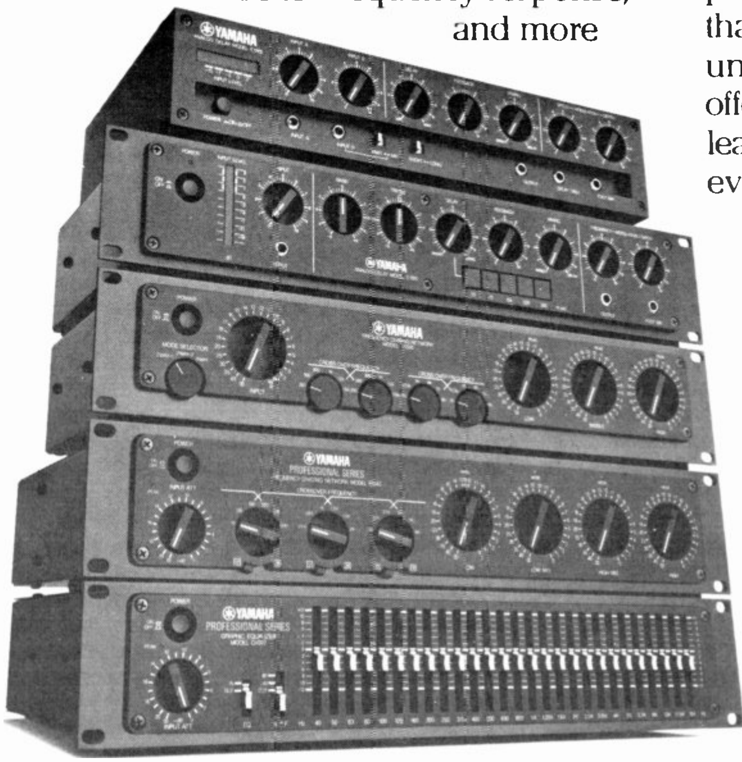

The analog delays: E1010 & E1005 The creative applications of these two analog delays are almost endless. They offer echo, flanging, reverb, time delay, and double-tracking—just to name a few. And being analog, these delays retain the original audio signal for a true musical sound.

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cause the audience to lose the effect that might have been achieved if the loudspeakers were at the front, near the screen. With speakers located around the room, or up high, a certain amount of reverberation can result which can also be distracting. The speakers could be of poor quality and the resulting sound could be too thin or too bassy for good clear audibility. Also, the amplifiers could be old and not capable of handling the audio or dynamic range of the taped music. The same of course, could be true of the speakers. To get sufficient volume out of the system for the intended effect, the amplifier/speaker combination might have to be driven to the point of terrible distortion. Some of the hotels, and many of the newer conference centers, have improved sound systems, but even they might not have what the show requires and deserves in dynamic range, quality, and volume. It has become almost a standard requirement that the sound should come from the front, where the screen is, but many installed sound systems still have overhead or side speakers (to avoid feedback during programs in which a live mic is used).

When a system is brought in with the show, the equipment is usually run by someone experienced with that equipment. Precaution has been taken to provide amplifiers powerful enough, sturdy and good-quality speakers, and thought has been given to the equipment set-up based on the design of the set, the shape and size of the room, the size of the audience, the type of show, and the producer's requirements. It may be expensive to ship the system, including a technician, keep that person at the site and then ship the whole thing back, but it is an expense well worth serious consideration.

The other choice is to have a local rental company bring in the equipment and set it up. In a great many cases, these people know the site, and are aware of the problems from previous experience. They have equipment in stock, possibly know what setups have worked for different situations, and can be very helpful to the client in clearing the way with the location personnel for early delivery, storage, and expedient removal of the equipment. They can also cope with the local union problems, if any, and can arrange with the site for proper tables, platforms, etc., needed for a stage arrangement.

There can be problems, however. The person responsible for the show's setup should visit the site with the rental company to discuss equipment requirements and setup. If left alone, the rental house could be doing several shows simultaneously, and the best equipment could be somewhere else. Using "what is available" can sometimes ruin a show. If specific wishes of the client, or the producer, cannot be met with in-house equipment, the proper units should be sent, or gotten from wherever it's available. Substitutions should be indicated at the time of discussion so that the client can be certain that the best results will be achieved. Too many times, the rental company has not been selected carefully enough, and the units brought in on the morning of the show are just not good enough.

A few precautions. If you're involved with the production of the show, make sure you know what you want or need for the presentation in sound distribution, even if you have to pay a consultant for the help. Spend a good deal of time on the audio, just as you did with the visuals. If you are in the sound rental business, you can hurt yourself by not giving the customer what he asks for or needs to have the best show possible. Make sure the equipment meets the presentation's requirements, and fills the bill for the room. Be sure all the equipment works properly and all connecting cables are in good shape. Be sure to test the system at the level that will be used with all the people in the room. Use equalizers and level meters and whatever else you can to make sure the sound is good, clean, and fully understandable throughout the room. Whatever your part of the total production, spend as much time on the audio as possible, so the time you spent on the visuals won't go to waste. ■


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Mfr: Soundcraftsmen
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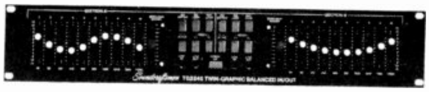

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Mfr: Bose
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MIXERS

• Despite their compact size, the BC102 and BC104 offer the same circuits and components as do larger consoles. The BC102 has 10 input channels and two output groups while the BC104, with the same 10 channels, offers four group outputs. Two or four output groups are switchable to respective mixing buses with pan facility. Reverberation and foldback Send with individual gain control and choice of pre- or post-fade signals is available as is PFL and Solo facility from all channels. An EQ in/out switch is provided with indicator. 48v Phantom supply is switchable on all channels. Comprehensive Equalization includes High Pass Filter, High and Low frequency cut/boost and continuously variable mid-frequency with cut and boost. The metering and monitoring, with individual selection switches, enables monitoring of all outputs, playbacks, Rev Send, FB Send and Rev Return. Standard on both consoles are talkback send and return facilities along with separate stereo level controls for headphones and Control Room monitors.
Mfr: Tweed Audio
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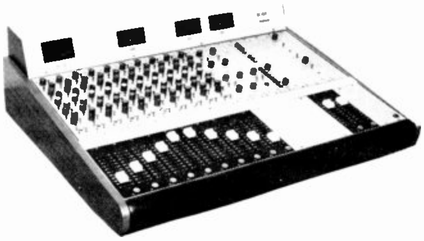
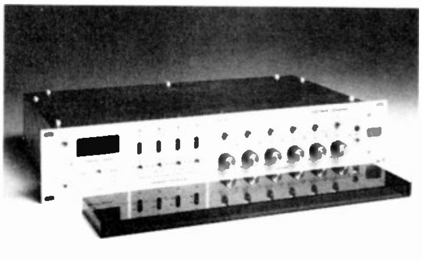


ELECTRONIC CROSSOVER



• The UREI Model 525 Electronic Crossover features 4 panel-selectable operating modes: stereo 2-way or 3-way and mono 4-way or 5-way. Crossover frequencies are continuously adjustable from 50 Hz to 10 kHz, with the actual frequency measured and displayed on a digital frequency counter, with 1 Hz resolution. A subsonic switch-selectable filter is included to roll-off frequencies below 30 Hz, providing protection of the low frequency transducers in the P.A. system. Inputs and outputs are XLR/QG connectors or terminal strips.
Mfr: UREI
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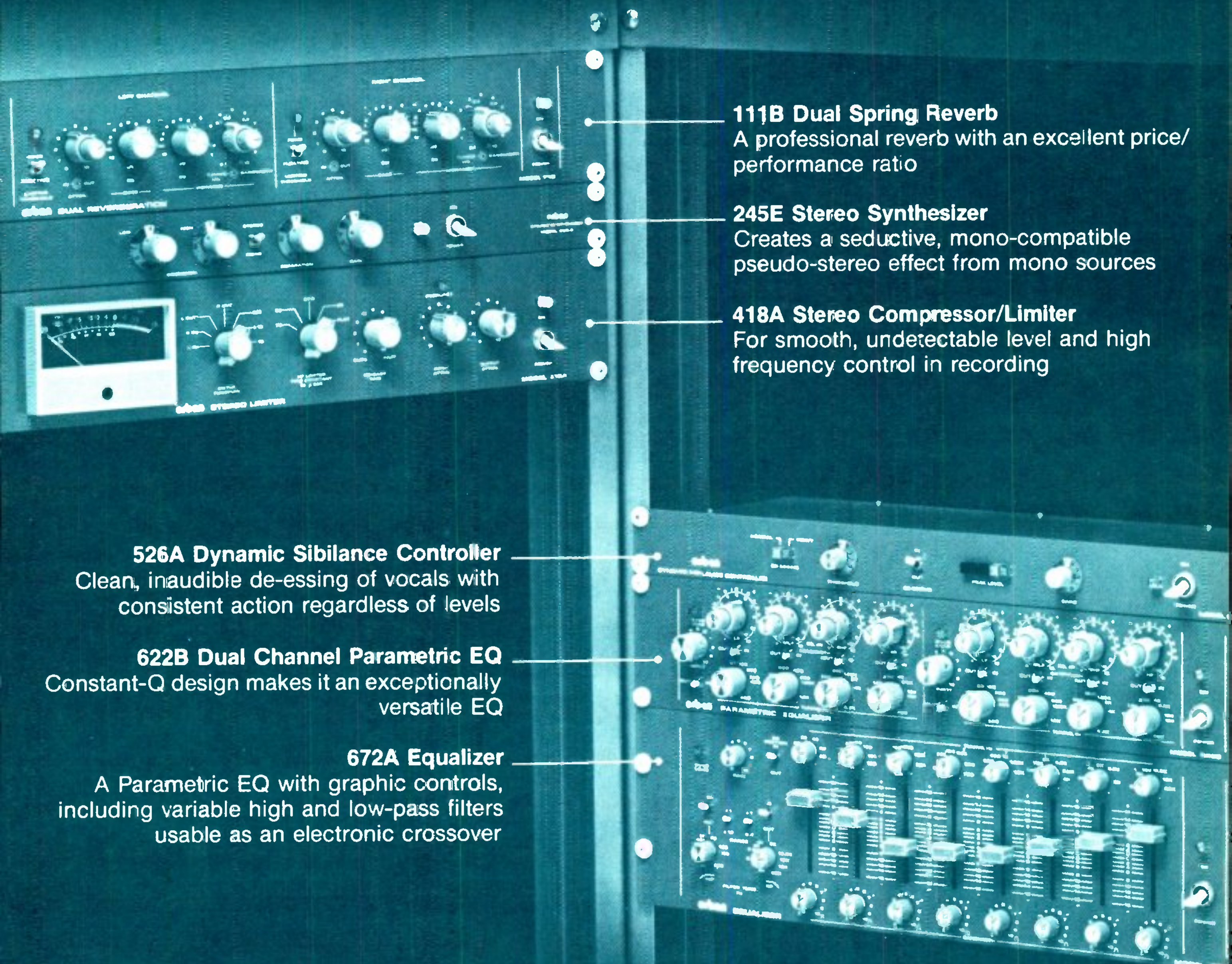
Six Ways To Make Your Sound Sound Better



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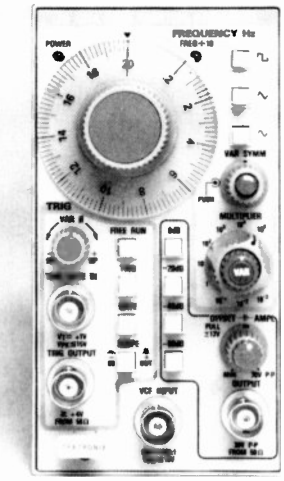
FUNCTION GENERATOR



• The FG 501A is a one-wide, TM 500 function generator, that provides low distortion outputs from 0.002 Hz to 2 MHz. It is capable of generating five basic waveforms—sinewave, squarewave, triangle, ramp, and pulse—at output levels up to 30 volts peak-to-peak with up to ±13 volts of offset from a 50 ohm source. Waveform triggering and gating are provided with a variable phase control to permit up to ±90 degrees of phase shift for generating haversines, sin² pulses, and haver triangles. A step attenuator provides 60 dB of output signal attenuation in 20 dB steps with an additional 20 dB of variable attenuation.
Mfr: Tektronix
Price: $680.00
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PHASE METER

• The Model 200 has a 0-360 degree mode for lagging test signal or ±180 degree mode for signals whose phase relationship varies around zero. The input impedance is one megohm for all input levels. The frequency range for a 35mV RMS signal is 20 Hz to 300 kHz. The accuracy over this range is ±2 degrees. There are front panel calibration controls which can also be used to offest any phase shift in the test setup and an analog output providing 10 mV per degree for plotting or recording purposes.
Mfr: FSI
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MIXING CONSOLE

• The Speck 800-D is a 28 input, 16/28 output mixing console. It is totally modular with 28 input modules, a master module, and a complete communications module housed in a mainframe that contains 16 illuminated V.U. meters. Each input has eight panable assigns, three band parametric equalizers, three sends, a long throw slide fader, and a second line input with an independent slide fade, two band equalizer and pan. The stereo program buss is independent of the multitrack assign section, allowing the console to feed a full complement of ½ track, ¼ track and cassette recorders simultaneously during mixdown.
Mfr: Speck Electronics
Price: $25,190.00
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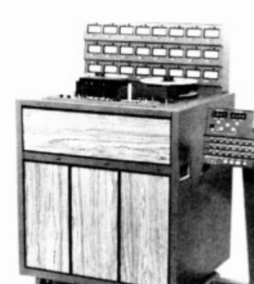
MULTITRACK TAPE RECORDER



• The JH-24 features totally transformerless electronics providing improved frequency response, signal-to-noise ratio and RFI rejection. Available for use with the JH-24 is the AutoLocator III, a microprocessor-based tape counter/locator with 10 memories and built-in Tape Velocity Indicator function.
Mfr: MCI
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• The PL80 dynamic cardioid vocal microphone marks the culmination of the application of a new concept in computerized microphone design—"the Fast Fourier Transformation" (FFT). Designed for the professional and made of zinc and aluminum along with a dent-resistant Memraflex grille screen, the PL80 employs a shock mount to reduce handling noise plus a built-in Acousti-foam blast filter to reduce "P-popping."
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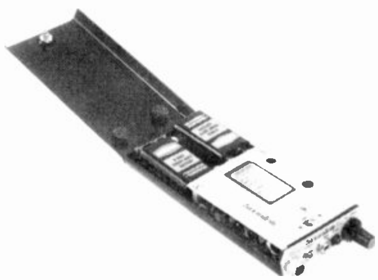
Price: $199.95
Circle 54 on Reader Service Card

BROADCAST CONSOLE

• The Series 24A offers a wide range of input, monitoring, communications and metering modules. The design of the mixer permits the modules to be fitted in any number and any combination without modification to the expandable main frame. This design also allows the console to be used in a number of applications, including: engineer-driven program production, outside broadcast mixing, master control room network switching and general audio visual production.
Mfr: Allen and Heath Brenell Ltd.
Circle 55 on Reader Service Card



PORTABLE RECEIVERS

• The Mark Q and Mark Q/dB-S mini-VHF receivers are made especially for cueing, bilingual remote listening, recording and amplification for the hard-of-hearing. The Q receiver has been mechanically designed to be worn on the body or to directly interface with recorders when necessary. It is also crystal-controlled and can be supplied on any frequency between 120 and 240 MHz. The dB-S receiver is equipped with an audio scaling expander/compander to achieve a S/N ratio better than 80 dB and help eliminate buzz zones and other forms of low level interference.
Mfr: Swintek Enterprises, Inc.
Circle 56 on Reader Service Card

MIXING CONSOLE

• The 821, a multipurpose mixing console, features built-in reverb with master level and pan control, separate input preamp input/output jacks for signal processing of input channels, slide-type master output controls and master monitor control. Other input channel features include mic/line switching; monitor, reverb and auxiliary sends; 3 band input channel equalizer and slide channel fader. The 821, with optional 220v, 50Hz line voltage available, employs a steel chassis and is fully rack mountable.
Mfr: Neptune
Circle 57 on Reader Service Card

Circle 42 on Reader Service Card

Circle 40 on Reader Service Card

New Literature

FUSED FEEDTHROUGH DATA SHEET

• A two-page bulletin of fused feedthrough adaptors contains electrical, mechanical, and environmental characteristics as well as mounting information. The adaptors are used in electronic equipment requiring protection from large input signals. The adaptors are 50 ohms, 1.75:1 VSWR maximum and DC to 2 GHz. Standard fuse is ⅛ amp with 0.01 sec. blow time (optional fuse is ⅜ amp with a 0.1 sec. blow time). **Mfr: Cablewave Systems, Inc., 60 Dodge Avenue, North Haven, CT 06473.**

APPLICATION NOTE

• An informative new application note on custom assembly of HEXFET MOSFET chips for hybrid circuits is now available. HEXFET chips or dice are the MOS elements used in the many packaged HEXFET power transistors. Hybrid packaging of the dice can save weight and volume compared to standard packaging. Technical data and other information are available in the note. **Mfr: International Rectifier, 233 Kansas Street, El Segundo, CA 90245.**

MULTI-SWITCH BROCHURE

• A new 8-page brochure in full color describing Tini DW Multi-switch switches with 10 mm and 15 mm centers is now available. The new brochure discusses basic design features of the subminiature Tini DW Multi-switch, the first American made switch made on 10 mm centers. Switching functions, material specifications and terminations are described. Special features including available pushbutton legends and the new "flip-flop" pushbuttons are also covered. **Mfr: Switchcraft, Inc., 5555 No. Elston Ave., Chicago, ILL 60630.**

TAPE LINE

• A brochure highlighting the new line of magnetic tape from Sony has been introduced. The 11-page booklet, entitled "Sony Magnetic Tape," offers consumer-oriented information on how the company selects the raw materials and combines them to make a variety of tape formulations. It also explains how the tape, cassettes, reel transport mechanisms and even winding tension are prepared for final assembly. **Mfr: Sony, Magnetic Tape Division, 9 West 57th Street, New York, NY 10019.**

PRODUCT DIGEST

• A comprehensive 48-page catalog listing a variety of semiconductor devices in 13 major product categories was recently introduced. Condensed specifications include easy-to-use charts and tables for quick product selection. Several sections include schematic diagrams and dimensional drawings for many standard products. Special features of the new product digest catalog are a complete numerical listing of all products with corresponding data sheet numbers and a pre-addressed data sheet/quote request form. **Mfr: International Rectifier, 233 Kansas Street, El Segundo, CA 90245.**

CONNECTOR CATALOG

• A revised 18-page Adapta-Con connector catalog was recently reissued. The catalog, ACBP-6, features ten photographs and 30 drawings, information on UBS and UBC series cripm contacts, and other technical data. Special features of the catalog include standard information for material and finishes, mechanical features, and electrical data. It also includes instructions on how to order unshrouded headers, which electrically connect rigid PC boards with UBC cripm housing or ribbon cable socket connectors. **Mfr: ITT Cannon Electric, 666 East Dyer Rd., Santa Anna, CA 92702.**

TEST SYSTEMS CATALOG

• An illustrated, eight-page catalog describing a complete line of modular systems and accessories for static and dynamic burn-in, full function evaluation and environmental testing of integrated and discrete solid-state devices is now available. The two-color literature provides details regarding the new System Super 20 for complete performance evaluation of RAMS to 128K, the System 35D for dynamic burn-in testing of microprocessors and other IC's, the System 25 for reliability evaluation, the Systems TRH40 and TRH90 for temperature-humidity testing, and the System HD30 for high capacity burn-in. **Mfr: Marin Controls Company, 517K Marine View Avenue, Belmont, California 94002.**

AV PRODUCT CATALOG

• A new, 24-page full-color catalog is available at no cost from Visual Horizons, Inc. Approximately 250 products are explained and illustrated, including new products just developed. The catalog also contains a feature article, "Slide Shows Made Easy"—a complete course in planning and producing slide shows—as well as a full section on supplies to make overhead transparencies quickly and inexpensively. **Mfr: Visual Horizons Inc., 208 Westfall Rd., N.Y. 14620.**

INSTRUCTIONAL PA BROCHURE

• The "White Horn White Paper," the sixth addition to Electro-Voice's PA Bible, is now available. This addition deals with the advantages of using constant directivity horns in sound reinforcement applications. It discusses the major differences between constant directivity horn designs and also includes sample system configurations. **Mfr: Electro-Voice, Inc., Box 429, 600 Cecil St., Buchanan, Mich. 49107.**

METERS

• A brand new, 60-page four-color catalog listing the complete line of stock analog and digital panel meters, meter relays, controllers and test instruments. New products include the U.L. listed 260 Series 7-volt-ohm-milliammeter, the model 420 function generator, and the Model 454 oscilloscope. **Mfr: Simpson Electric, 853 Dundee Avenue, Elgin, Illinois 60120.**

METERS CATALOG

• A new catalog indexing a complete line of standard-range meters, special options and testers was recently made available to the public. Five different Shurite series include 2½-inch and 3½-inch types, and a 1½-inch edgewise series, available in AC and DC milliammeters, ammeters and voltmeters, and DC microammeters. The new catalog also lists such standard options as ±3 percent accuracy, custom dials and pointers, special ranges, zero suppression, and several special options. All series are illustrated, have dimensioned mounting drawings, and customary range and resistance values. **Mfr: Sigma Instruments, Inc., 170 Pearl Street, Braintree, MA 02184.**

db Editorial

DURING THE SUMMER MONTHS, we looked over various aspects of studio acoustics, construction and systems. From "Recording Studio Design and Acoustics" in June, to "Studio System Planning" in August, our authors emphasized the new; new design projects, new building plans, and new hardware systems.

But, not every engineer out there is in the market for "new." In fact, many of us are doing the best we can to cope with "old," and our studio updating is done piecemeal, rather than all-at-once.

Along the way, troubles may crop up. Yesterday's acoustic design may not satisfy today's tastes. Or, after several generations of hardware have come and gone, there may be some residual hums and buzzes left behind.

In broadcast audio, the cumulative effects of signal processing may be causing unnecessary deterioration. And, in sound reinforcement, perhaps the neighbors are beginning to complain, as the SPL creeps up.

In all these cases, it would be great to start all over again from scratch, but such thoughts are more often fantasy than fact. And so, this month's authors were asked to consider the fine art of "de-bugging" the existing systems, rather than creating new ones.

Replacing a tape recorder is simple enough—just wheel the old one out, and the new one in (not cheap, just simple). Replacing studio acoustics is not so easy. You can't trade in your used acoustics, nor can you return the new ones for credit if you're not satisfied. At Acoustilog, Inc., Alan Fierstein spends a good deal of time tracking down acoustic problems in the studio, and working out solutions to suit the unique requirements of the client. Drawing an analogy to the tape recorder, Fierstein recommends a specific sequence for acoustic tests and measurements. And, he cautions, don't wait until the job is done to see if it's proceeding in the right direction!

Grounding and shielding problems have plagued studio operators at regular intervals, especially if more than one pair of hands does (or, did) the wiring. It's worse yet if, somewhere back in the past, a "fix job" was done by the good-old "cut-and-splice" technique (1. Cut: more noise? 2. Splice: less noise? 3. Next wire).

As Albert B. Grundy points out, grounding and shielding are not the same thing. If some of your studio's grounds are shields, and/or vice versa, you may want to think about a complete re-wiring job. Actually, you may not want to think about it at all, but there'll be no avoiding it. Before starting the repair work, check our application note on "Keeping it Clean and Quiet": the often-ignored hardware listed there may go a long way towards cleaning things up.

In broadcast audio, "de-bugging" may be more a matter of reviewing compression settings, meter-reading procedures and the signal path in general. Vladimir Nikanorov takes a look at some of the points in the signal path where things may go wrong.

In sound reinforcement work, corrective maintenance may involve trying to keep the SPL out of the neighbor's living rooms. For the open-air amphitheatre, Michael Rettinger reviews the fundamental equations for horn design, and we discover that a bass horn with a sufficiently narrow coverage angle to keep the sound where it belongs, may be possible in theory, but highly-impractical in practice. A few baffles, strategically placed, is often a quicker and easier "de-bugging" procedure.

Concluding with a change of pace, we look in again on the design of audio pads, using a personal computer. Many readers responded to our last audio pad program (in May) with variations that include minimum-loss and bridged-T designs. These are presented here, and may be incorporated in the May program.

And speaking of complimentary subscriptions (which we did in May), if you have a computer program that you'd like to share with our readers, please send it in. For every one printed, we'll send out a "comp sub" to the person who submitted it. ■

ALAN FIERSTEIN

Acoustical Troubleshooting

Here it is, a thorough guide to acoustical troubleshooting that may hold the answers to some of your studio's problems.

WE ARE FREQUENTLY CALLED ON BY STUDIOS to come in and analyze rooms with various types of problems and, hopefully, to recommend a practical solution to whatever ails them. When first contacted, we get a brief verbal description of the problem. Often, it's a familiar one, though its cause may be unique. The standard test equipment (shown in Figure 1) is usually sufficient if no special tests are required. Fortunately, most studios' acoustic problems fall into a few basic categories. Unfortunately, time, money and aesthetics may demand different solutions to the same problem in different studios.

STUDIO TESTING

The first measurement we make in a studio is the Reverb Curve. This is simply a graph of reverberation time (correctly abbreviated T60) at different frequencies. We call a room "live" if T60 is long, or "dead" if it is short. But, is it a balanced T60? Do all frequencies decay at the same rate, or does the room seem to prolong some tones while eating up others? The reverb curve tells the story, and the appropriate treatment is determined from calculations. Actual on-site measurement of the reverb curve *is* necessary. Too often, we have been called into a room where only calculations were used to predict the final outcome, yet due to variations in materials and construction, the desired response was not achieved. It is alright to predict by calculation, but only when small corrections are to be made. Actually, we prefer to measure every step along the way, because this is the only way to monitor the progress, and effectiveness, of the correction. For example, if a large amount of bass absorption is needed, it makes good sense to test the effect of adding a few bass absorbers, than to inadvertantly deaden the room too much (and add unnecessary expense).

Different room areas, and musical tastes dictate the required reverb curve. Some people like live drum booths without bass buildup. Others like dead drum booths that can double as vocal booths. And many times we are asked to recommend a treatment that will make one room identical to another (sometimes the competition's)!

The price paid for a live room is lack of separation between instruments. Obviously, if a room is live in the low end, a lack of low frequency separation is to be expected. If a room is to be designed with many reflective surfaces, we of course know that parallel walls should be avoided, or flutter echo will result. It is not so commonly known that diagonally opposite corners of a room can also cause severe flutter echo. Flutter echo of this nature has a unique "galloping sound" which you have to hear to appreciate. Not only is this flutter particularly annoying, but the room corners appear to catch sound that comes from almost anywhere in the studio and set it into action. We have observed this problem in more than one studio, and have developed a technique to determine the exact surfaces involved.

Suppose that a flutter echo is heard in the room shown in FIGURE 2. We examine the room surfaces and rule out reflections from walls A and B, due to their absorptive surfaces. If we also assume that the absorptive floor and ceiling combination is not to blame, then there are just two possibilities left: walls C and D, or corners AC and DB. If absorptive material is placed in corner DB, and the flutter is suppressed, fine. If, however, it is impractical to do the test in this manner, then instrumentation is employed. A microphone—placed in corner AC,—feeds a logarithmic amplifier and triggered oscilloscope. FIGURE 3 is taken from a 'scope display of a typical flutter echo created by a hand clap near the microphone. The time between peaks is 108 milliseconds, and the (round trip) distance that sound travels during this interval is 122 feet. This indicates that the diagonal, measured to be 60.5 feet, is the culprit. As you can see, this is a precise and powerful way of positively determining the problem-causing surfaces in a room

Alan Fierstein is President of Acoustilog, Inc.

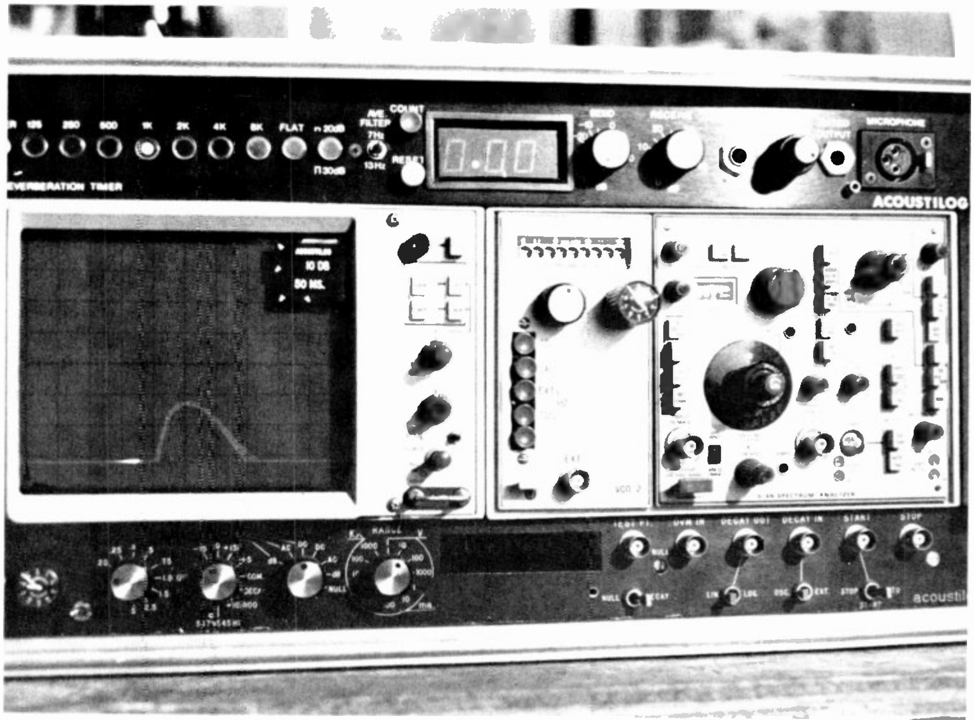


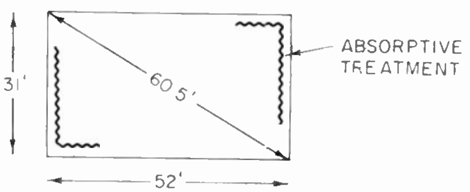
Figure 1. Typical equipment used for acoustic troubleshooting, including (top to bottom): Reverberation Timer, Dual-trace Storage Oscilloscope with TDS, calibration/control center.

Even if walls A and B had also been suspect, the distance calculation would still conclusively lead us to the prime offender.

Excessive ambient noise is often present in studios. Even though most popular recording may be done at a high level, where loud room tone isn't a problem, the noisy studio is still a source of embarrassment to the studio owner, and it becomes a real problem when using distant miking. Typical noises that plague studios come from air conditioners, footsteps from the floor above, subway and truck traffic, and of course the studio next door.

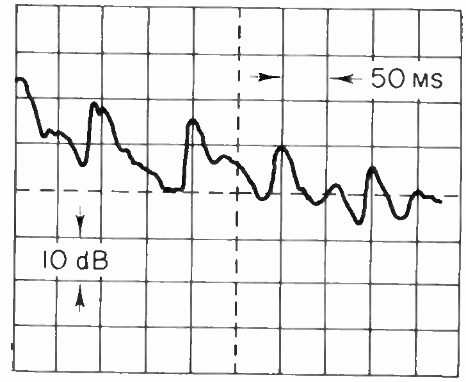
A spectrum analyzer is useful for tracking down noises, as well as checking the degree of improvement after the recommendations have been followed. For example, air conditioning velocity noise shows up as a low or mid-frequency noise. By enlarging the cross sectional area of the duct, the velocity noise decreases and the noise shifts downward in frequency, becoming less objectionable. The spectrum analyzer displays this frequency shift and allows the determination of the Noise Criteria (NC) curve.

Figure 2. Top view of a room with a corner-flutter echo.



Microphones furnished with some popular real-time analyzers are hopelessly inadequate for the task of measuring studio noise levels. When placed in a sound field of less than 25 dBA or thereabouts, they generate so little output signal that it actually gets swamped out by the microphone's own electrical noise. Add-on microphones are available that solve this problem. Our system consists of a 1-inch Bruel & Kjaer capsule and an Ivie IE-30A one-third octave analyzer. Using this microphone, we measured one room with a noise level of only 11 dBA, and the only noise you hear is the ringing in your own ear.

Figure 3. Logarithmic output of reverb timer shows pronounced flutter peaks every two divisions.



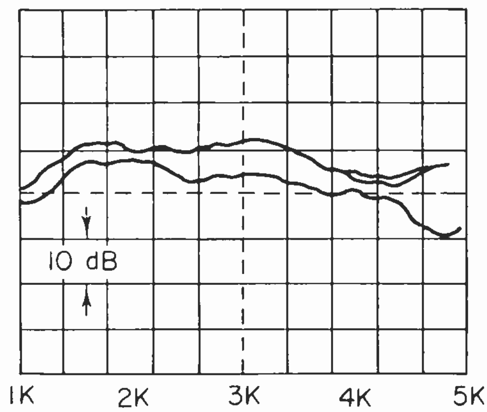


Figure 4. Three TDS sweeps of (top to bottom): Floor reflection, ceiling reflection, foam rubber reflection. The two top sweeps are almost identical, hence they appear to be only a single trace. (Note non-logarithmic scale.)

CONTROL ROOM TESTING

When a live microphone in the studio causes feedback through the control room window, the spectrum analyzer again comes in handy. With an accelerometer placed on the glass, a light tap on the window will show the glass' resonant frequency. If this corresponds with the fundamental feedback frequency, it may be a good time to consider new glass. The wall construction, floor slab, and speaker mounting method may be to blame for feedback, too. Hopefully, the air ducts are not carrying the sound, but this can be easily determined by systematic probing, while observing the analyzer during controlled feedback or by energizing the control room monitors with pink noise.

Some of the other tests required for troubleshooting control rooms have been described under Studio Testing. However, the control room has a different purpose, so in many cases the parameters must be measured or interpreted differently. While a studio is designed to stimulate and enrich a musician's output, a control room should be designed to provide a similarly-hospitable environment for the mixer and the equipment. Poor acoustic conditions in the control room are a serious handicap to the mixer. However, what constitutes poor control room acoustics is one of the current "great debates," as **db** readers are aware. Because so many conflicting articles on control room design have appeared, studio people who look to current literature for acoustical facts often get opinions instead. For this reason, when analyzing a specific control room problem, good practice demands a complete measurement and analysis of the room. It may be that a really serious problem was not noticed, or that a supposed acoustical problem has a completely unexpected cause. Incidentally, we have found it advantageous to have studio personnel watch and participate in the measurement process. This way, the engineers gain a better understanding of the room they must work in, and fully understand the reasons for making the acoustical changes that are recommended. An oscilloscope display is more convincing than an "expert" opinion.

The reverberation curve of a control room is measured in a similar manner to that of a studio, but there are a few important differences. First of all, the monitor speaker is used as the sound source. The monitor loudspeaker's characteristics and its position in the room will influence the apparent reverberation time, and this is an important parameter in a control room with a permanent monitor position. Secondly, while studio reverberation time is measured in the reverberant field, control room T60 is measured behind the mixing console. Although other positions can be tested while exploring for room modes, the room must be designed with the primary listening position optimized. Finally, while the reverberation curve of the studio is designed to create wonderful and perhaps unique sounds, the control room should bear a strong resemblance to reality, acoustically speaking. Reality, in this case, means acoustical conditions similar to that of the average home listening room. The average home listening room has a reverberation curve that is reasonably flat, with a slight rise at the low end and a droop in the treble region. Contrary to popular opinion, a very dead room usually has a poorer response characteristic, due to the low number of averaging reflections. It is actually easier to provide a wide uniform listening area in a very live control room than in a fairly dead room. The trade-off with a live room is the possibly decreased stereo imaging and less detail in the actual sound being monitored. So, a reasonable compromise must be made.

As stated earlier, formulas and past experience may predict the amount of absorption that should be designed into the room, but the effect of differences in materials and construction methods will not show up in "before-the-fact" calculations. To repeat, the progress of a room's acoustical balance should be monitored *during* construction, and of course after. Since this procedure is seldom followed, acoustical troubleshooting demands that the reverb curve be measured as a first step.

The reverb curve represents factors which, if changed, will affect other parameters of the room, notably equalization. The reason for this interaction has been discussed in a previous article (see reference 1), but it basically centers around the fact that real-time analyzers employ time integration to provide a readable display, and this time integration captures multiple reflections and cannot distinguish them from the direct sound. Even if the reflected sound level is too low to significantly affect the real-time analyzer display, the effect of a non-flat reverb curve is easily detectable by the trained ear. So it pays to treat and correct the reverb curve first, and then as the other measurements and corrections are applied, they will not have to be redone. An appropriate analogy can be made between rooms and tape recorders. When aligning a tape recorder, both azimuth and equalization affect frequency response. However, if one makes the mistake of adjusting equalization without adjusting azimuth, the tapes recorded on that machine will sound different when played on a correctly adjusted machine. The key here is that the azimuth is adjusted first, and then the equalization. In this way, the equalization does not have to be redone. The same thing applies to rooms: The reverb curve is adjusted first, and only then is corrective equalization applied, if warranted.

When engineers complain about harshness, distortion, too much top end, etc., a possible cause is an uneven reverb curve, or, they could be hearing strong reflections in a particular frequency band. The impulser method described earlier in Studio Testing, is a good test for strong reflections. It is especially appropriate to test this because, for efficiency reasons, most popular control room monitor speakers have poor dispersion characteristics. The high frequencies are often beamed within a very narrow angle, and this doesn't mean on-axis. The dispersion characteristics of the speaker must be known to the acoustical consultant so that appropriately-placed absorbers and diffusers can be installed to break up strong, interfering reflections.

When called in to remedy an existing situation, the types of surfaces must be familiar to the acoustician if he is to intelligently recommend remedial measures. Testing materials used to require the use of a quiet, open field, or an anechoic chamber, along with a large sample of the material to be tested.

Time Delay Spectrometry, or TDS, offers a powerful testing method that can actually be brought into the room being tested. Simply described, TDS combines the precision and repeatability of a sine wave sweep with a frequency-shifting technique to separate, for easy viewing, the multitude of reflections that are typical in rooms. One can view the spectrum of the direct sound, or that of any reflection. The change in the shape of the spectrum before and after encountering a particular surface allows the evaluation of that material's absorption coefficient. However, one caution must be noted. TDS can deliver too much precision to a measurement. For example, a TDS sweep may contain many horrendous-looking notches that are actually inaudible, due to their narrow width. Looking at a room with TDS only is equivalent to studying a street map with a microscope. Care must be taken when interpreting TDS curves. (See reference 2.) FIGURE 4 shows the TDS sweeps of three surfaces: a hard floor, an acoustically-sprayed ceiling panel, and a piece of 1-inch low-density foam rubber. The negligible difference between the floor and the ceiling panel curves indicates that the ceiling spray is completely useless as an absorbing material. The lower sweep shows the foam's absorption to vary between 5 and 10 dB.

Before any serious study of the monitor speaker response can be started, the driver polarity must be checked. It is common knowledge that the effect of two out-of-phase woofers is a thin bass sound. The effect of reversed tweeters is a blurred stereo image. Also, the monitor with the reversed tweeter will exhibit an abnormal frequency response. Since monitors may have fairly uneven responses, even when correctly polarized, the easiest way to test polarity of the tweeters is with the impulse test. At the same time, the time displacement between the woofer and tweeter can be seen on the oscilloscope. It is interesting to observe the degree of alignment achieved by monitors that have been designed with that goal in mind.

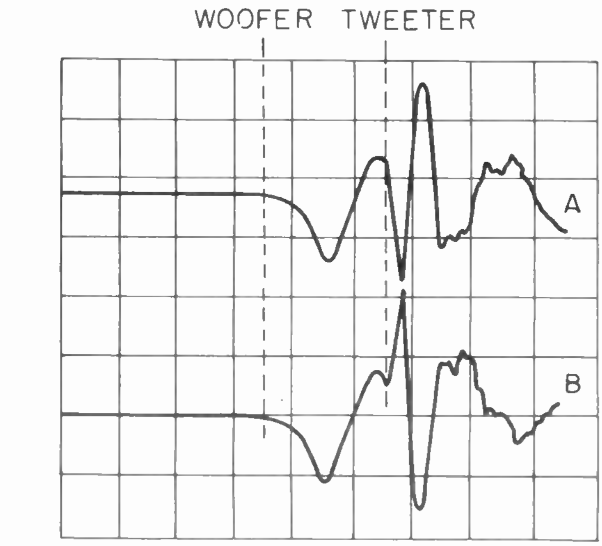


Figure 5. Received impulse waveform from a two-way speaker, showing the effect of flipping one driver's leads. Note the time misalignment between drivers.

FIGURE 5A shows the misalignment of a popular monitor speaker. Reversing the phase of one of the drivers yields the trace in FIGURE 5B. Notice the peak which has reversed direction.

Improper loudspeaker placement frequently causes response problems. Time-delayed reflections from various surfaces combine with the direct sound, causing a frequency-dependent phase shift. Therefore, the steady-state response, as measured on a real-time analyzer, may be modified with nearly 6 dB boosts, nearly infinite dips, or anything in-between. For example, the room curve shown in FIGURE 6 has a serious notch at 100 Hz. An interesting way to test if a time-delayed reflection is causing the problem is demonstrated in FIGURE 7A, which shows the oscilloscope screen in the dual-trace mode. The upper trace is from a microphone in front of the loudspeaker, and the lower trace is taken at a particular point in the room from which a strong reflection reaches the listener. The setup is shown in FIGURE 7B. Notice that the intensities are nearly identical, but the waveforms are almost 180 degrees out of phase. The excitation signal is a 100 Hz sine wave. The time delay causing this phase shift is therefore about 5 milliseconds (i.e., half the period of a 100 Hz sine wave). This can be seen on the oscilloscope as the time interval between peaks.

This 5 ms. time delay is caused by the reflection traveling over a longer path length than the direct sound. Because these antithetic waveforms are at a similar distance from the mixing position, the listener seated there will hear a serious dip in the response at 100 Hz. The solution is to find a better position for the loudspeaker, or change the geometry of the room. In this case, the room had to be entirely rebuilt.

Before discovering the cause of the notch pictured in FIGURE 6, equalization had been attempted to bring up the level of sound in that band. At high SPLs, the woofers were being damaged because of the excessive energy being fed into them at 100 Hz. Remember, a 6 dB boost requires that four times the power be supplied to the speaker.

Frequency response problems and phase reversals are not always acoustical in nature. The acoustician should be well-versed in the art of a complete electro-acoustic checkout from the console to the loudspeakers, before assuming that acoustical corrections are needed. If monitor equalizers and console monitor outputs require termination, terminate them. Otherwise, transient response and frequency response will suffer. When proper levels are set throughout the monitor chain, there should be no audible hum or noise at the mixer position. What is frequently mistaken for speaker distortion is really an incorrect matching of headroom levels in the monitoring electronics. For example, we frequently encounter situations where the input level controls on the monitor amplifier are set at the 12 o'clock position, which is 20 dB below

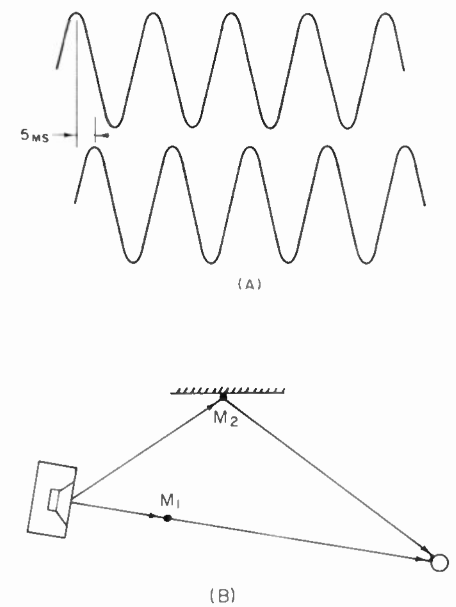


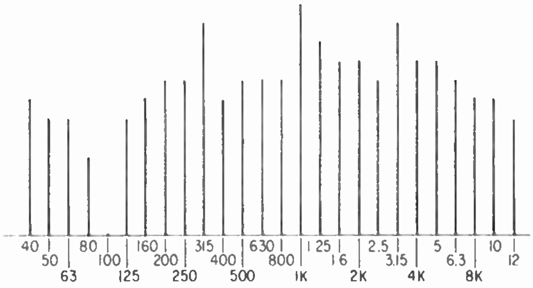
Figure 7(A). Two microphones simultaneously displaying waveforms at two places in the room. The almost-180 degree phase shift for the notch seen at 100 Hz in Figure 7(B) Mi is a microphone in the direct path from the loudspeaker to the listener, L. Mz is a microphone placed on a reflective surface. Sounds at Mi and Mz are equidistant from the listener, but have travelled unequal distances from the original sound source. Therefore, the listener will hear interference effects.

maximum sensitivity. This requires that the console monitor output or the monitor equalizer supply 20 dB more signal than normally required to drive the amplifier to full output. If the amplifier sensitivity is 2 volts, which is equivalent to +8 dBv, the equalizer had better be able to handle +28 dBv or it will distort before the amplifier has reached its full output. The solution in this case is to set the amplifier level controls higher, so that amplifier power becomes the ultimate limiting factor, as it should be.

CONCLUSION

Acoustic troubleshooting requires both a variety of standardized tests to ensure correlation with the rest of the real world, and specialized tests to fit the demands of each studio's unique problems. A good troubleshooter also has the experience to evaluate test results and correctly diagnose difficult problems. Finally, if practical solutions are to be achieved, cooperation and understanding between the consultant and the studio personnel will be necessary. Solid scientific testing, combined with subjective listening tests, will go far toward achieving that goal. ■

Figure 6. Received frequency spectrum in a control room with a serious notch at 100 Hz.



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1. Alan Fierstein "The Equalization Myth," **db**, The Sound Engineering Magazine, August 1977.
2. Don & Carolyn Davis "Time, Energy, and Frequency Measurements for Sound Definition," **db**, The Sound Engineering Magazine, June 1980.

ALBERT GRUNDY

Shielding and Grounding Revisited

Recognizing the difference between shielding and grounding may reduce hum and eliminate potentially "electrifying" results....

TO ANYONE with even the most casual experience with audio equipment, "grounding problems" always seem to mean some sort of interference, usually hum from power-line-related frequencies. Sometimes, hum reduction and personal safety seem to be an "either/or" proposition; however, our electrical codes and manufacturing practices make safety the first consideration.

We still run the risk—no matter how small it may appear—that two pieces of improperly-grounded electrical equipment could have a sufficiently high potential difference as to become lethal. (See Joe Dollar's letter to the editor, in the August issue, and John Nady's wireless microphone feature in last month's **db** for more on the subject—**Ed.**)

The purpose of a grounding system is to insure a zero potential between all exposed metallic surfaces. These must also be at zero potential with respect to earth ground. The grounding system—safe or otherwise—is *not* intended to be a noise reduction system. Shielding will protect against interference from outside noises—grounding won't. It's only when the terms "grounding" and "shielding" are used synonymously that trouble starts.

GROUNDING

In relatively small installations, with say, 20 or fewer line cords, a common ground can be established using the third (green) conductor and the ground pin of the power plug. These should be plugged into electrical outlet strips that have their third-pin sockets wired together. The line cords from these strips should be plugged into three-wire wall receptacles, for which it is *known* (not guessed) that the ground pin is wired back to a true earth ground. If possible, they should be distributed between the phases of a multi-phase service, to minimize the neutral current.

Larger installations must be handled differently, and in fact should be independent of the three-wire power grounds. Some studios have clipped the ground pins on equipment power plugs, or used 3-to-2 adapters without connecting the adapter's green wire to ground on the receptacle box. Although this may reduce hum from ground loops, it is potentially dangerous (pun and all), and violates most electrical codes.

Albert Grundy is President of The Institute of Audio Research, NYC

A safer—and legal—method is to use special AC receptacles in which there is *no* continuity between ground and the metallic outlet box. Just like the neutral and "hot" connections, ground appears at a screw terminal on the receptacle, and nowhere else.

From each ground terminal, a *separate* #10 insulated wire is run to a central grounding point, such as a copper bus bar. This may be about two-inches wide, and long enough to allow all the ground cables in the system to be bolted to it with ¼-20 bolts and star washers. This bus bar is firmly connected to a good earth ground.

If a true earth ground is not available (for example, in a high-rise building), a cold-water pipe may be used instead. Drain pipes should probably be avoided, since sooner or later a section may get replaced with PVC tubing.

With a good grounding system, there will be zero potential difference between any two metallic surfaces (tape recorders, console, equipment racks, etc.) in the studio. From the point-of-view of safety, the job is done. However, from the point-of-view of hum (and noise) immunity, the job is just beginning.

Now, the audio cabling between each component in the total system must be wired in such a way that no additional ground paths are established in parallel with the hard-wired AC ground system just described. Such paths create the infamous "ground loops" that cause so many hum problems in random—and casually-grounded-systems.

In searching out ground loops, the culprit usually turns out to be the shield on a cable connecting two pieces of equipment (for example, tape recorder-to-console). If the shield is connected to ground at both ends, a parallel ground is created. The double-ended connection doesn't enhance the shielding properties at all, but it certainly does enhance the production of hum.

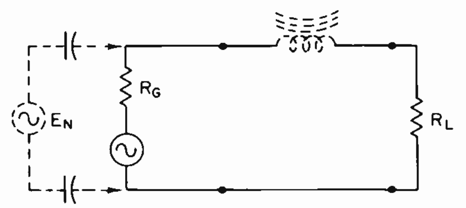
To keep the shield from becoming a ground, make sure it is connected at one end of the cable only. A good practice is to connect all shields to ground at all equipment outputs, and to make sure the shields are not connected where the cable is plugged or wired to an input. In other words, connect the shield to pin 1 within all female plugs, and make sure the shield is *not* connected to pin 1 within all male plugs.

Once all the audio cabling is in place, you can make sure the shielding is in order by disconnecting the power-line grounds at the central grounding point. Now, there should be infinite DC resistance between each separate metal chassis within the system. If there is not, then one of the inter-connecting audio cables is probably at fault. It should be located and corrected. Once a cable has been corrected, re-connect the other cables one at a time, to make sure the system remains in good order.

Removing ground loops from an existing random-ground system is an almost-hopeless task. The general rule seems to be that "every stop-gap solution is just another problem." It's usually better to strip the system and re-wire. Anything else may be just a waste of valuable time.

SHIELDING & ELECTROSTATIC COUPLING

Unwanted or interference signals enter the audio system by either electrostatic coupling or electromagnetic induction, as seen in FIGURE 1.



The dashed-line circuit on the left symbolizes interference voltages, capacitively coupled to the transmission loop. If the system's source impedance, R_G is very low, the noise voltage across it will be kept at a minimum.

The dashed-line inductance symbolizes the introduction of an induced current. The high load impedance, R_L minimizes this current....

Electrostatic, or capacitive, coupling is most common at what are generally classified as radio frequencies; these include TV, CB, static, ignition, etc. The interference voltage is coupled in parallel with the audio transmission loop, and therefore has minimal effect when the source impedance is low, as in most modern constant-voltage equipment.

ELECTRO-MAGNETIC INTERFERENCE

The magnetic fields found around audio equipment are usually from power cables (60 Hz), motors and transformers (120 and 180 Hz). The electro-magnetically induced interference appears in series with the audio transmission loop, and is therefore greatest in a low-impedance circuit. However, the high input impedance of constant-voltage equipment provides a high total loop impedance.

To summarize, in an audio transmission loop, the low source impedance minimizes capacitively-coupled high frequency interference voltages, and the high input impedance minimizes magnetically-induced low frequency interference currents.

The parallel ground circuits discussed earlier are very low impedance circuits, and are therefore susceptible to low frequency interference only. A ground loop acts as a single-turn secondary for any nearby power transformer. The induced current causes a potential difference between the two pieces of equipment, which is converted to a signal voltage at the input resistance in the receiving end of the loop.

The most common method of protection from electrostatic interference is, of course, shielding. And the most common method of creating ground loops is to connect those shields at both ends. ■

db Application Notes

Keeping it Clean and Quiet

FOR RECORDING ENGINEERS, AC power line hardware is certainly not as fascinating as the latest generation of recording consoles and signal processing devices. Yet, the success of all that "high-glamour" stuff often depends on low-glamour AC, and a noisy power line can cause a lot of grief in any "hash-prone" control room.

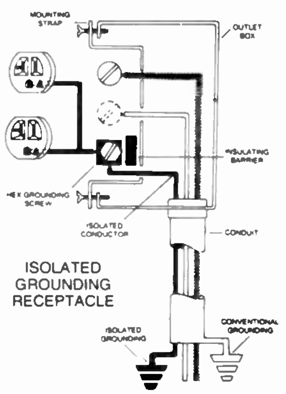
A spec sheet from Topaz Electronics cites common-mode noise (occurring between ground and both the hot and neutral conductors) as a frequent source of problems for computers and other noise-sensitive gear which use common (i.e., power-line ground) as a reference. Also, protection from high frequency components of transverse-mode noise (between neutral and hot) may be needed.

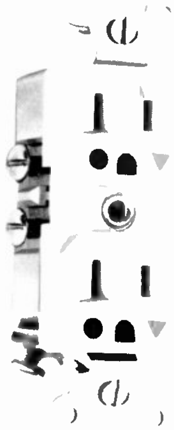
According to Topaz, a line-noise suppressing "Ultra-Isolator" will provide up to 150 dB common-mode, and 40 dB transverse-mode, noise attenuation from various power line disturbances.

ISOLATED GROUNDS

For a safe (and legal) grounding system using wall-mounted AC receptacles, an isolated-grounding receptacle, such as Slater Electric's IG-8200, may be used. As described in Albert B. Gundy's feature in this issue, the grounding terminal is *not* connected to the receptacle's mounting strap, nor should the regular AC ground (green wire) be connected to the receptacle. Instead, the green-wire ground is connected to the metal outlet box, and a *separate* ground wire is attached to the receptacle.

For dimly-lit areas, illuminated receptacles are also available (but without the isolated-grounding feature). Two long-life neon bulbs are imbedded in the receptacles, making it easy to find without turning up the house lights. (Just the thing for screening rooms and dimly-lit sessions of any kind.) ■





Superficially resembling a conventional AC outlet, the orange triangle next to the grounding receptacle identifies the isolated-grounding feature of this receptacle (Slater IG-8200).

Note the separate isolated-grounding wire, which is in addition to the conventional grounding system.

(photo and line drawing courtesy of Slater Electric, Inc.)

Isolation transformers are available to handle loads with power ratings from 125 VA to 130kVA(!). Some studios use small isolators to keep troublesome spike-producing control room equipment from introducing noise into the studio AC lines. Shown here is a Topaz 91095-32 Ultra-Isolator with a 500 VA power rating. The transformers are also available with terminal strip connections, for hard-wired installations. (Topaz photo)

VLADIMIR NIKANOROV

Cleaning Up the Sound of Radio

Given the competitiveness of the broadcast field, reading these helpful hints might prove to be a "sound" investment of your time.

IN THE PRE-TELEVISION ERA, broadcasting simply meant mono-only AM radio. The important word was "format," and "sound" didn't get that much attention. (Life was much simpler then.)

As television began to attract attention, something needed to be done to capture—or rather, to re-capture—the radio audience. The "top-40" format, devised in the '50s by Gordon McLendon, Todd Storz and G. Bartell, played a key role in revitalizing broadcast audio in the face of the new competition from video.

When FM radio came along, it was at first ignored by mass-market broadcasters, who regarded it as a limited-interest medium, best suited for classical music and talk shows. But eventually, someone got around to playing rock-and-roll on FM, and the rest is history. Today, FM as well as AM stations compete for mass-market attention, and each station's "sound" is as important as its format. (See Eric Small's "Measuring the Sound of Radio" in last month's **db**—Ed.)

As that sound takes shape in the broadcast studio, all sorts of things can happen to it (and usually do, according to Edsel Murphy—Ed.). Consoles and compressors, tape recorders and transformers all play a part in influencing what the listener eventually hears. That influence should be positive, yet often it's not, due to a variety of causes.

BROADCAST VERSUS RECORDING STUDIO PROCEDURES

This article will look in on a few of the places where, "If anything can go wrong, it will" (Murphy again—**Ed.**) But first, let's take a brief look at some of the constraints under which the broadcaster must operate. Although broadcast and recording studio operations are similar in many cases, there is at least one fundamental difference. The broadcaster is subject to the rules and regulations of the FCC. One of these is "Proof of Performance"—a number of well-defined procedures, among which are checkout and alignment of frequency response, harmonic content, carrier shift, noise level, etc.

For the procedure, the FCC requires a test setup as if the radio station was under normal operation. The measurements include all circuits, from the microphone or other source input, to the output of the modulation monitor. (FIGURE 1) This includes pre-emphasis circuits, telephone or microwave lines, antenna circuits, etc. However, compressors should be excluded from the measurements.

Frequency response is measured at 50 Hz, 100 Hz, 400 Hz, 1,000 Hz, 5000 Hz, 10,000 Hz and 15,000 Hz. Harmonic distortion is measured at 25, 50 and 100 percent of modulation level. FM station measurements are made with the 75 microsecond pre- and de-emphasis networks in the chain. (FIGURE 2)

Experience has shown that these performance measurements are irrelevant to the character of a station's "sound." Broadcast facilities with identical parameters at the station may sound completely different at home. The difference may be characterized by transparency of sound, the stereo image, and—almost always—by loudness.

According to ANSI (American National Standards Institute) Standard SI-1-1960, "Loudness is the intensive attribute of an auditory sensation in terms of which, sounds may be ordered on a scale from soft to loud." (Or, in English, "Loudness is a measure of a sound's intensity"—**Ed.**) A popular opinion in broadcasting is that the listener prefers auditory sensations

Vladimir Nikanorov is technical director of Bonneville Broadcast Consultants, Tenafly, New Jersey

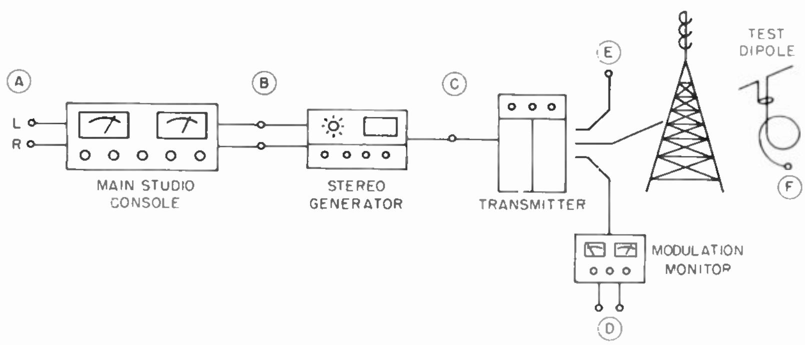


Figure 1. A typical FM station block diagram, showing test points for Proof-of-Performance tests. (A) console inputs; (B) console output(s); (C) stereo generator output; (D) modulation monitor; (E) transmitter RF; (F) field tests.

(that is, sounds) that are as loud as possible. Such sound—oops, auditory sensations—are presumed to attract attention to the station transmitting them.

COMPRESSORS

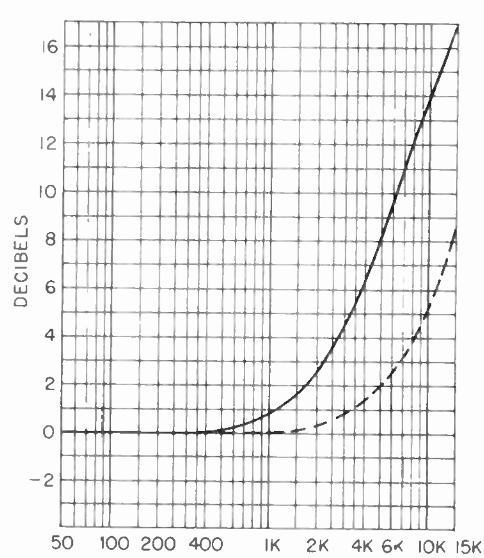
For a broadcast engineer, a louder sound means a higher level into the on-air compressor. Most compressors which are currently used on-the-air are of the "above-threshold" type, with adjustable input level control. Below the compression threshold, no dynamic gain-changing takes place. This is in contrast to linear compression, where the dynamic range of the total signal is processed, regardless of the input level. (The designations "linear" and "above-threshold" are used by dbx to define these modes of compression. For further details, see AES Preprint 1505: "Above Threshold Compression with One Control" by Leslie Tyler—**Ed.**)

Originally, compressors were designed, and employed, to limit dynamic range to manageable proportions in order to transmit the sound without transmitter overload or excessive noise. It was quickly found that increased levels into the compressor bring up RMS levels, and therefore, loudness.

As a result of compression for the effect of loudness, the on-air audio goes through several fundamental transformations. With heavy compression, the gain change caused by a series of sudden high-level passages will always be audible, no matter how good the compressor is. For example, if the compression ratio is 20:1, then the change in output level for an input signal that is 2 dB above threshold will be 0.1 dB. For an input signal of +20 dB, the output level will be +1 dB. Audible changes of this sort can become especially intolerable if a single broad-band compressor processes, say, a well-modulated timpani or bass drum with an orchestral background. The high-level peaks will punch holes in the sonic "picture."

Yet another change of audio signal also occurs when a peak is applied to a compressor. The gain-controlled amplifier and its side-chain receive the peak simultaneously. For the first moments—until the side-chain is activated and the control signal changes the amplifier gain—the peak is going through full-blast. The phenomenon is well-known as overshoot.

Figure 2. The standard 75 μs Pre-emphasis curve (solid line). The Dolby 25 μs curve is also shown (dashed line).



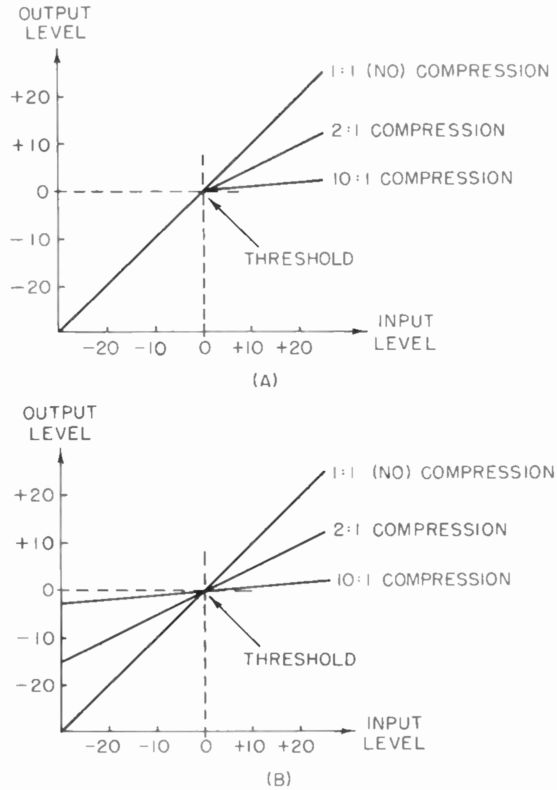


Figure 3. (A) Characteristic compressor curves in the above-threshold mode; (B) Characteristic compressor curves in the linear mode. Here, dbx uses the term 'threshold' to identify the point (unity gain) which separates downward and upward compression. (Both graphs from AES Preprint 1505.)

At this instant, depending on the combination of input level above threshold, attack- and release-times and signal spectrum, there will be several interesting transformations of the audio waveform. For example, if the peak fronts rise considerably faster than the compressor's attack time, then high-frequency content may pass completely unattenuated, bringing up sibilance. However, faster attack times result in ringing and audible fluctuations of peak levels. Examples of both are shown in FIGURE 4.

Release time determines speed of the gain change, moment-by-moment. Under heavy compression, a release time can be chosen so that low and medium level passages are raised to the maximum levels, causing considerable increase in loudness. Such a combination also causes considerable change of the original wave form and presents a significant, and negative, difference in sonic quality. FIGURE 4 also shows sine wave bursts that are subjected to heavy compression with different release times. As the release time is shortened, loudness is increased, but sound pulses become abrupt (pumping) and noise of low-level passages will modulate the audio (breathing).

In addition to sibilance as a result of excessive overshoot, and the sudden changes of level (ringing, pumping and breathing) which accompany fast release times, the heavy compressed audio creates another set of problems in the broadcasting loudness race.

The AGC stage of a conventional broadcast compressor may create overshoots up to 30-to-40 percent. The audio-plus-overshoot enters the following (limiter) stage. The limiter reacts to the full level, thus reducing the audible signal to a level considerably lower than it should be. For example, if an audible peak in the limiter is +10 dB, with an additional 3 dB of overshoot, then the level which will be reduced is 13 dB.

In turn, the limiter creates its own overshoot level, which is monitored by the modulation monitor. The overshoots, mostly inaudible, trip the monitor at 100-percent modulation level, leaving the audible signal below the limit of modulation. Now, the station is not "loud" enough: compression is increased, and things go full circle. With increased compression, the meaningful audio is held down by inaudible overshoot on the modulation monitors, and the desired loudness again calls for *more* compression. This is what happens if the overshoot is not controlled.

Unfortunately (for the listener), overshoot is often controlled by clipping. Some designers glibly label their clippers "instantaneous limiters," which they certainly are. Clippers cut off excessively fast peaks at some pre-determined level, regardless of the initial wave front. Clipping, in combination with heavy compression, virtually guarantees that overshoots will be audible, by adding harmonic content to the audio signal. For example, clipping of just over one percent becomes audible with classical or semi-classical music. A listener's sensitivity to clipping also increases with decreasing loudness, which makes home listening very critical.

All these ills would have less effect on listening if the compressor was only used as initially intended; the dynamic range of the music would be left to the artistic judgement of the performers and record producers.

It is unlikely that broadcast compression will soon (or perhaps, ever) return to this simplified state. However, a periodic review of compression procedures should be given as much attention as the required Proof-of Performance tests. In the recording studio, the engineer gives each compressor a lot of attention, as he strives to create a competitive "sound." In the broadcast studio, the compressor may fall into the "set-it-and-forget-it" category. Although moment-by-moment knob-twiddling is neither practical or possible, the compressor could possibly stand a little more attention than it has been getting.

METERING

VU meters with an integration time of 300 milliseconds don't truthfully show the levels of peaks which happen to be shorter than 300 milliseconds. Sometimes, when two programs look about the same on the VU meter, but somehow sound different, the chances are that the peak content of the two is different. In this circumstance, the difference is usually minimized by compression—loudness is just an extra benefit. If the desire is to avoid consistency in levels without introducing heavy compression, a combination of VU and peak-program meters can be effectively used for monitoring the levels.

A 1,000 Hz sinewave which reads 0 VU should read -6 dB on the PPM. Then, program peaks of 300 milliseconds or longer will show the same levels on VU and PPM. Faster peaks will remain basically the same on the PPM, and will show lower levels on VU meters. The configuration helps to avoid overloads and to control average-to-peak ratio of program material, which is the key to consistency.

TEST TAPES

There is no written standard for setting levels in a broadcast production studio. Ampex test tapes use a reference fluxivity of 185 nWb/M. Magnetic Reference Laboratory uses 200 nWb/M, and the DIN standard—widely used in Europe—is 320 nWb/M. Tape recorder output level may be anywhere from 0 dBm to +8 dBm, or higher. Good engineering practice calls for measurements of head room and signal-to-noise in order to determine the proper levels. In the US, +4 or +8 dBm is commonly used as line level. In Europe, the accepted level is +6 dBm.

CONSOLES

Published specifications of broadcast consoles may give few clues as to how the device will sound. For example, information about slew rate is often omitted. However, slew-induced distortion causes intermodulation products, which degrade sonic quality. To prevent objectionable results, slew rates of amplifiers should measure about one volt-per-microsecond, for each output volt.

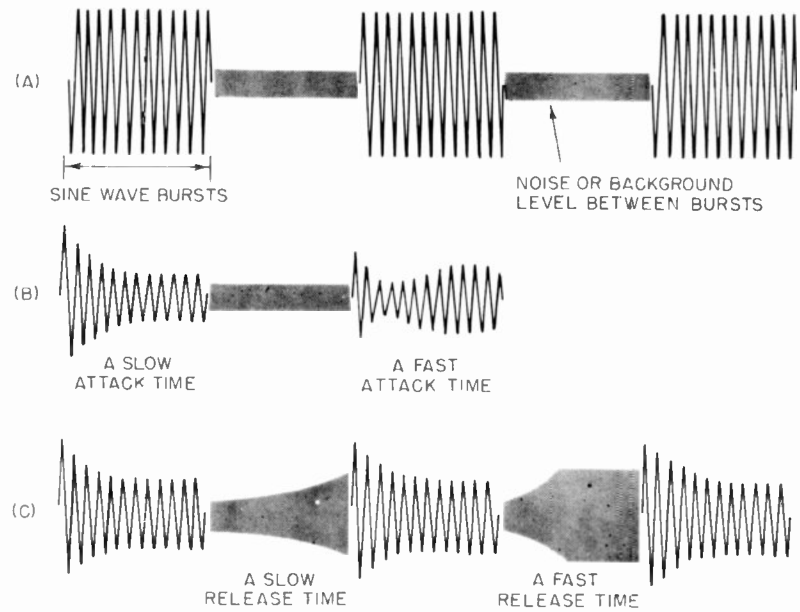


Figure 4. (A) A series of sine wave bursts; (B) The effect of varying the attack time; (C) The effect of varying the release time, after heavy compression.

Since broadcasters deal with pre-recorded material, which is a number of generations removed from the original master tape, some will claim that the quality of broadcast electronics can be lower than that which is necessary for the original recording. But since these recordings have already been through a number of transformations, it is absolutely necessary to preserve whatever transparency, aliveness, presence and clarity is still there. Therefore, the broadcast console must take advantage of the latest state-of-the-art technology, rather than relying on "second-best" or worse.

TRANSFORMERS

Many people feel that one of the major sources of audio degradation is the transformer. It is cited as a cause of linear and non-linear distortion, poor phase response, hum pickup and sonic coloration.

The use of transformers in broadcast production and on-air operation may be particularly ill-advised, since they overshoot and ring like any other inductance. Transformer specifications are usually presented for high-level conditions, but it is at low levels (−10 dB and lower) where performance becomes critical. Microphone preamplifiers are especially critical, yet transformerless designs are still uncommon in broadcasting. However, it is well-known that a transformerless preamplifier can be easily designed, based, for example, on the LM 394 integrated circuit, with monolithic super-matched transistors. The Trans-Amp transformerless microphone preamplifier is commercially available (from Valley People, Inc.), and is a part of recording consoles manufactured by MCI, Tangent, Harrison, Soundcraft and others. Most of these consoles feature the complete absence of transformers.

CONCLUSION

While this article is by no means a step-by-step guide to debugging the broadcast studio, it does point out a few areas in which a closer scrutiny may reveal the need for a little corrective maintenance. In many cases, sources of audio deterioration are identifiable, and therefore, curable. ■

Remember that a test tape (or, for that matter, *any* tape) does not have a "plus-four" (or "plus-whatever") on it. It has a "reference fluxivity of 'X' nanowebers-per-meter." It's up to you (and the playback level control) to decide what level that will represent. Typically, it is "zero" on the tape machine's meter, and "plus-whatever" at the output plug (+4, +8, etc.—depending on how the machine was wired at the factory).

Of course, the output level of the recorder has no effect whatever on the fluxivity of the recording made on the machine. However, the reference fluxivity of the last test tape used does have an effect.

When a machine has been set up with test tape 'A', the output level, when playing back test tape 'B', will be higher (or lower), according to the formula;

$$N\text{dB} = 20 \log A/B$$

A and B are the reference fluxivities of the two test tapes.

Of course, a "zero" on the meter will always produce the same output level, but if that zero now represents a recorded fluxivity of 320 nWb/M rather than 185, it means that a higher input level (in this case, 4.76 dB) was applied to the tape.

Therefore, if a program is recorded simultaneously on identical machines that have been aligned with different test tapes, an A/B comparison during the recording session will show no difference in playback level. However, if either tape is played back later on the other machine (A-on-B, or B-on-A), a level difference of N dB will show up. If both tapes are interchanged, there will be a level difference of 2N dB between machines.

Although playback levels can be corrected by rebalancing, using the test tone at the head of the recording (you *did* remember to record one, didn't you?), the tapes cannot be spliced together, unless you don't mind sudden level jumps at the edit points. ■

MICHAEL RETTINGER

Restricting Sound Radiation: Theory vs. Practice

Want to keep the sound level up without having the neighborhood up in arms? If long horns turn out to be impractical, perhaps a sound shield holds the answer.

IN A RESIDENTIAL AREA near an outdoor entertainment facility, the difference between acceptable and non-acceptable sound levels produced by the sound reinforcement system may only be a matter of some 10 dB. Yet it is unwarranted to ask performers to lower their music level, or to reduce the gain of the house amplfiers by that much. It may also be impossible to introduce a sound shield or noise barrier of sufficient height between the amphitheatre and the residential area to achieve satisfactory reduction of sound levels. (For more about this type of barrier, see the author's "Sound Reinforcement Systems in Amphitheatres" in the May, 1980 **db**-Ed.)

What then can be done to pacify the residents, short of housing in the entertainment area? One possibility—theoretically, at least—is to design a horn system that will limit coverage to the theatre audience, with sufficient off-axis attenuation to satisfy the neighbors. At mid- and high-frequencies, this may be no problem, but specially-designed long horns will probably be required to satisfactorily handle low frequency response.

Long horns? It is well-known that the directional characteristics of a horn become narrower as the horn becomes longer, assuming its throat area to remain constant, as well as its flaring constant, or taper; that is, the factor by which the cross-sectional area of the horn becomes larger with distance from the loudspeaker at the throat.

HORN DESIGN THEORY

For an exponential horn, the fundamental design equation is:

$$\frac{D_X}{D_O} = \epsilon^{.5mx}$$

D_X = horn mouth diameter
D_O = horn throat diameter
m = taper of the horn
x = horn length
ϵ = 2.718 (epsilon)

Michael Rettinger is an acoustics consultant and author of "Acoustic Design and Noise Control."

TAPER

The taper of an exponential horn is proportional to the horn's cut-off frequency, and is given by the formula:

$$m = \frac{4\pi f_c}{C}$$

f_C = cut-off frequency
C = velocity of sound, in feet-per-second

For best results, the cut-off frequency should be an octave lower than the lowest frequency which the horn is to reproduce. Thus when 100 Hz is the lowest frequency, = m0.555.

HORN THROAT DIAMETER

This may be as large as 1.5 feet, when the horn is to be used in conjunction with a large-diameter woofer. When a single woofer is used, the throat diameter is made a little smaller, to achieve better loading for the diaphragm. However, when two woofers are installed on a single horn, the diameter of the throat may be even larger than 1.5 feet.

HORN MOUTH DIAMETER

The horn mouth diameter is equal to Nλ, where λ is the wavelength of the lowest frequency to be reproduced.

N is the ratio of mouth diameter-to-wavelength. It may be found from the graph in FIGURE 1, once the required coverage angle has been specified. The radiation angle—which is half the coverage angle—is that angle at which the sound pressure level is down 10 dB. (Some designers specify the radiation angle as the 6 dB-down point—**Ed.**)

To achieve a 10 dB reduction in sound level at any angle, the sound pressure at that angle should be .316, as compared to unity at zero degrees (SPL = 20 log .316/1 = −10 dB—**Ed.**). Referring again to FIGURE 1, the value of N for radiation angles up to 60 degrees may be found. From the graph, note that the larger the horn mouth diameter (and therefore, N), the *smaller* the radiation angle (for low frequencies). FIGURE 2 shows that this is due to the greater path length difference, L, for radiations from a large piston (i.e., mouth diameter) than from a small one, at the same radiation angle. The greater the path-length difference, the lower the frequency at which destructive sound

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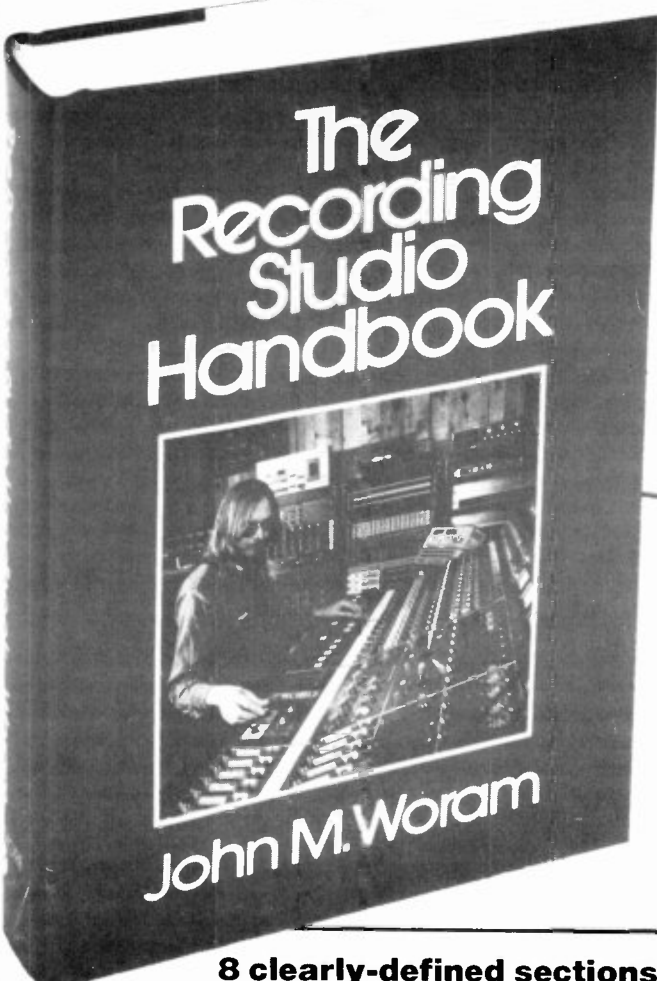

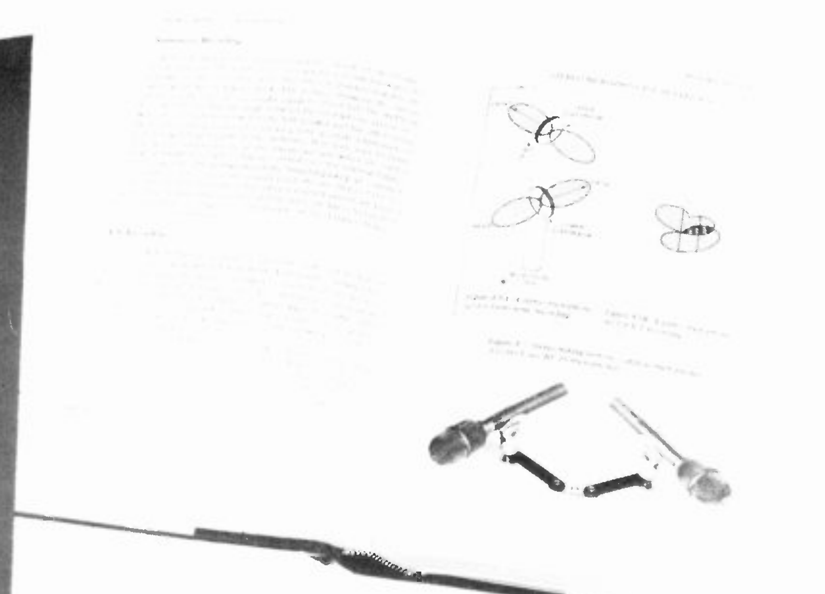

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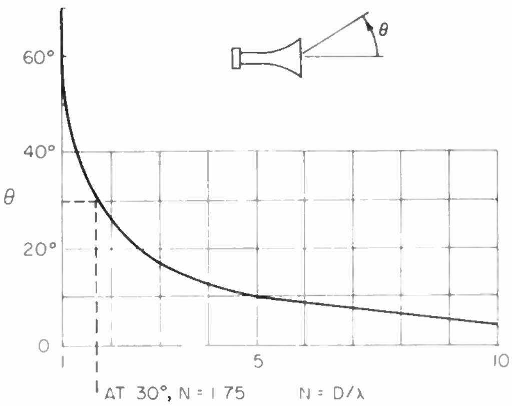


Figure 1. The curve above finds the value of N that will give a 10 dB reduction at any radiation angle, from 0 to 60 degrees.
Example: At a frequency of 100 Hz, what is the mouth diameter required for a 10 dB reduction at 30 degrees?
From the graph, at 30 degrees, N = 1.75
D = N = 1.75 (11.28) = 19.75 feet
Note: at 30 degrees, a practical low frequency horn may be down 10 dB at about 500-to-1,000 Hz. At 100 Hz, the -10 dB pont may not occur until about 100 degrees.

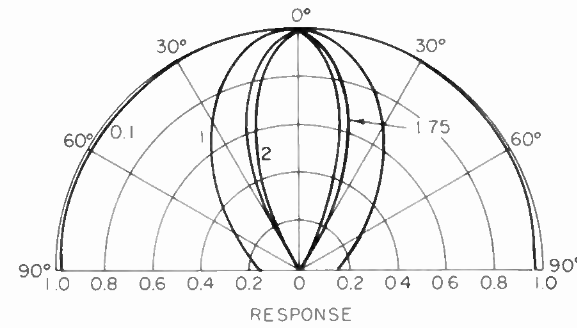


Figure 3. Directional characteristics on an exponential horn for various ratios of horn mouth diameter to the wavelength of the sound radiated by the horn.

interference occurs. The calculations of the effect were first made by Lord Rayleigh in 1878. (See "The Theory of Sound" Dover Publications, Volume 2, para. 302.)

FIGURE 3 plots the directional characteristics of an exponential horn for various values of N, to re-emphasize that, for a constant frequency, and therefore for a constant wavelength, the larger the diameter of the horn mouth, the narrower is the directional response.

CALCULATING HORN LENGTH

If we re-write the fundamental equation for an exponential horn, given above, we may determine the horn length, x.

$$x = \frac{\log N\lambda - \log D_0}{.217m}$$

N = 1.75 (from FIGURE 1)
λ = 11.28 (wavelength of 100 Hz)
D_0 = 1.5 feet (for a large-diameter woofer)
m = 0.555 (calculated above)

From our calculations, it appears that, in order to achieve a 10 dB low-frequency reduction at 30 degrees off-axis, we shall need a horn that is over 9-feet long, with a mouth diameter of almost 20 feet! Obviously, such a horn is not readily available on the open market, and other methods must be used instead.

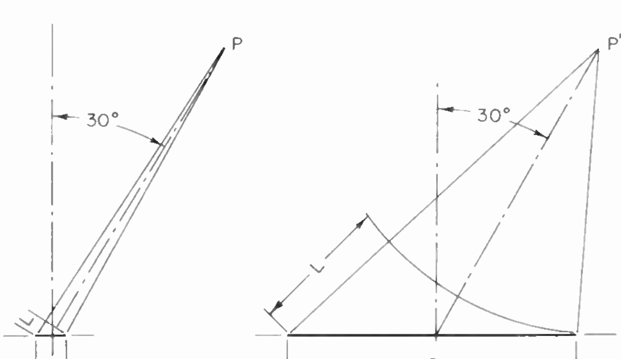
For example, in order to use readily available loudspeaker/horn combinations for sound reinforcement systems, without directing an objectionable amount of sound into nearby residential areas, large sound-absorbent baffles may be placed about the radiators. When the noise-sensitive area is in the longitudinal direction of the loudspeakers, the baffle is placed horizontally over the radiators. When the neighborhood is to the side of the radiator, the baffle may be placed vertically near the speaker, as shown in FIGURE 4.

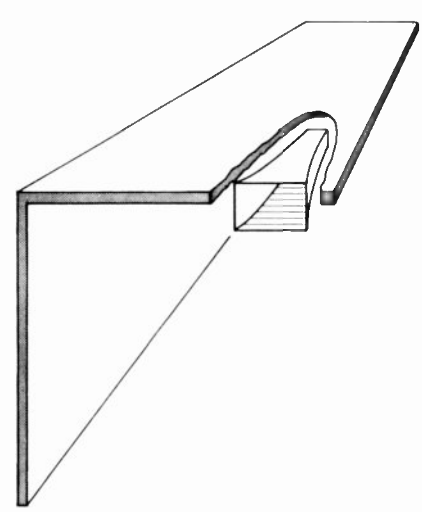
Sometimes, an attempt is made to cover the front part of the seating area with the sound radiated from the stage speakers, and the rear portion with speakers at the back wall, radiating towards the stage. However, this kind of system must be very carefully set up, because it may become disturbing to the performers on stage, if they can hear the delayed sound from the rear speakers.

It is not advisable to set up an array of speakers in the rear, whose reproduction—directed towards the stage—is 180 degrees out-of-phase with the stage speakers, to obtain a "dead wall" or segments of silence. Such ideas work well on paper, but are ineffectual in practice. ■

Figure 4. Horizontal and vertical baffles about a practical loudspeaker/horn system to reduce the radiation of objectionable sound levels into adjoining noise-sensitive areas when the entertainment facility is in the open.

Figure 2. For the same radiation angle (30 degrees), the greater the path length difference, the lower the frequency at which destructive sound interference results.





Application Notes

Audio Pad Design Update

IN THE MAY DB, our Application Note on Audio Pads listed a computer program useful for designing T and H pads. However, every now and then, the program will indicate that a negative resistance value is required. (For those who wrote in explaining why this may happen.

This is encouraging for those who fear that computers will eventually take over the world, for it illustrates again that the poor things are really quite stupid, and certainly not smart enough to do anything dreadful all by themselves. (Of course, they're just marvelous for wreaking havoc with a little human help, but that's another story.)

Anyway, our computer has not been programmed (yet) to reject the impossible request, hence the occasional "minus" value, when the human operator forgets about "insertion losses." The Audio Cyclopedia defines insertion loss as, "the loss created by the insertion of a device in an electrical circuit." In other words, a pad may be designed for any value of attenuation greater than its own insertion loss. It may *not* be designed for losses that are less than the insertion loss.

The insertion loss for a pad depends upon its input (Z_1) and output (Z_2) impedances, and is found from the formula:

$$20 \log (S + \sqrt{S^2 - 1} \qquad S = \sqrt{\frac{Z_1}{Z_2}}$$

As pointed out by reader W. C. Stuchell, a simple addition to our program will calculate the minimum insertion loss of any pad. Simply add the following lines to the program;

```
32  L = 20*LOG ((Z1/Z2)^.5 + (Z1/Z2 - 1)^.5)/2.302
34  PRINT "MINIMUM LOSS IS ";L;" DB."
```

Since most personal computers use natural logs (base 2.718) rather than the more-familiar common log (base 10), the value of 2.302 in line 32 performs the necessary conversion.

For a pad with a 600-ohm input impedance, and a 500-ohm output impedance, you should now discover that the minimum insertion loss is 3.76 dB. Therefore, if you design such a pad for a loss of less than 3.76 dB, be prepared to encounter a negative resistance.

As a "double protection," the program can be designed to ignore impossible requests, by adding another line:

```
75  IF A(N) < L THEN PRINT "AN ATTENUATION OF
    ";A(N);" DB IS NOT POSSIBLE. TO CONTINUE,
    PLEASE ENTER A VALUE GREATER THAN ";L;"
    DB.": GOTO 60
```

This line simply discards values that would require a negative resistance. The program returns to line 60 to await further entries.

DESIGNING BRIDGED-T PADS

Mr. Stuchell also sends along a program for calculating the resistance values for a bridged-T pad, which may be used only when $Z_1 = Z_2$. The following lines, added to our May program, will produce the desired values of resistance.

```
292  VTAB 15
293  IF Z1 < > Z2 THEN PRINT "—BRIDGED-T PAD IS
     NOT POSSIBLE—":GOTO 300
294  PRINT "B-T PADS";
295  PRINT TAB (10) "R1,R2 = ";
296  PRINT TAB (20) "R3 = ";
297  PRINT TAB (30) "R4 = "

490  IF Z1 = Z2 THEN GOSUB 800

680  VTAB 22 (or whatever number will put the cursor at the
     bottom of the CRT display)

800  R3 = Z1/(K - 1)
810  R4 = Z1*(K -1)
820  VTAB (N + 16)
830  PRINT A(N); " DB";
840  PRINT TAB (10)Z1;
850  PRINT TAB (20) INT (R3 * 100 + .5)/100;
860  PRINT TAB (30) INT (R4 * 100 + .5)/100
870  RETURN
```

Lines 250 and 440 are changes to the original program, to make room on the CRT display for the additional information.

Lines 292-to-297 are simply the column headings for the new data.

Line 490 sends the program to line 800, where a sub-routine does the necessary calculations, but *only* if the input and output impedances are equal. Otherwise, the lines 800 and above are ignored.

Line 680 simply puts the cursor at the bottom of the screen when the program is over. Otherwise, it may wind up in the middle of the bridged-T data display.

Lines 800-to-870 calculate the values for R_3 and R_4, and then display them on the screen. R_4 is a new resistor, wired in parallel with the series combination R_1 and R_2. R_1 and R_2 are equal to the input (or output) impedance, and therefore don't have to be calculated. ■

Note: *For those who took us up on the invitation at the bottom of the May Application Note, your "compsub" is on the way.*

The AES Audio Workshop Program

The 67th convention of the Audio Engineering Society will be held in New York City later this month (31 October-3 November) at the Waldorf-Astoria Hotel. Of special interest to **db** readers is the much-expanded workshop program being offered this year. With nine sessions, this will be the Society's biggest-ever workshop. As before, the emphasis is on the practical—rather than theoretical—application of new technology.

For the benefit of **db** readers who are planning to attend the convention, here's a brief outline of the workshop schedule.

Friday, October 31 9:00 AM

DIGITAL EDITING

Chairman: Peter Jensen, Digital Recording Systems Co., New York, N.Y., and Elkins Park, Pennsylvania

Tapes recorded in the various digital formats must be edited to make final master tapes. A panel of experienced users will compare the editing capabilities available with each current system, with special emphasis on convenience, accuracy, and cost. Several digital systems will be demonstrated.

Friday, October 31 2:00 PM

SOUND REINFORCEMENT

Chairman: David M. Andrews, Andrews Audio Consultants, New York, New York

A panel of sound-installation professionals and equipment manufacturers will discuss common problems encountered in the development and installation of contemporary sound reinforcemeny systems. Areas for specific discussion will include architectural acoustical materials and room treatments, equalization and testing, and the use of delay lines.

Friday, October 31 7:00 PM

HIGH-SPEED DUPLICATION

Chairman: Tim Cole, MTI Corporation, Montclair, New Jersey

A panel of manufacturers and users will discuss the state of the art of the duplication process. Duplicating and other production equipment for varying levels of capacity as well as magnetic heads and tape will be covered in preliminary remarks. An open forum will follow to discuss common problems faced by the duplicator in the areas of standards, quality control, and raw materials. It is hoped that this workshop will begin a continuing dialogue among tape duplicators at future conventions.

Saturday, November 1 9:00 AM

POTENTIALS OF PERSONALIZED, PRIVATE RECORDING STUDIOS

Moderator: Larry Blakely, Cameo, Framingham, Massachusetts

A panel of experts will discuss the private recording studio and the role it will play in the future of the record and music business. With the major changes taking place in today's record industry, the private recording studio will likely hold a vital position in the exploration and promotion of new recording artists. The panelists will discuss many facets of the private recording studio from the viewpoint of the record industry, producer, musician, and recording engineer.

Saturday, November 1 2:00 PM

EDUCATIONAL FAIR

Chairman: Almon Clegg, Panasonic, Secaucus, New Jersey

A gathering of representatives of universities and educational institutions offering courses in audio will be on hand for personal discussions with prospective students. Information will be provided on entrance requirements and curriculum, with details of special courses available of interest to those entering or involved in the audio field.

Saturday, November 1 7:00 PM

VIDEO FOR AUDIO

Chairmen: Jack Zupko and Tom Bentz, Panasonic, Secaucus, New Jersey

The merging of the audio and video arts makes it imperative that the engineer who is expert in one of these fields be at least cognizant of the fundamentals of the other. Designed for the audio engineer who must deal with the interfacing of the two technologies, this workshop will provide an introduction to video equipment, a basic discussion of the NTSC system, and an explanation of the specifications and terms used in the video world. A question and answer period will follow the presentations.

Sunday, November 2 2:00 PM

MICROPHONE TECHNIQUES FOR RECORDING AND BROADCASTING

Chairman: Robert B. Schulein, Shure Brothers, Inc., Evanston, Illinois

The knowledge of microphone techniques and how they are used to create desired audio illusions is absolutely necessary for audio professionals. With the aid of examples and demonstrations, the presentations in this workshop will lay the theoretical foundation for these techniques, and will discuss in practical terms their application to recording and broadcasting situations.

Monday, November 3 9:00 AM

AUDIO IN MEDICINE

Chairmen: Martin Polon, Audio and Video Consultant, UCLA, Los Angeles, California, and Associate Editor, Video Magazine, New York, New York and Philip Kantrowitz, Bronx, New York

A panel of experts will discuss the effects of sound on the human body, with particular emphasis on its relationship to heart diseases and body damage.

Monday, November 3 9:00 AM and 2:00 PM

MULTITRACK TAPE RECORDER MAINTENANCE

Chairman: John R. French, JRF Company, Hopatcong, New Jersey

Most manufacturers of multitrack tape recorders will be represented at this double session. Each manufacturer will provide one or more machines along with field-engineering personnel to discuss the recommended procedures for alignment, troubleshooting, and preventive maintenance. There will be separate areas for each manufacturer so that several can be visited in one session.

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db People/Places/Happenings

• **Bruce Martin** and **Norman Kassel** of **Martin Audio Video Corp.,** announced the appointment of **Courtney Spencer** to the position of Vice President. Mr. Spencer, currently the General Manager at Martin, will retain that position in addition to assuming his new executive capacity.

• In a series of moves, **Allen and Heath Brenell Ltd.,** announced the appointment of three companies to act as official agents for AHB Ltd.'s range of professional audio products. **ACI Filmways,** of Hollywood, will be the new west coast agents, **Otari Electric Company, Ltd.,** of Tokyo, the Japanese agents, and **White Electronics,** of Ontario, the Canadian agents.

• **Peter Giddings** has been appointed International Sales Manager of **Clear-Com Intercom Systems.** Giddings is co-founder of **Beyer Dynamics** of England, former VP of marketing for **Revox Corp.,** and former Director of Marketing for **Hammond Industries** of England.

• **Robert W. Carr** has joined **Sescom, Inc.,** in Las Vegas, Nevada, as Vice President of Marketing. He had previously been with **Shure Bros., Inc.**

• **James W. Beattie,** General Sales Manager of **Crown International Inc.,** recently revealed the sale of 140 model D150AIOC audio amplifiers to the **U.S. Army and Air Force Exchange Motion Picture Service.**

• **James K. Dobey,** who retired last year from his post as the chairman of the board of **Wells Fargo & Company,** has been elected to the board of directors of **Ampex Corporation.** Mr. Dobey is currently a director of **National Gypsum Company** and Wells Fargo & Company, and a trustee of the **Wells Fargo Mortgage and Equity Trust.**

• **Brittania Row Studios,** co-owned by the members of the British rock group, **Pink Floyd,** have been granted membership in the **Association of Professional Recording Studios.** They join **Berwick Studios** and **Branston Studio** as the newest members of APRS.

• **Robert W. Carver,** President of **Carver Corporation,** Woodinville, Wash., has caused a Complaint to be filed in the U.S. District Court for the District of Massachusetts against **Joel M. Cohen** and **Sound Concepts, Inc.,** Brookline, Mass. The Complaint, identified as Civil Action No. 80-1935-K, alleges infringement by Mr. Cohen and Sound Concepts, Inc., of U.S. Letters Patent No. 4,218,585 issued to Mr. Carver on August 19, 1980 and entitled, "Dimensional Sound Producing Apparatus and Method" by virtue of the manufacture, use and/or sale of the Sound Concepts IR2100 Image Restoration Control. The Complaint seeks damages and injunctive relief.

• **Robert W. Ponto,** known throughout the consumer electronics industry through his long term association with **Shure Brothers, Inc.,** announced the formation of **Roger Ponto Associates,** Manufacturers' Representative in the Pacific Northwest. For the past three years, Ponto has been Vice President and partner in the representative firm of **Fleehart & Sullivan, Inc.**

• **James B. Lansing Sound, Inc.** continued its support of the **Aspen Music Festival's Audio Institute,** a "hands-on" approach to teaching live recording techniques, which completed its third operating season August 24. JBL Vice President for Market Planning, **John Eargle,** past president of the **Audio Engineering Society** and author of "Sound Recording," visited the school to lecture at each of its three consecutively-run sessions held throughout July and August. In addition, the company donated several hundred Aspen Music Festival/JBL T-shirts to the school for resale among students and concertgoers.

• **Karen White** has been named Promotional Manager of the newly-renovated **Concorde Recording Center** in Los Angeles, according to **Warren Entner,** managing director of the studio complex. Additionally, White will serve as general manager of **Mariner Productions,** Entner's in-house production company. White comes to Concorde after eight years as production assistant to record producer **Steve Barri,** four years of which were spent at **ABC Records** and the other four at **Warner Brothers Records.** White's duties at Concorde will include general promotion of the studio and production coordination.

• **Professional Sounds, Inc.** has expanded their design, installation, and maintenance operations to include the Nashville area. Their central office will remain in Falls Church, Virginia.

• **Shure Brothers Inc.,** Evanston, Ill., has announced that **William P. Finnegan** has joined the company as Vice President of Marketing. In this new position, Finnegan will have responsibility for the marketing of all Shure products sold domestically and in foreign markets. In addition, he will be responsible for the company's OEM sales. Finnegan comes to Shure from **Quasar Company** where he held the position of Director of Marketing.

• **Sound Ideas Studios** has become the first New York studio to receive a **3M Digital Mastering System,** consisting of four-track and 32-track recorders. "We're taking a significant step in offering the quality of recording which is possible with the digital multi-track system," says **George Klabin,** owner of Sound Ideas. To provide greater versatility to the artist, Sound Ideas' digital system can be used in either newly rebuilt Studio A or in Studio C, which can handle up to 40 musicians. One of the first groups to test the multi-track system at Sound Ideas was the **BT Express,** a rhythm-and-blues instrumental group recording for **Columbia Records** (producer), **Morris Brown).** Rhythm-and-blues, jazz and commercial jingles are Sound Ideas' primary productions. A digital commercial will be one of the first recordings with the multi-track system. Sound Ideas also offers 3M's electronic digital editing system and a digital preview unit.

• **Donald J. Linehan** has been named Manager, Marketing Communications and Merchandising for **3M's Magnetic Audio/Video Products Division, Industrial markets,** announced **Jack B. Hanks,** marketing operations manager. Linehan, who joined 3M in 1968, had been marketing communications supervisor of 3M's Micrographics Products Division prior to this appointment. In his new position, he will be responsible for the development of sales promotion and merchandising materials as well as advertising programs for the business, educational, broadcasting, and professional recording markets.

The UREI Equalizers

In the hands of a professional, an equalizer can work wonders. For the person who owns a UREI equalizer, miracles are often made.

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Whether to correct or create, UREI offers a complete line of the most effective audio frequency shaping instruments available:

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Two channels of ten calibrated, step less, vertical adjustment controls with ±12dB boost or attenuation. Also available in a single channel version-Model 533.

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A single channel device, it provides ±12dB of boost or cut in 27 ISO 1/3 octave increments from 40Hz to 16kHz.

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Specifically designed for room equalization, it offers 27 ISO 1/3 octave calibrated adjustments from 0 to −15 dB attenuation plus band-end tunable high and low-pass filters.

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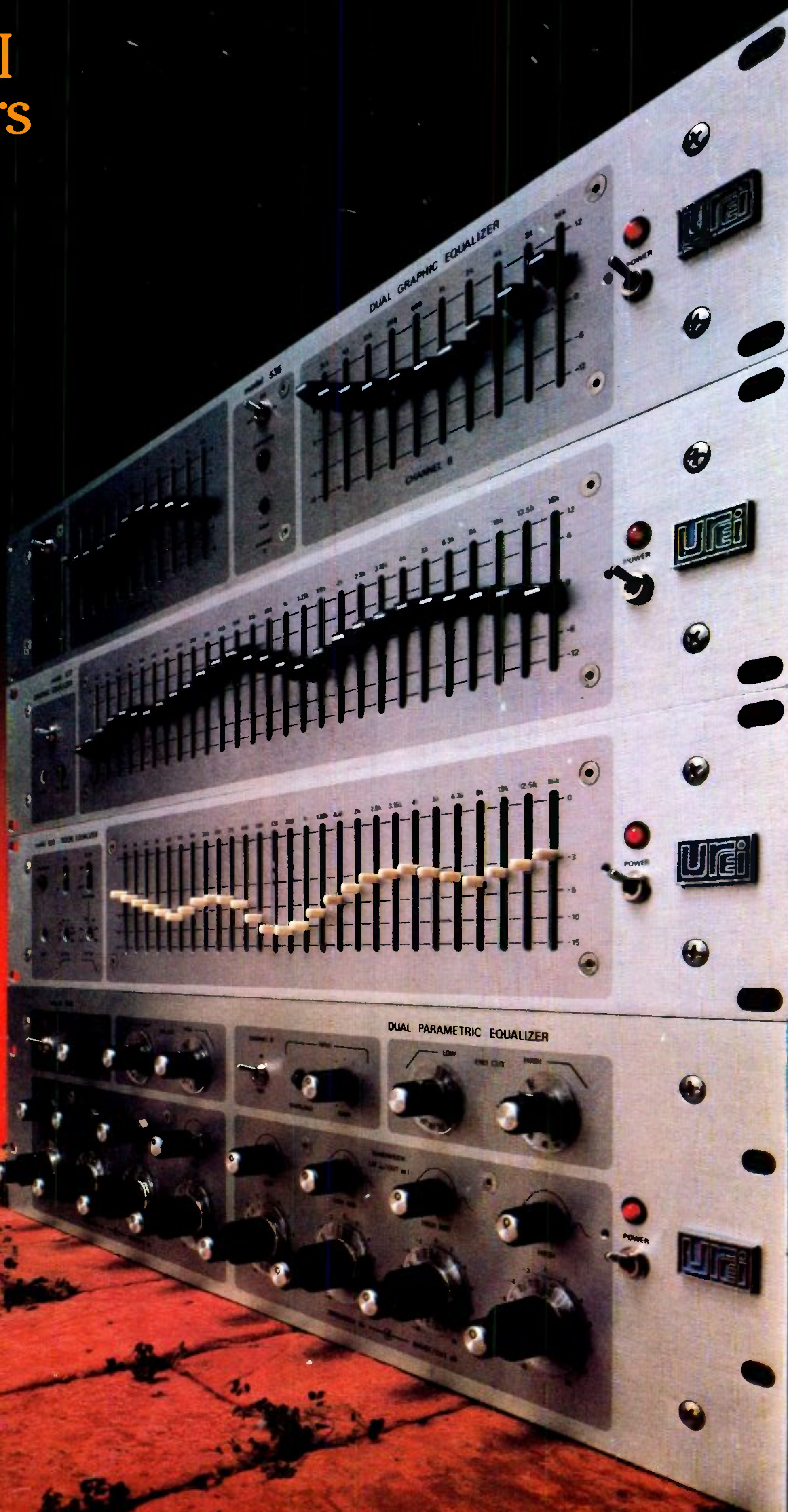

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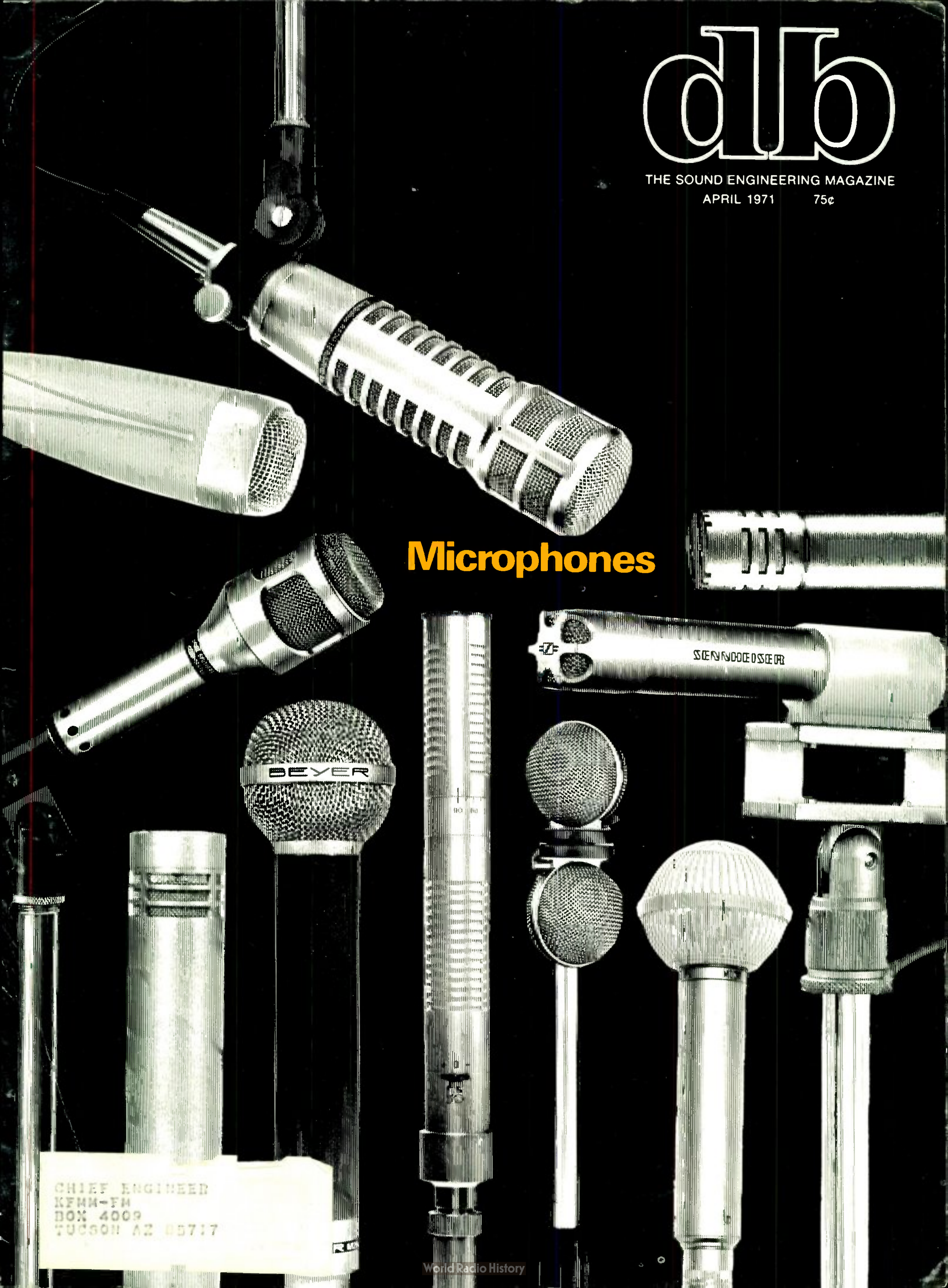
THE SOUND ENGINEERING MAGAZINE
APRIL 1971 75¢

Microphones



COMING NEXT MONTH

• Noted acoustic consultant Robert Hansen has written a down-to-earth article on the practical steps one must take in planning and actually building the studio. Every studio designer or would-be one will want to refer to it throughout his job.

Sound Insulation Requirements for Rock Studios is the title of Michael Rettinger's latest contribution. He details how it is possible to keep rock in and rumble out of the recording studio.

Part 2 of *Acoustics for Audio Men* by Melvin Sprinkle continues the basics begun in the March issue. This three-parter will prove to be a veritable textbook on the subject.

And there will be our regular columnists: George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, Arnold Schwartz, and John Woram. Coming in **db**, The Sound Engineering Magazine.

ABOUT THE COVER

• Our cover this month requires little in the way of explanation. It is composed of many familiar (and some unfamiliar) brands of microphone. Try your hand at identifying them. There are no prizes if you do.

advertisers index

db

THE SOUND ENGINEERING MAGAZINE

APRIL 1971 VOLUME 5, NUMBER 4

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

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George Alexandrovich

THE AUDIO ENGINEER'S HANDBOOK

FLAT FREQUENCY RESPONSE—WHEN, WHY and WHY NOT

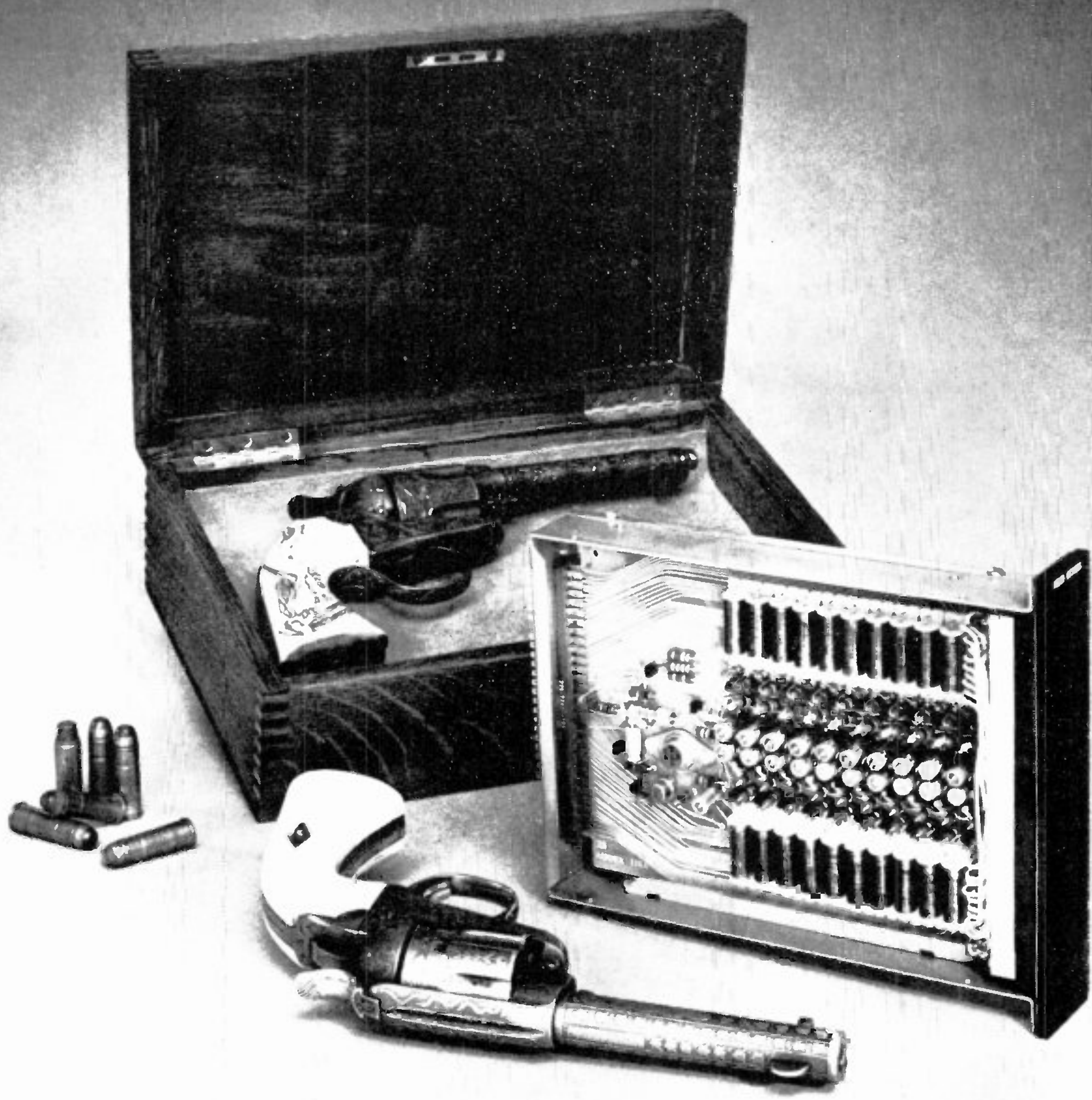
• Discussions about frequency response are as old as audio technology itself. Not so long ago scores of audio engineers and circuit designers were burning midnight oil trying to achieve better amplifier circuits, and correct frequency response of transducers and storage media. They were fighting a long battle against rolling off response at both ends of the audio spectrum with dips and peaks between. The advent of solid-state technology, supplemented by the efforts sponsored by computer and space system manufacturers, considerably changed the picture. We can purchase an amplifier today with frequency response flat from d.c. to the megahertz range. Yet, a lot remains to be done. This is not so much how flat we can make response, but how to control it in order to achieve clean, pleasant, true-to-life reproduction of a recording or performance.

Flat frequency response of the audio chain, or a part of it, is important in certain instances. These may be remixing and mixdown operations, re-recording, broadcasting, or audio distribution functions. Audio signals passing through many stages of amplification and control, have to retain their balance in order for the output signals to resemble the original information. The term *flat response* actually refers more to over-all response, than to the performance of each part of the system. You may consider an amplifier which shows 0.5 dB rolloff at 15 kHz flat; yet, if you have ten such amplifiers in a chain, the final result may be 5 dB rolloff at 15 kHz. On the other hand, if you have several amplifiers, half of which have a dip at a certain frequency, and the other half have a peak equal in amplitude but opposite in sign, then such an amplifier chain can be considered being flat. It does not mean that this is how we should construct our systems, but serves as an illustration to the following discussion.

In my audio experience over a period of more than twenty years, I have been acquainted with many recording-industry purists who believe that every step in sound recording should be identified with absolutely flat response—be it transducer, amplifier, or studio. They do not recognize any form of sound compression, limiting, equalization, or any other forms of tampering with signals. They also believe in one microphone session, and most of the time they would not record anything else but classical music. At times, the results of their efforts may be satisfactory if they happen to get a good-sounding studio and the right equipment. Most of the time their recordings are noisy and flat sounding, and not acceptable by today's standards. The trouble is that the sound you hear, even in the best hall, does not sell unless it is processed and enhanced just in the way color photography emphasizes certain colors. Let us follow the path of the audio signal from the musician to the speaker in your room.

First of all, today's micing techniques do something you can not do with one microphone, nor with a live orchestra. By placing a micro-

Switching? Don't fight it!



Buss assignment and matrixing multichannel consoles don't have to involve a hassle with 640 buttons. Compact Audex switchers simplify console programming by reducing controls to a maximum of 22 buttons and a dial, grouped for one-hand operation.

And the modular Audex switchers provide computer-like lighted digital display, for instant read-out. Gives the professional soundman incredible speed, incredible versatility, plus easy add-on capability as needs increase.

Audio Designs offers three Audex versions to cover the exacting requirements of the most sophisticated sound studio. Audex I separately switches a program source to 16 program busses while preventing selection of more than one buss at a time. Audex II, when used as a buss assigner, will cumulatively switch a source to any one or all 16 program busses and 4 echo busses.

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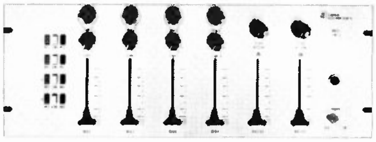
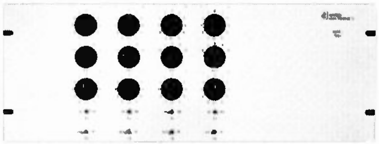




phone in front of every instrument, the ears of the listeners are "extended" close to each instrument. The output of each mic may be recorded on a separate tape track. In order to enhance or emphasize the sound of specific instruments, recording engineers may use an equalizer on particular channels to bring out the range of frequencies wanted to be heard from this instrument, and subdue the sounds picked up from adjacent instruments. The result is that the frequency response in this mic channel is twisted all out of shape. If this is the drum mic, it is possible to make the drum beat heard clearly while the squeaking of the foot pedal or the hissing of the air conditioning is eliminated. Before we know it, a signal is being fed into the tape recorder. The first thing that happens is that we attenuate low frequencies, boost highs, and record it that way. Once the session is over, a multi-track recording has to be remixed into two or a single channel. In playing back individual channels post-emphasis is applied—we equalize the signal by boosting the low frequencies and attenuating the highs. Then we mix the signal with other signals and re-record the mix on the new track, again applying pre-equalization. If this recording is meant for tape duplication the process is repeated again in duplication, then again in the home playback setup. One can not help wondering how after all those re-recording steps, recorded tapes and discs sound so good.

In this long list of manipulation with signals, one can count on a possible additional equalization of each channel, inclusion of artificial reverberation, compression, and limiting.

Finally, this recording is being played back in the home or in some other room. In order to compensate for the deficiencies of the speaker system and room acoustics, we introduce environmental equalization. This type of equalization can be quite severe—at times, filters boosting or filtering certain frequencies may have peaks exceeding 10 dB. After this is all done, you can hardly distinguish the obtained sound from the original copy.

What it all amounts to, is that when we refer to frequency response, we have in mind the precision with which the sound is processed. *Flat* refers to the relationship of input with respect to output. You may convert audio signals to digital form, store it in a computer memory, then read it out, and reconvert it back to analog information—and still have flat response. It is not *how* you process the sound; it is how it *compares* to the original.

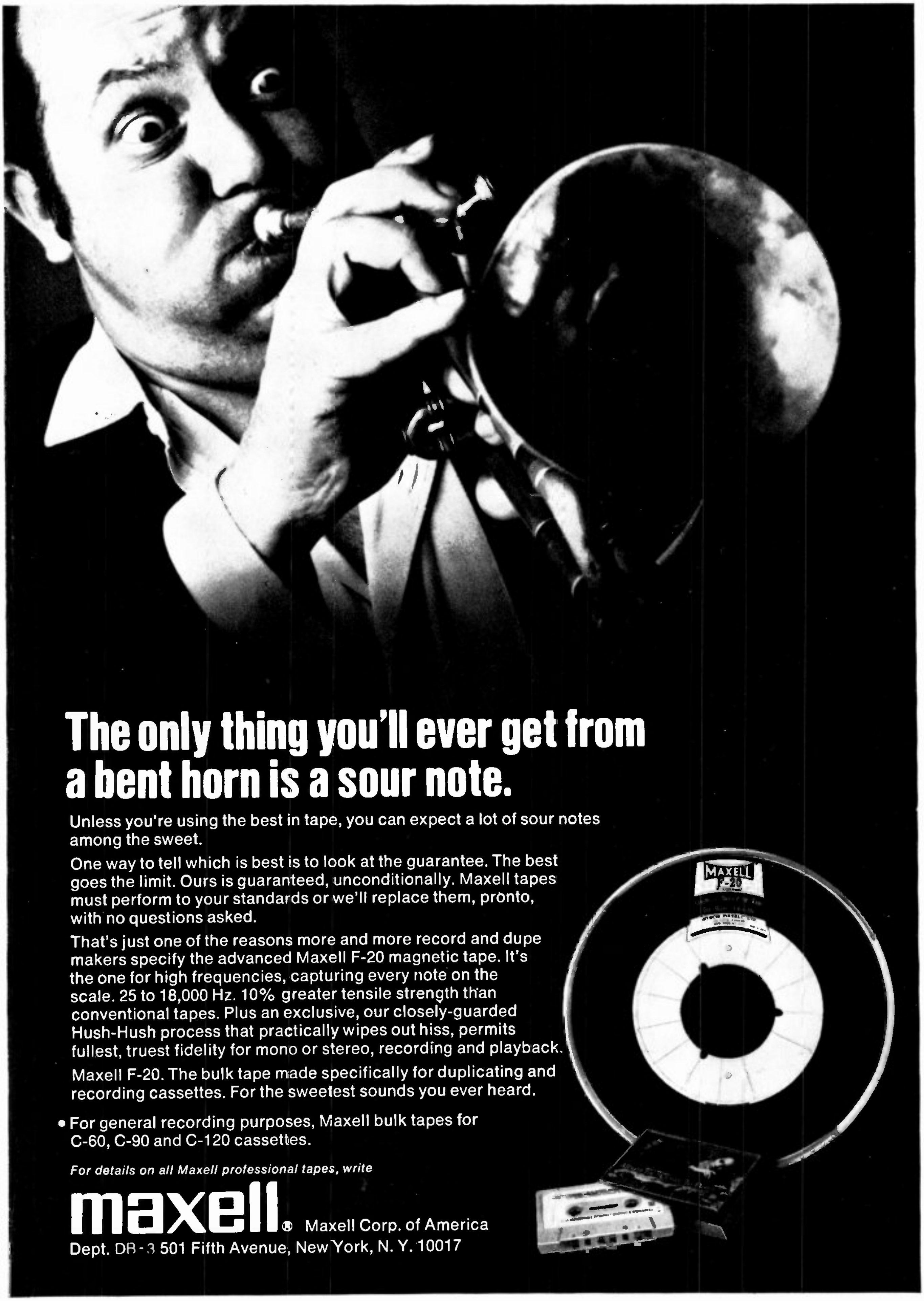
Until now, we have been concerned with faithful reproduction of sound and general frequency response. But how far out in frequency do we have to go on both sides of the audio spectrum in order to achieve true fidelity? Can we hear as high as 20 kHz or as low as 20 Hz.

Lately, we all have heard much talk about different kinds of pollution. Sound is one of them. We hear speculation that loud sounds can effect our health, our minds, cause heart attacks, and so on. Well, it seems that there is little we can do in controlling the level of the hi-fi systems in the homes of the consumers.

Our technology has gone quite far unchecked. It is no secret that direct-coupled amplifiers, in combination with certain speakers, can produce high intensity sound waves in both subsonic and supersonic regions. Let us stop and think about what we have there. Think of the walls of Jericho, or new methods of detaching the retina of the eye with ultrasound. Maybe you haven't seen how ceramic magnets are cut apart with dull chisels driven at ultrasonic frequencies. Or have you ever seen the test of any mechanical structure on a vibration platform, and what happens when some parts begin to resonate? Some twenty-five years ago, I saw an experiment performed where tobacco smoke was blown into the path of ultrasonic beam—it condensed and disappeared.

Many uses of sound are not yet known to us, nor are the after effects on human organs. We know that some plants and flowers, when exposed to sounds of different frequencies, change their growth rates. Until we uncover the secrets of the world of sound, can we try to limit a generation of frequencies which seem only to be important as a sales pitch in consumer products. I think that going beyond a 20 Hz-20 kHz range is more than any one can appreciate. It only makes dogs restless.

I would like to see a built-in rolloff at these frequencies in every system. Signal harmonics in the 20 kHz region can already be transmitted over the air like radio waves. If we suspect that some sort of brain waves can influence


The only thing you'll ever get from a bent horn is a sour note.
Unless you're using the best in tape, you can expect a lot of sour notes among the sweet.
One way to tell which is best is to look at the guarantee. The best goes the limit. Ours is guaranteed, unconditionally. Maxell tapes must perform to your standards or we'll replace them, pronto, with no questions asked.
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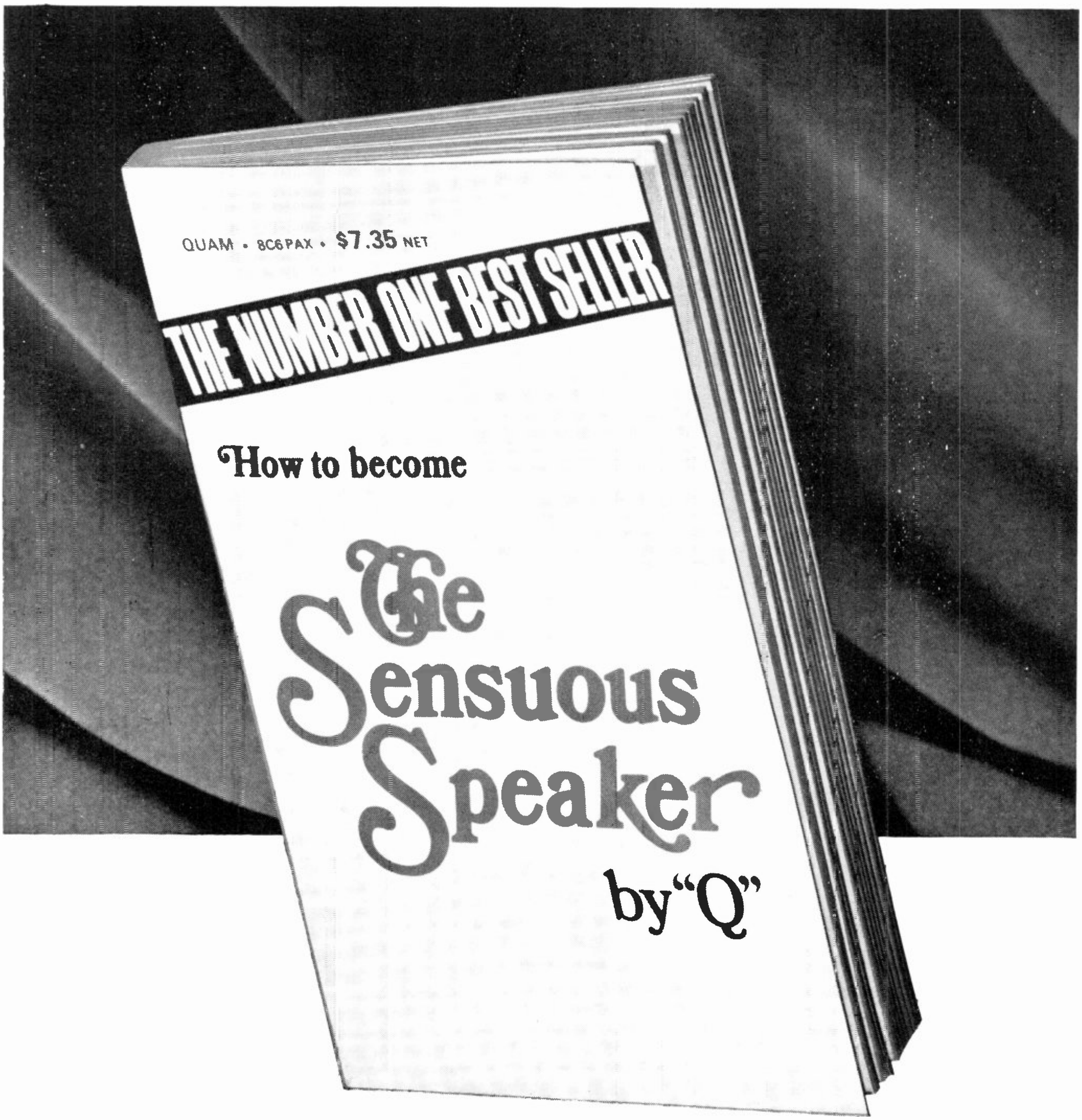
our behavior in extrasensory perception experiments, then let us be very careful with unknown signal sources many times more powerful than energy sources in the human brain. Only recently it was announced that 10¢ diodes under certain conditions can generate frequencies in gHz range.

Many audio engineers prefer using transformers in their systems for isolation purposes, but what they really accomplish is restriction of frequency bandpass characteristics. This, in turn, helps subdue parasitic oscillations, motorboating, and cross-talk. I consider it an easy way out. A more elegant solution to this problem would be no need for d.c. isolation, but where bandwidth of the system is sharply restricted to certain limits. Actually, frequency response and bandwidth are two related terms. Sharp rolloff of the frequency response is the limit of the bandwidth.

Here is another observation in regard to flat response. Our ear, by no means, hears all frequencies with equal efficiency. The Fletcher-Munson curves tells us quite explicitly how imperfect our ear is, and how it reacts at different sound levels. We all know that recordings are processed while being monitored at excruciatingly high levels in mixing rooms. The balance of the sounds for such recordings are valid only for these levels. If you take the same recording and play it back in your living room, at moderate or low levels, your ear becomes insensitive to the extremes of the audio spectrum.

In order to get the balance obtained in the recording studio higher, you should turn your gain control to the ear-splitting levels, boost your low and high frequencies through equalization, or be content with the sound you get. Maybe today's youth are going deaf because they try to recreate the same balance as the recording engineer, heard by raising the reproduced levels well above 100 decibles.

There are instances when flat frequency response up to 100-200 kHz are needed—as during the duplication of recorded tapes at tape speeds that are 8-16 times faster than normal speed, or when cutting discs at half speed (extended low-frequency response of the end product). However, when we generate frequencies that are in turn reproduced by the speakers so that they resonate with our chest cavity and drive us slowly insane, while loosening our tooth fillings, I begin to wonder if we were not better off with good old 78's. ■



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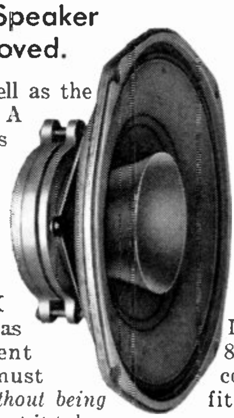

But few manage it as well as the Quam Model 8C6PAX. A jillion of these speakers have already been installed in factories, offices, restaurants, and other locations. What's the secret of success? It's the Sensuous Sound!

The Quam 8C6PAX *knows* that music often has to compete with inherent situational sounds. It must manage to be *audible without being obtrusive*. It has to have what it takes.

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Maybe it sounds like too much trouble for you to become the Sensuous Speaker.

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Circle 16 on Reader Service Card

NORMAN H. CROWHURST

THEORY AND PRACTICE

• Last month we discussed the simulation of hybrid coils without inductors (except for the dummy line, if that needs inductance as an element, or element, but no transformers needed for coupling purposes). This left us with the need for amplifiers using both inputs and outputs up in the air.

The requirement is for an output circuit that delivers signal to a line load, whether or not some signal is fed from the line toward the amplifier. The method we said we would use is that of delivering a current into a 500-ohm load, from a circuit where the voltage can float at either or both terminals.

This is illustrated at *Figure 1*. In quiescent condition, transistors Q2, Q3, Q4, and Q5 all pass precisely 12.5 milliamps. Under that condition, the 12.5 milliamps that passes through Q2 also passes through Q3, and the 12.5 milliamps that passes through Q4 also passes through Q5. This means no current flows in either direction through the 500-ohm output load.

When signal is fed in through Q1, assuming it to be momentarily positive in polarity, of 1 volt magnitude, the 3 milliamp steady current through Q1 (controlled by the fact that its emitter voltage is 3 volts above supply negative, and the emitter resistor is 1 k rises to 4 milliamps.

This increases the voltage from the base to supply positive and negative, at transistors Q2 and Q5 from 3 volts to 4 volts, raising the current from 12.5 milliamps to 16.7 milliamps. As the current in Q4 and Q3 remains at 12.5 milliamps, a current of 4.2 milliamps must pass through the 500-ohm load, yielding an output of 2.1 volts, positive at the collector of Q2, negative at the collector of Q5.

The constant current in Q3, Q4 is maintained by the use of 3-volt zener diodes and the 27 k resistor. As the steady current in Q3, Q4 is 12.5 milliamps, somewhat less than 1 milliamp will assure adequate base current for these transistors, with a sufficient amount to polarize the zener diodes to their working voltage. The voltage between these bases is twice 15-3 = 24 volts, so 27 k will serve.

To achieve a quiescent condition, the bases of Q2, Q5 must also be at precisely 3 volts from supply positive and negative respectively. Using 1 k resistors in the collector and emitter of Q1, this requires 3 milliamps as this transistor's operating current. With a 1 k emitter resistor and an assumed current gain of 60, the d.c. base input resistence will be 60 k. In parallel with the 12 k bottom resistor, this makes 10 k which must have 3 volts developed across it. The upper resistor must then develop 27 volts, requiring 9 times 10 k, for which a chosen 91 k resistor will serve.

To balance the circuit perfectly, the 240-ohm resistors must be close tolerance, and/or the 1 k resistors must be chosen so the quiescent condition exactly balances the currents in Q2 and Q3, and in Q4 and Q5.

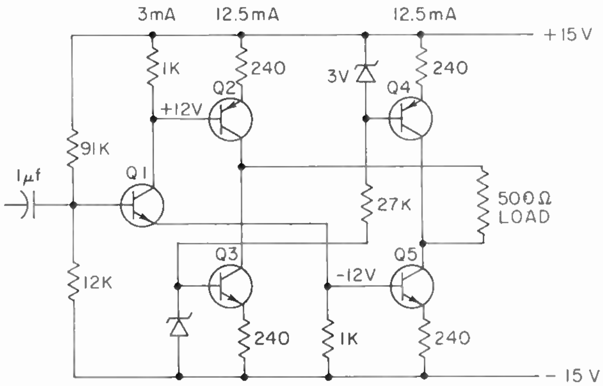


Figure 1. An output circuit that provides a "floating" signal across a 500-ohm load, just the same as a transformer can.

THE OLIVE AUTOMATED REMIX CONSOLE: SUDDENLY "GOOD ENOUGH" ISN'T GOOD ENOUGH ANY MORE.

The Series 2000 Console, our modular multitrack recording and remix centre, incorporates numerous innovations and refinements. Our latest is *automation* — available in a module that just plugs in.

Quite honestly, and without exaggeration, we believe our remix console with the automated programmer will revolutionize the recording industry.

Freed from the physical limitations of trial-and-error mix downs, the producer can control and refine *up to 64 functions* simultaneously or individually! Simply choose what you want to automate. For example, on a 24-track mix down you could have independent operational control of 24 level functions, 6 echo returns, 8 sub masters, 16 panning controls and quadraphonic positioning.

Like a computer, the programmer condenses information onto one track of the master tape, "remembering" each and every console operation on all channels. The producer can make a dry mix to establish levels on the channels then, when remixing, listen to the tape, making chosen modifications *to each track individually.* The Remix Programmer monitors and stores these adjustments for automatic repeat on any future mixes. The final mix can be modified at any time without endangering the master.

Automated Remix Programmer Specifications:

- Channel capacity: 16, 32, 48 or 64 depending on your requirements.
- Accuracy: levels will be accurate to less than 1 dB over the first 40-dB range. Accuracy greater at top of range (working range) and less at lower end, similar to accuracies normally found in a fader.
- Response: information is updated better than 10 times per second. Thus the fastest operator reactions will be preserved.
- Resolution: the system will recognize and reproduce changes smaller than 1/10 dB over the working range (0 to 40 dB).
- Storage medium: may be any audio bandwidth tape recorder channel. The multitrack master tape would be the most convenient source for this channel.

The benefits to a studio operator are enormous. Now one man can do a complex mix of 16 . . . 24 . . . 32 tracks with complete freedom. Approximately 75% of the time and effort presently put into mix-down sessions is eliminated. A new and unmatched creative potential is released. The perfect mix is a dream? No more!

Look for the revolutionary Olive Automated Remix Programmer with the Olive Series 2000 Console at the AES Convention in Los Angeles, April 27th to 30th. If you can't make the show write or phone us. Olive Electro Dynamics Inc., 2670 Paulus, Montreal 386, Quebec, Canada. Tel. (514) 332-0331.

olive

WHENEVER THERE IS A NEED

Circle 20 on Reader Service Card

The input impedance is about 9 k (10 k in parallel with 91 k) so a 1 mFd input capacitor will yield a cut-off at 20 Hz.

This circuit has no voltage gain—to be precise, a little more than 2:1, as calculated above. When maximum current or zero current flows in Q2, Q5, the voltage drop across these transistors is 6 volts or zero. The drop across the 500-ohm load reaches 6 volts either way, which allows a substantial margin to allow the 500-ohm resistor (load) to float away from ground potential, and still allow adequate collector voltage to reach Q2 and Q5, as well as Q3 and Q4.

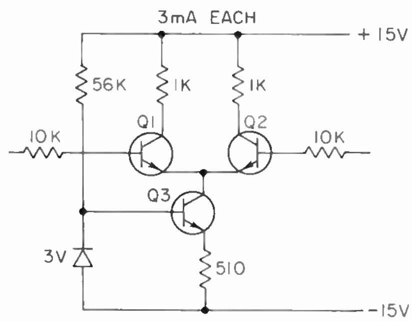
As current in all these transistors is controlled extremely tightly by the base voltage and their emitter resistors, collector voltage has virtually no effect on the signal delivered at the combined collector circuit, which is determined by the fluctuating current in the 500-ohm load.

Now to turn to the floating input, which is a little simpler to achieve, and the circuit is shown at *Figure 2*. The circuit is rendered "floating" by transistor Q3, which is held at constant current by the zener diode and the 510-ohm emitter resistor, which yields a controlled current of 6 milliamps (actually that is based on a resistor value of 500 ohms).

Collector voltage of Q3 can float, just so long as the total collector current adjusts itself to 6 milliamps. This means that, if the voltages applied to the 10 k input resistors connected to Q1 and Q2 bases are identical, their emitters will be just sufficiently negative of that voltage to bias each of them to a collector current of 3 milliamps.

If these transistors have a current gain of 60, their base current in quiescent will be about 50 microamps, requiring 0.5 volt drop in the 10 k resistors. A signal of 1 volt balanced will drive one transistor to 6 milliamps and the other to zero.

Figure 2. An input circuit that can accept a floating input signal, also as well as a transformer can handle it.



With 1 k collector resistors, which provides a push-pull output from this input stage, if desired, each collector voltage is 3 volts below the 15-volt supply positive. Full signal produces 3 volts peak from each, or 6 volts peak-to-peak. So this input stage has a gain of 6. Further gain than the 12 provided by input and output stages can be developed by conventional intermediate stages.

Apart from the 1 volt signal, from one terminal to the other, required to produce a differential output from this input stage, the pair of input terminals can float from almost 10 volts negative of ground to some 9 volts positive of ground, while still giving all the transistors an operable collector voltage.

As this common voltage drives Q1, Q2 emitters and bases positive or negative, the bases maintain the same quiescent current, which means the 0.5 volt *drop* in the 10 k resistors remains unchanged, whatever the *actual* voltage is. When the input terminals get to −12 volts, the collector voltage across Q3 vanishes, while pushing it to +9 volts, this voltage rises to 21 volts.

The voltage at the collectors of Q1, Q2, relative to their emitters and bases, also varies. At quiescent, each collector is 3 volts negative from supply plus, or +12 volts. Maximum output swing is ±3 volts from this, or from +9 volts to +15 volts.

To achieve balance in the output, Q1 and Q2 should be matched, otherwise equal base current, produced by equal voltage drop in the two 10 k resistors, will not produce equal collector currents. If the intermediate amplifier is only single-ended, which can be just as good as a balanced amplifier, and the input shown for *Figure 1* is single-ended, one of the 1 k collector resistors may be omitted, and the output taken from the other. In this event, precise matching of Q1 and Q2 is not so vital to balance.

On the other hand, the output circuit could be rendered balanced by coupling push-pull signals directly to the bases of Q2, Q5. Then it would be necessary to control the quiescent voltage of both sides so base-to-supply voltage of these transistors matched that of transistors Q4 and Q3.

While some form of direct coupling is not impossible, especially as the d.c.

Neve

the sound of Neve is world wide

Built for the Whitney Recording Studio Inc., Glendale, California. Incorporates 24 input channels fully matrixed to 16 plus 2 output groups, 24 track monitor matrix, comprehensive reverberation, cine and talk-back facilities and 8 limiter/compressor units.

AT THE A.E.S EXHIBITION, HOLLYWOOD

NOW IN U.S.A.

Rupert Neve have opened a manufacturing, distributing and servicing plant in the U.S.A. for still better service.

The 'S' range of Consoles is available for very quick delivery.
Consultant Designed Consoles to customer requirement.

RUPERT NEVE & CO LTD Cambridge House, Melbourn, Royston, Herts. SG8 6AU, England.
Telephone : Melbourn 776 (STD 076 386) 10 lines. Telex 81381

RUPERT NEVE INC. P.O. Box 122, Bethel, Connecticut 06801. Tel. (203) 744 6230 : Telex 969638.

Circle 19 on Reader Service Card

value of 3 volts is common throughout the circuit, it is not too feasible, because the quiescent condition should be set up and maintained quite precisely, for both input and output stages, as well as any in between. This is easier to achieve by using a.c. coupling with separate values to control quiescent at input and output.

The constant-current circuits set up by the transistors with zener diodes exhibit a quite high collector resistance at that constant current. The zener voltage will not change more than a small fraction of a volt, and the emitter voltage of the transistor to which the zener controls the base voltage will follow it, again within a small fraction of a volt.

Thus the emitter resistor controls the emitter current very closely indeed. The only possible cause of collector current change is due to change of current gain with collector voltage, which is small with most transistors. And anyway, this change is compensated for by the fact that the zener will change its current so the base current adapts.

Put more directly, this change can only change collector current by the amount of change in the base current, because the emitter current is held constant. Thus, if current gain is 60, the collector current change must be well within 1/60th of its value, almost certainly well within 1/100th.

Thus the floating effect provided by each of these circuits is extremely good. A really high-quality transformer may do better at mid-frequency, if the tappings are accurate to a small fraction of 1 per cent (which this type of transformer commonly provides). But at frequency extremes, where winding capacitances and magnetizing currents of a transformer invalidate its precision somewhat, the transistorized version can actually improve on the older transformer-ized version.

As an old-time transformer-designer, who loved the challenge of this kind of design, this is a little bit of a blow to my ego—but at the same time, a bit of a relief, too. ■

Crown

PROFESSIONAL STUDIO EQUIPMENT

Specs	15ips	7½ips
w. & fl.	0.06%	0.09%
f. resp. ±2dB	40Hz to 30kHz	20Hz to 20kHz
S/N	-60dB	-60dB

CX822

RECORDERS & REPRODUCERS

SX711 Claimed by its pro audio owners to be the finest professional tape recorder value on the market today - price versus performance ■ Frequency response at 7½ips ±2dB 20Hz-20kHz, at 3¾ips ±2dB 20Hz-10kHz ■ Wow & flutter at 7½ips 0.09%, at 3¾ips 0.18% ■ S/N at 7½ips -60dB, at 3¾ips –55dB ■ Facilities: bias metering and adjustment, third head monitor with A/B switch, sound-with-sound, two mic or line inputs, meter monitoring same as CX822, 600Ω output ■ Remote start/stop optional, automatic stop in play mode ■ $895 for full-track mono deck as shown, $995 for half-track stereo deck

SP722 Ideal reproducer for automation systems ■ Meets or exceeds all NAB standards ■ Remote start/stop optional, automatic stop in play mode ■ $595 for half-track stereo reproducer

Crown tape recorders and reproducers are available in 42 models with almost any head configuration, including 4 channels in-line. Patented electro-magnetic brakes maintain ultra-light tape tension and never need adjusting. They are made by American craftsmen to professional quality standards, with industrial-grade construction for years of heavy use.

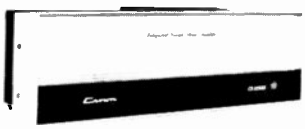
All Crown amplifiers are warranteed three years for parts and labor. They are 100% American-made to professional quality standards. All are fully protected against shorts, mismatch and open circuits. Construction is industrial-grade for years of continuous operation.

For more information, write CROWN, Box 1000, Elkhart, Indiana 46514

STUDIO MONITOR AMPLIFIERS

D40

Delivers 40 watts RMS per channel at 4Ω ■ Takes only 1¾" rack space, weighs 8½ lbs. ■ IM distortion less than 0.3% from 1/10w to 30w at 8Ω ■ S/N 100dB below 30w output ■ $229 rack mount



D150

Delivers 75 watts RMS both channels at 8Ω ■ IM distortion less than 0.1% from 1/10w to 75w at 8Ω ■ S/N 100dB below 75w output ■ Takes 5¼" rack space, weighs 20 lbs. ■ $429 rack mount

DC300

Delivers 300 watts RMS per channel at 4Ω ■ IM distortion less than 0.1% 1/10w-150w at 8Ω ■ S/N 100dB below 150w output at 8Ω ■ Lab Standard performance and reliability ■ "As close to absolute perfection as any amplifier we have ever seen" - Audio magazine, 10/69 ■ $685 rack mount

See Crown demonstrated at the Los Angeles AES Show, Room 53

Circle 18 on Reader Service Card

John M. Woram

THE SYNC TRACK

• Since this April issue of **db** is featuring various aspects of the microphone, it is more than appropriate to discuss microphones in this column.

By an odd coincidence, it was just two years ago this month that a feature story by the well-known R. D. Worther was to have been published, revealing certain innovations in surveillance equipment, including an ultra-miniature microphone and transmitter concealed within—of all things—a common 6 penny nail. I say, "was to have been published", for as readers will recall, the page plates were confiscated at the last minute by the authorities, and Mr. Worther was never seen again.

At that time, there was considerable speculation as to Mr. Worther's fate. One Federal bureau claimed his body had been found by pro-western kulaks near an obscure railway junction on the Peking branch of the Trans-Siberian Express. More recently, some American tourists vacationing in the Tirenean Alps have reported being approached by a person reportedly resembling Mr. Worther. Unfortunately, the man was spirited away by the local police, and again we are plunged in mystery.

And so, although we continue to speculate on the whereabouts of the enigmatic Mr. Worther, we have been more fortunate in recovering the missing page plates. Some time ago, they turned up in a locker at a small mid-western bus depot. Although we cannot divulge how they were discovered—nor have we in fact seen more than a copy of them—our informant's credentials are impeccable, and he has told us enough to convince the editors of his absolute reliability.

Since this is a technical journal, and political reportage is really beyond our domain, we shall confine the rest of this column to the relevant aspects of this most unusual case; specifically, a description of the surveillance microphone.

The development of the ultra-miniature microphone was not without a large share of problems. It hardly need be mentioned that a microphone concealed within a 6 penny nail would have to be wireless. Obviously enough, any wire connected to a nail would be cause for a certain amount of suspicion by even the most naive. After a bit of experimentation, a unique d.c. carrier system was perfected. Once the "nail" was hammered into the appropriate wall, the transmitter would be powered by a unique application of nature's laws of molecular dispersion. By chemical reaction, the nail would derive the requisite power from the lime content in the plaster wall. Unfortunately, the chemical reaction eventually discolors the plaster surrounding the nail, thereby revealing its presence, so for longer range applications, a wall that has been covered by wallpaper or panelling must be chosen.

Obviously, there is no particular advantage to an omnidirectional polar pattern, since the microphone is to be hammered into the wall. Generally, a cardioid pattern is most suitable, since proximity effects need not be considered important. But the nature of surveillance work usually means that a good portion of the program being monitored will be in some foreign tongue. Since the untrained ear often has difficulty differentiating between several alien speakers, it is often a good idea to employ two nails for a stereo pickup. That way, the voices can at least be localized, which should help in minimizing confusion. It should of course be remembered that the distance from either nail to the speaker(s) should be about one quarter the distance between the two nails, to prevent phase shift distortion.

Considerable time and money was spent developing a mic diaphragm that was rugged enough to be hammered into a wall, yet sensitive enough to provide wide range pickup characteristics.

Shortly after the first batch of microphones was completed, a near disaster almost cancelled the entire project. The nail-mics were sent to a reliable agent in the newly emergent nation of Kallabriya, where a neutral nation was constructing an embassy in the capitol city of Wredjiyo. Unfortunately, the surveillance nails were mixed with the regular construction supplies, and the entire supply was hammered into the embassy walls. To compound the error, the receiver was concealed in an ice-cream vendor's cart in the street outside the embassy. The vendor was of course another trusted agent, and the hearing aid he wore was in reality a speaker connected to the concealed receiver.

When the poor man switched on the receiver, the massive surge current representing hundreds of nails connected, as it were, in parallel, launched the cart swiftly down the street until it collided with the limousine of the just-arrived Russian ambassador. The agent was thrown clear of the cart and landed almost in the lap of the diplomat. Since the ambassador had spent considerable time in Washington as a minor official, he saw nothing unusual in this peculiar demonstration, and in fact ordered ice cream cones for all the members of his entourage. No doubt the well-known Russian weakness for ice cream played an important part in preventing what otherwise might have become an unpleasant international incident.

Recent reports from sources within the embassy reveal that the walls are now beginning to deteriorate due to the massive chemical reaction from so many nails, and that embassy officials are beginning to suspect something is amiss.

On a more positive note, the issuance of the next batch of nails was more carefully administered, and the devices have since been judiciously hammered into key walls throughout the world.

Our fears that the unique device might fall into the wrong hands were quickly dispelled. We had wondered what would happen if some nails were secreted in the Capitol's various caucus rooms and the information leaked to the enemy. Our informant correctly reminded us that it has been some years since anything worth repeating has been heard on Capitol Hill, and although some nails had indeed been

NEW PRO-4AA

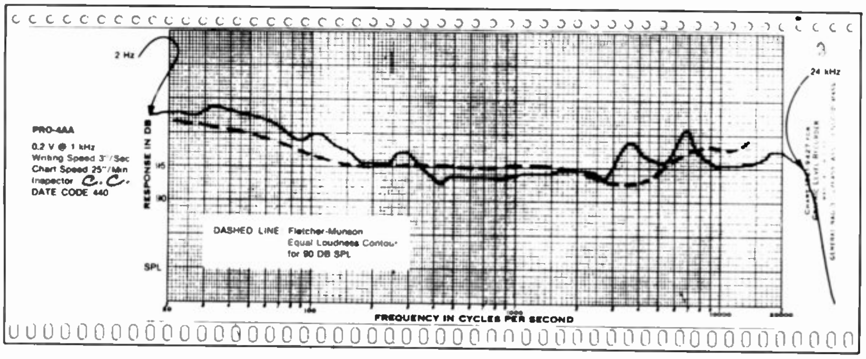
The good sound you monitor is the Sound of KOSS

Better than speakers

Here is a new, rugged, wide-range dynamic headset with good sensitivity that delivers precision monitoring far beyond that obtained by speakers. You exclude outside noise; you eliminate disturbing room reflections. Freedom from distortion and flatness of response are exceeded only by the famous Koss electrostatic units. You'll find that the "Sound of Koss" is the *good* sound.

Price **$60.00**

TYPICAL RESPONSE CURVE OF PRO-4AA



WHAT THE SPECS DON'T TELL YOU—

EXTENDED BASS RESPONSE TO 2 CYCLES/SECOND—The compliance of the diaphragm is large, the seal is good against the ear and the physical design permits very long excursions to give you lo-lo bass at full level—and the PRO-4AA does it with liquid-filled cushions for perfect seal and superlative comfort, whether you wear glasses or not. This bass range response *exceeds by 4 octaves* that needed to insure good low-end monitoring.

TAILORED CURVE SHAPE FOR FATIGUE-FREE LISTENING — Characteristically the PRO-4AA runs ± 3 dB except below 150 Hz and for a rise at 6.5 kHz. As long as we must have *some* variation, we put the belly of the curve in the mid-frequencies to follow closely the Fletcher-Munson equal loudness contour for 90 db SPL at 1 kHz. This keeps fatigue low in long, intense sessions because you don't reach for the extremes of the range as you tire and your ears become less sensitive. Last, but not least, the music sounds *good*.

HIGHER HIGHS THAN YOU CAN USE—The 24 kHz point on the chart is about where the "2" is in the figure 1521—the PRO-4AA needs longer chart paper than this standard charting equipment provides. Even though you can't hear these frequencies, the capability promotes good transient response in the range you *do* hear.

WHY DOES KOSS RAISE THE BOTTOM?—The wide-range coupler used for Koss measurements effects a perfect seal to promote high bass level almost to dc. An air leak to the sealed cavity lowers the bass response level. Koss feels that modern side-burns and luxuriant hair are good for a full 1-second leak or 4 dB less. This makes the PRO-4AA the flattest dynamic headset we know how to design at this time!

See and hear the PRO-4AA at your dealer today or write the factory for a 16-page catalog on "How to Choose Stereophones."

ELECTRICAL SPECIFICATIONS

Frequency Response Range, Typical: 10-20,000 Hz ■ **Efficiency:** Medium ■ **Total Harmonic Distortion:** Negligible at 95 dB SPL. ■ **Source Impedance:** Designed to work direct from 4-16 ohm amplifier outputs. When used with headphone jacks where series resistors are employed, response should not be measurably affected, but slightly higher volume settings will be required. ■ **Power Handling Capability:** Maximum continuous program material should not exceed 5 volts as read by an ac VTVM (Ballantine meter 310B or equal) with average indicating circuitry and rms calibrated scale; provides for transient peaks 14 dB beyond the continuous level of 5 volts.

PHYSICAL SPECIFICATIONS

Cushions: Fluid-filled for high ambient noise isolation averaging 40 dB throughout the audible range. ■ **Headband:** Extendable, stainless steel with self-adjusting, pivoting yokes; conforms to any head size. ■ **Boom Mount for Microphone:** Knurled, anodized, aluminum knob on left cup with threaded shaft and 2 compressible rubber washers; accepts all standard booms. ■ **Headset Cable:** Flexible, 4 conductor coiled cord, 3 feet coiled, 10 feet extended. ■ **Plug:** Standard tip, ring and sleeve phone plug. ■ **Element:** One inch voice coil virtually "blow-out proof"; takes surges up to 20 times rated maximum power levels. Has 4 square inches of radiating area from 2 mil thick mylar diaphragm. ■ **Weight of Headset Only:** 19 ounces.

KOSS Stereophones

Koss Electronics, Inc., 4129 N. Port Washington Rd., Milwaukee, Wis. 53212 • Koss Electronics S.r.l. Via Vallorta, 21 20127 Milan, Italy

Circle 17 on Reader Service Card

discovered during the remodelling of a Congressional cloak room, they had become corroded over from lack of use. ■

41st A.E.S. Convention

Although the 40th A.E.S. convention is just now coming, it is not too early to begin making plans to attend the 41st convention, to be held in New York City this fall.

The following twelve technical sessions have been scheduled, and persons interested in submitting papers should contact the appropriate session chairmen as soon as possible.

Technical Session	Session Chairman
Transducers	James Novak Jensen Mfg. Co. 5655 West 73rd St. Chicago, Ill. 60638
Electronic Music	David Friend Tonus, Inc. 45 Kenneth Street Newton Highlands, Mass 02161
Design of Audio Transmission Systems	John Woram RCA Records 1133 Ave. of the Americas New York, N. Y. 10036
Disc Recording and Reproduction	Lawrence Shaper Empire Scientific Corp. 1055 Stewart Avenue Garden City, N. Y. 11530
Broadcasting	Leonard Feldman Engineering Consultant 97 Oxford Blvd. Great Neck, N. Y. 11023
Magnetic Recording and Reproduction	Marvin Camras IIT Research Institute 10 West 35th Street Chicago, Ill. 60616
Sound Reinforcement and Architectural Acoustics	Peter Tappan Bolt, Beranek and Newman 1740 Ogden Avenue Downders Grove, Ill. 60515
Medical Electronics	Philip Kantrowitz 2435 Frisby Avenue New York, N. Y. 10461
Digital Techniques in Audio	Ronald Schafer Bell Telephone Laboratories Murray Hill, N. J. 07974
Audio Instrumentation and Measurements	Emil Torick CBS Laboratories 227 High Ridge Road Stamford, Conn. 06905
Amplifiers and Signal Processing Devices	Saul Walker Automated Processes, Inc. 35 Central Drive Farmingdale, N. Y. 11735
Acoustical Noise Control	William Siekman Riverbank Acoustical Labs P. O. Box 189 Geneva, Ill. 60134

Martin Dickstein

SOUND WITH IMAGES

• These pages started reporting in depth on the latest development in home entertainment when the article on the CBS *EVR* invention was printed in the September, 1968, issue.

Then, in the December, 1969, issue, I followed up with a description of the RCA unit called *SelectaVision*. In April, 1970, I continued with a discussion of the Sony entrance into the home entertainment field with its *Video-player*, and in October, 1970, I described the demonstration by CBS at the AMA show in New York of the EVR. The December, 1970, and January, 1971, issues of db provided full discussions of the newest entrant, the *Video Disc* by Teldec. In December I also provided a chart comparing some of the estimated projected consumer costs and some of the markets in which the different systems would probably find their first applications. (Incidentally, I plan to update the chart as more information becomes available and the units themselves are readied for market introduction.)

For about the past year and a half, there have been meetings in almost every field of endeavor—in which audio-visual equipment plays any part at all—to discuss the meaning, impact, uses and applications, the best time to jump in, and to what extent (if at all), initial expenditures, and the future of the latest and apparently greatest method of reaching the public.

One such recent meeting was held toward the end of last year by the National Visual Communications Association, New York. The organization was founded in 1952 "to provide industry executives and professional visual communicators with an opportunity for an exchange of ideas and a source of information on techniques and new developments." The Association is a non-profit professional and scientific organization with a membership open to all persons actively engaged in visual communications whether in sales, service, manufacturing, or in the use and application of the equipment.

At this conference, the 17th annual Days of Visuals, there was a small display of equipment and software for audio-visual application and also talk and discussion sessions. Speakers at the various meetings included Mr. Leslie H. Waddington, President of NVCA, on the subject "Media 70's—What Next?", Mr. Malcolm E. Shaw, President of Educational Systems and Designs, Inc., Westport, Conn., on "Television in Management and Sales Training." At one session, Eastman Kodak representatives discussed the subject "Concepts in Communication."

One entire afternoon of the two-day convention was devoted to a discussion of several different devices being developed in the video cassette field. A talk was scheduled on the EVR by a representative of Motorola, the manufacturer of the player unit for the CBS cartridge; the Sony Videoplayer; the Ampex *Instavision*; and the AVCO *Cartrivision* system. Unfortunately, the representative of Sony could not attend, but the other

(Continued on page 23)

If performance turns you on turn on the Scully 100

Then to really blow your mind look at the price tag

It's a fact. Scully *has* put it all together. A 16-track, professional studio recorder/reproducer that actually out-performs recorders costing at least twice the 100's $13,700.

The big secret? Like simple. Take the same Scully engineers that design the studio equipment that's been the standard of the business for years. Let them come up with the first really modular unit that lets you buy *only* what you need. Forget all the factory-loaded accessories and extras if you already have them on your consoles. Add them if (and when) you need them.

Then let Scully offer a totally new combined record/playback head, spill-proof silent switching, and a completely new solid-state electronics package. The result? The 100 Series. Half the size and half the cost of available equipment. And performance specs that are outta' sight!

For a demo that'll let you see and hear what we're talking about, write Scully, 480 Bunnell St., Bridgeport, Conn. 06607 (203 335-5146). We'll send you complete 100 specs, and the name of your nearest friendly Scully distributor or Sales/Service office. (There's probably one a dime phone call away.)

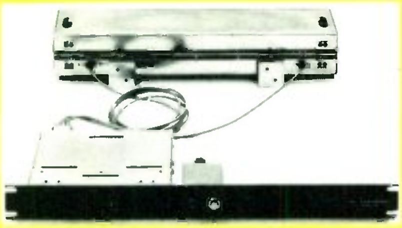
What can you lose . . . except all your ideas about how big and how expensive professional recorders really are?

Dictaphone

Scully Division

NEW PRODUCTS AND SERVICES

REVERB UNIT



• The O/P Reverberation is designed for use in those professional recording studios whose people wish a substantial improvement in the sound quality obtainable from a spring. Flutter is reduced in the conventional way—by using multiple springs, while noise is kept below audibility by means of correct design of the pickup amplifier. A floating threshold peak limiter is included. This reduces objectionable noises in the form of popping springs on transients. Console and rack mount, with or without power supply is available.

Mfr: Parasound, Inc.
Circle 58 on Reader Service Card

CASSETTE DUPLICATORS

• New models have been added to this existing line. As the model DC1542/30 they duplicate two-track cassettes while as the model DC1544/30 4-track capability is provided. The master unit will accept 3¾ or 7½ in./sec tapes. Three large hysteresis synchronous motors are used for the capstan drives and an additional five torque motors used for the takeup function. Heavy balanced brass flywheels driven by flat belts are used in the cassette mechanism to cut flutter to less than 0.2 per cent. The time needed to duplicate four copies is one eighth the time required to normally play one cassette. Additional slave units can be used to a capacity of 20 copies at a time.

Mfr: C.E.E.
Price: $2825 (two track)
$3450 (four track)
Circle 56 on Reader Service Card

TELEPHONE TRANSMITTER/RECEIVER

• Here is a quick, easy, and inexpensive way to transmit or receive recorded information over the telephone with high quality. The PC-48 telephone transmit/receive coupler provides unmatched convenience for transmission of recorded material from *any* tape recorder through any standard telephone. Simply plug the jack into the output of the recorder and slip the loop over a telephone earpiece. Recorded material is heard at the other end without distortion and may easily be monitored and edited. The PC-48 also doubles as a high resolution telephone pickup by merely plugging the jack into the tape recorder input.

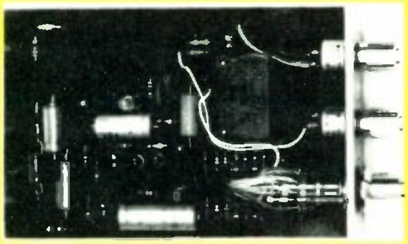
Mfr: Trinetics, Inc.
Price: $9.95
Circle 68 on Reader Service Card

PAGING PROJECTORS

• The first of a series of newly designed, low-cost, paging projectors. the 12-watt PA12 has been introduced. It features computer-calculated horn flare, a design factor that provides excellent response characteristics and dispersion. A newly-designed diaphragm and voice-coil assembly. plus a powerful Alnico V magnet structure make these units highly efficient, requiring less amplifier power for a desired sound pressure level output than other speakers. Excellent speech articulation and intelligibility and a voice-coil impedance of 8 ohms are offered. The round horn provides a nominal 130-degree dispersion angle. It may be oriented in any desired position in a vertical plane by loosening a single wingnut on the mounting base. The housing is high-pressure injection molded, has high resistance to impact, and is extremely sturdy. Its molded-in Mesa tan colored finish will not fade, chip, or peel. Frequency response of the PA12 is 325 to 14,000 Hz. For use in line-voltage installations, the company offers the TR12 transformer in 70.7 or 25-watt versions for the PA12 speaker. This transformer mounts on a special bracket at the rear of the PA12 horn and has solderless push-clips for selecting wattage taps. The cover for the TR12 is made of special non-yellowing clear formed acetate allowing easy inspection of connections. The transformer has been vacuum varnished for full protection from the weather, moisture, and fungus.

Mfr: Electro-Voice Inc.
Price: PA12–$27.00
TR12–$10.00
Circle 73 on Reader Service Card

TEST OSCILLATOR CARD



• Model 692-OSC test oscillator card has been designed to complement the INTEGRA II 692 card series. It packaged on a card 3½ inches wide by 5½ inches long and can be used in conjunction with the 692 cards, or can be inserted in a special card holder for complete shielding from adjacent equipment as well as for mounting into a 5¼ inch rack mounting frame. The unit covers a complete audio range in one continuous sweep from 20 Hz to 15 kHz, or in five selected frequencies from 20 Hz to 15 kHz. A unique feature is that frequency can be remotely controlled by simple modification. Specifications are: Output level + 10 dBm; distortion 0.2 per cent maximum; output uniformity within 1 dB; output impedance 3 ohms; mic feed output impedance 200 ohms. Power requirements 24 V at 100 mA with remote control, 10 mA without remote control.

Mfr: Fairchild Sound Equipment Corp.

Circle 70 on Reader Service Card

TAPE CLEANER

• Although marketed as a video tape cleaner, 2-inch audio tape can probably also benefit from the Magnetek 1 video tape cleaner system. It has been designed to acomodate 1 and 2-inch tapes and to remove physical errors (dirt, oxide particles, etc.) from the tape surface. Simple to operate, only a minimum of operator skill is required. In the cleaning process the oxide surface of the tape passes over precision-ground tungsten carbide blades which remove loose or embedded oxide particles, backing materials, or other foreign matter. Embedded particles are sliced to avoid depressions or pitholes. Particles are then wiped simiItaneously from both surfaces of the tape by continuously moving silicon-impregnated tissue synchronized with the tape movement.

Mfr: Advanced Transducer Systems, Ltd.

Circle 57 on Reader Service Card

PULSE GENERATOR ADAPTER

• This unique device is a low-cost laboratory pulse generator which claims not to sacrifice quality or reliability. It is designed to be driven from any oscillator, such as a sine-square wave generator, and may be operated at any frequency from 1 Hz to 10 MHz dependent only on the driver. The output of the pulse generator adapter offers variable amplitude; 0 to 12 volts; variable pulse width, 5 millisecond to 100 nanoseconds; and variable frequency. The rise time of the output pulses are up to 20 nanoseconds. Output impedence is 50 ohms and is compatible with TTL, making it the ideal instrument for logic development.

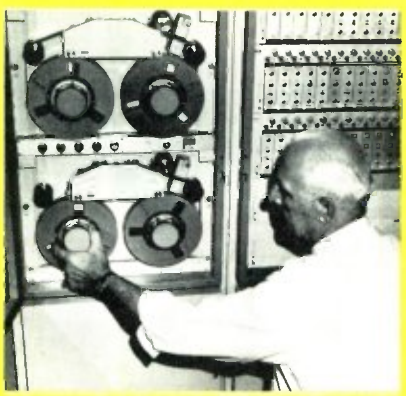
Mfr: Blulyne Electronics Corp.

Price: $59.95

Circle 72 on Reader Service Card

MULTI-CHANNEL RECORDER

• Simultaneous recordings of as many as 31 different incoming and outgoint messages is possible with this new recorder. A single reel of one-inch tape provides a full 12-hour recording of all 31 channels. When playback is necessary, an electronic timing device shows, to the second, the time the message was originally recorded. The Paterson, N. J. Police Dept. is now using this equipment for continuous communications recording. All radio instructions and information between headquarters and field forces, as well as telephone calls to headquarters are recorded. Some phone lines are not recorded—all recorded phone calls hear the customary "beep". The system is only 72-inches high in two racks. While one tape deck records for twelve hours, the second and third are on automatic standby or may be used for playback or rewinding. In the event of failure, a stand-by deck would take over. Recording is at 15/16 ips, yet quality is extremely clear. Response is claimed +3 dB from 300 to 3000 Hz. It is anticipated that radio stations, airports, courts, railways, and electric power stations will find use for equipment of this type.

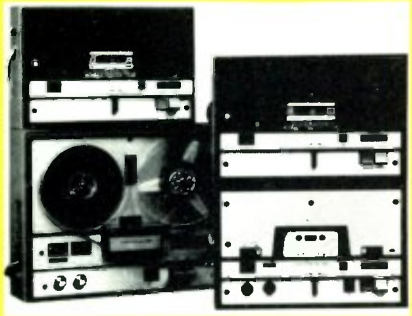


Mfr: Philips Broadcast Equip. Corp.

Circle 61 on Reader Service Card

CASSETTE TAPE DUPLICATING SYSTEM

• A unique concept in cassette duplicating systems, incorporating the heavy-duty, high performance mechanism of the Wollensak audio-visual cassette recorder, has been developed. The modular system permits placement and operation in a variety of work layouts or space arrangements. Because the system is designed on a modular, plug-in basis, the user can start small—with a single master and copier—and add units as his needs develop. Either of the two master units—reel-to-reel (model 6040 AV) and cassette (model 2750 AV)—can drive up to 10 cassette copiers or slaves (model 2760 AV). Two unique and patented features provide automatic high-speed rewind of copies and automatic sensing of stalled cassettes. The latter feature assures that the system makes only perfect copies. Duplicating speed is four times program speed. Both master and copier units are available in either two or four-track formats. The reel-to-reel master will accept master tapes at three speeds—7 1/2, 3 3/4 and 1 7/8 in./sec.



Mfr: 3M Co.
Price: $499.95 (master unit); $299.95 (copiers)
Circle 77 on Reader Service Card

BROADCAST AUDIO AMPLIFIERS

• A new solid-state, 50-watt plug-in audio amplifier with high gain and low distortion is available for use as a monitoring amplifier by broadcast and recording studios. It is capable of driving 4-, 8- and 16-ohm speakers or a 70-volt line for sound distribution and reinforcement systems. The amplifier, type BA-48A, produces 50 watts continuous, with or without optional output transformer, with total harmonic distortion of less than 0.5 per cent from 20 to 20,000 Hz. The BA-48A is designed for plug-in installation in the BR-22 mounting shelf which accommodates two amplifiers. Accessories include remote gain control module, input, bridging, and output transformers.
Mfr: RCA
Circle 69 on Reader Service Card

PENCIL-SIZED TESTER

• A pocket-sized instrument which indicates direction of flux lines and measures relative strength of any permanent magnet or magnetic field has been introduced. Said to be the only testing device of its type, the MagTester utilizes high-strength magnetic materials designed to resist demagnetization or polarity reversal. A magnetic wheel mounted at one end of the MagTester, with a north/south axis across its face, will align itself parallel to the flux lines of a magnetic field, thus indicating polarity. Testing the relative strength of a magnet is based on the principle of repulsion. By holding one end of the unit against the similar pole of the magnet being tested, an internal rod magnet is repelled upward within a clear plastic tube. The relative strength of the magnet under test is indicated by the length of travel of the rod magnet. A numbered scale provides a handy measure for comparison purpose.
Mfr: Sel-Rex Corp.
Price: $19.95
Circle 67 on Reader Service Card

REVERB SYSTEM

• The RV-10 is a new and patented variable delay reverb system that uses a fresh approach to generate mechanical reverberation. A 55 ms transducer delay resembles the artificial delay times used on other devices for modern recording studio techniques. Front panel adjustment of decay time and low frequency filtering permit this system to match other reverb systems and also to create new effects not available in other devices. An almost total immunity to mechanical shock or outside acoustical pickup is claimed, making this unit feasible for control room installations. Input sensitivity is +4 dBm. However, levels down to −20 dBm can be accommodated by internal strapping.
Mfr: Quad-Eight Electronics
Price: under $800
Circle 80 on Reader Service Card

(From page 17)

gentlemen provided interesting talks and literature of their respective systems, rough costs of the units and software, future developments, markets and applications.

Ampex, in describing the system of Instavision, indicated that the medium of half-inch magnetic tape, which permitted re-use by the consumer as many times as desired for home recording either from a camera or the t.v. set, is the most versatile. Their tape is contained in a cartridge from which the tape is threaded automatically when played on an Instavision unit. However, as the record/playback specifications conform with the Type 1 standard, the tape can be played on a machine other than the Instavision unit that also conforms to the same standards with use of an accessory hub insert.

Among the other features stressed by Ampex are the provision for two independent audio channels for stereo or language studies, slow speed or stop motion for study of single frames in industry or medical applications, simplicity of use for public use at home parties, etc. Still another feature is the auto-search provision which permits the unit to sense the signal put automatically on the tape when a recording is completed. Thus, the next time, the end of the preceding recording can be found easily; this results in a cueing or indexing system for different sections of recordings. Total recording time at standard speed is 30 minutes, or 60 minutes in the extended play mode.

This system, which according to the literature has been "designed for use in education, business, industry, medicine, sports and government applications," including the camera, also provided for simple electronic editing of program material from the camera—avoiding picture roll or tear. Sync signals are furnished by the recorder for the camera. The camera has a small electronic viewfinder which permits instant playback of the taped material. The recorder also has "digital" indication and automatic head cleaning (by just pressing a button on the machine).

Cartrivision, the name given to the system placed in the running for the money by AVCO, includes a color t.v. set in a cabinet with the video tape player mounted within the cabinet . . . sort of a video hi-fi unit. With a special adapter, the playback unit can also be played through any other t.v. set as well.

The medium of this system is also half-inch magnetic tape. AVCO will also provide, at extra cost, a camera for use by the consumer for home recording. This blank tape will be marketed in a yellow box for easy identification. One additional feature is an automatic timer which will permit recording from the t.v. set even though the buyer is not at home at the time as the desired program goes on the air. The recording will also turn itself off when finished. The cartridge will run in sizes from one-half hour to two hrs., and is of the reel-to-reel type with one reel above the other.

The reason for the special color of the box is that AVCO will also sell recorded tapes (movies, courses, and classes in various fields, sports events, etc.) in black boxes. Another color, red, will be used for boxes containing reels with material which will be available only for rental, and handled probably by a local outlet or library.

The only unit presently already on the market is the CBS EVR system. This system offers a playback-only capability. The unit is being made by Motorola for feeding color or black-and-white pictures to the home t.v. set. Thus, only recorded material will be available on the market in this system. However, provision has been made to connect a t.v. camera to the playback unit to feed live pickups to the t.v. set without recording.

The medium is a very thin plastic film fed through the machine like magnetic tape (without sprockets). The cartridge is completely sealed to prevent dirt or finger marks from getting on the film. The film is sealed until the cartridge is placed on the machine at which time small "fingers" push the end of the film out of the cartridge into a threading system which, in turn, automatically feeds to a take-up reel.

Total running time of the cartridge is fifty minutes of b/w or twenty-five minutes of color material. Since both visual tracks of the film have to be used simultaneously for color reproduction (b/w image in one track and color information in the other) the playback time is shorter but both audio tracks can now be used for stereo sound. With a b/w film, both tracks of visual information are used independently for full playback time but only one channel of audio is available with each track. Synchronization is achieved and maintained by a visual marker at each frame. This can not be seen on the screen. It is used by the machine to adjust for constant speed.

CBS has already contracted with many industrial firms to provide them with EVR cartridges containing the customer's original material converted to the CBS medium. Films and other previously-recorded material will also be made available for purchase by CBS. They are contracting for famous motion-pictures, sports events and other programs of interest for purchase through local outlets or record stores.

At present, this unit has a running head start in the video-cassette race since the equipment is already available while the other manufacturers are still getting theirs ready for marketing. One further step CBS has taken, in addition to transferring the customer's own material to EVR, is to provide full services of its special projects group to assist in designing the entire program for EVR recording from scratch even if the client does not have any previously-recorded material at all. The service extends from consultation through story boards, scripts, production, shooting, and the final step of recording and delivery of EVR cartridges.

Although Sony was not represented at the meeting, it is only fair to mention that their system uses a reel-to-reel cassette with magnetic tape. When we described the system shortly after it was presented, about a year ago, we concluded with the question "Can commercials be far behind?" Here it is just one year later and the answer is upon us.

Last month, another meeting was held to discuss the impact of the video cassette business. This t.v. seminar was held by the American Association of Advertising Agencies. The speakers were Mr. Morton Dubin, president of Video Tape Producers Association; Mr. Paul Caravatt, Jr., senior v.p. of Interpublic, a group of Ad Agencies; Mr. George Tompkins, general executive, Electrographic Corp.; Mr. Harland Kleiman, v.p. Video Cassette Division of Teletronics International, Inc.; and Mr. Paul Klein, president of Computer Television, Inc. The chairman and moderator for the meeting was Mr. Phillip L. Tomalin, senior v.p. of Ogilvy & Mather Inc., Advertising Agency and vice chairman of the 4A Broadcast Administration Committee.

Among the suggestions of how the cassette (or cartridge or disc) might be distributed for the public was the local library (for loan as presently with books and records), rental (also from the local library or book or record stores as with present best sellers), or by purchase from local stores or supermarkets as presently with records or books, or in give-away programs tied in with specific products through food chains, specialty stores, and chain stores.

Mr. Caravatt, with simple mathematics, showed how the software market could go as high as 11 billion dollars in the next decade or so. He mentioned some of the programming opportunities for classrooms, libraries, sales, and medical training, old or specially produced motion pictures and Broadway shows, travelogues, cartoons, highway-safety messages, political programs, special programs for the handicapped, home economics for the housewife, etc. He described the new medium as "Telechoice", an all-inclusive word for the diverse systems with which the public would have almost unlimited choice of what to watch on their t.v. set.

Mr. Tomkins explained how video cassettes would be borrowed from local libraries and would be a perfect means for advertisers and their agencies to produce special how-to programs for hobbyists and enthusiasts in every conceivable field, with inclusion of the manufacturer's products as part of the program. Where else will the advertiser have the opportunity to reach the precise customer he wants to buy the products specially made for that person?

Mr. Klein said that there is no question but that the advertiser and the agencies would play a very important part in the future of the video cassetts production and development business, but it was his contention that the way to go was not necessarily with the customer buying the cartridges. He felt that by having a computerized central location for cartridges (or cassettes or discs), the customer could dial for any information or program material he wished by referring to a listing (to which he would subscribe) with a special code. The home set would be tied to the central system with a cable (as is now being done in areas with c.a.t.v. installations). This way, the home viewer could have program material available for every channel on the set, not just the few that cable and air broadcast. Charges for the service could be through the customer by subscription, charges on the phone system or special dialing system for getting the special material desired and could extend to advertisers who furnished material for cable transmission.

According to the experts who are taking this new business very seriously, it will be up to the sponsors and advertisers and their agencies to produce more and better program material with which to attract the viewing public, including, of course, commercials. First, the public will have to decide which type of system to buy (or maybe one of each, as at present, with a record player and tape machine) and then have a multitude of choices for viewing material. The agencies will have to decide how best to reach the public with so much choice at his disposal. Decisions! ■

A Forum on Microphones

Here is what happens when you place several recording engineers in a room, give them a topic to discuss, and turn on the tape recorders.

SOME weeks ago, **db** held the first in an anticipated series of informal meetings, bringing together engineers from several recording studios to discuss topics of mutual interest. On this occasion, the subject was microphones.

Bob Schulein, manager, Electroacoustical Systems at Shure Bros. Inc., flew in to meet with the group and add the manufacturer's point of view.

Beyond calling the evening, *A Forum on Microphones*, no particular format was prescribed, and the meeting quickly became a lively bull session, with everyone contributing his own viewpoint—arguing at times, agreeing at others. It must have been a successful evening, for the original typed transcript of what was said ran to over 125 pages, from which we have distilled what follows.

Schulein: Being a design engineer, I'm concerned with building a better microphone. I have an advantage in that I can make accurate and repeatable laboratory measurements. However, I don't have any first hand knowledge of how you people are using microphones, or what you are looking for. You know what you like to hear, and you know when you're getting it. But you may not know *why* one mic works better than another—you just trust your ears, because if you don't like what you're hearing, all the charts and specs in the world aren't going to get you to change your mind.

Woram: I think most recording engineers, unless they're particularly interested in specifications, couldn't tell you much about the published data on their favorite mics. However, they certainly could tell you in subjective terms which mic they prefer for any particular application.

Schulein: To you, a microphone is just another link in the complete chain. In a given situation, you know that along with all the other variables—equalization, limiting, your console, tape recorder, producers taste and so on—a certain mic will do what you want it to. Back in my realm, the microphone is the most important thing in the world. I could tell you the differences between the mic you prefer and any other one, and if we can determine what the right measurements are, we may be able to predict which mic is the best for any application.

Bradley: With an exception for that all important variable—personal taste. I doubt if many of us would agree on any choice of microphone for a particular instrument.

Baxter: Right! John and I are always arguing over microphones. He really likes the Shure SM 76 on acoustic guitar. Most of the time, I don't.

Katz: My favorite acoustic guitar mic is a Neumann 87 with the bass roll-off switched in. The guitar can be a very subtle instrument. And a twelve-string especially has the tendency to have a run away low end. I'll put the 87 approximately over the hole and use a limiter to tighten it up a bit.

Figure 1. The group gathered around an omni to record the conversations that are here transcribed. The location is Studio A at RCA in New York City.

FORUM PARTICIPANTS

Steve Katz	Sound Exchange
John Bradley	Ultrasonic Recording Studios
Mike Colchamiro	Ultrasonic Recording Studios
Dick Baxter	RCA Record Division
John Woram	RCA Record Division
Larry Zide	db Magazine
Bob Bach	db Magazine

It would be interesting to find out what mics you fellows are now using, and then measure these mics in some way that would relate to how you're using them.

Baxter: Have you ever tried the omni pattern on the 87, and moved in even closer?

Katz: I don't think I could get in any closer. Six inches is about as close as you can get.

Woram: That's one of the reasons I prefer the SM 76. Since it's very small, you can get in as close as you like, and still not be in the musician's way. And since it's an omni, you don't have any proximity effect.

Schulein: You know, when I first got involved with the recording industry, I thought there might be a very simple way to work things: Just get a basically smooth microphone, and an equalizer on each mixer. After all, the biggest difference between microphones is their frequency response. If you could equalize them, I think you'd find many mics are quite similar, particularly omnis.

Colchamiro: That might be okay with omnis, but with cardioids, there are a lot of other considerations-proximity, off-axis response, working angle, and so on. All of these things play an important part in selecting your favorite mic for a particular instrument.

Schulein: What do you think about a so-called basic mic with a few variables? Say, a presence peak which you could switch in or out, and maybe a high end roll-off. And a three-position switch at the low end for a boost or a cut. And you had all this in either a uni or an omni.

Baxter: Do you need all that equalization at the mic? There's so much equalization done at the board now.

Schulein: Well then, a basic mic, and you could use your own equalizers.

Baxter: Then we get back to, what is a basic mic?

Colchamiro: A basic mic is one that accurately picks up what the performer is doing. Maybe *accurate* isn't the right word. It's just the mic that sounds best with maybe some equalization under the conditions at the moment.

Woram: I'm not much on equalizing everything. If I can't get what I want with a very slight amount of equalization, I'd rather go out and put up another mic. Listening to some tracks, you can almost say, "oh, it's plus 8 at 100 hertz." You can almost hear the equalizer.

Katz: Equalizing an instrument is such a delicate thing. You've got fundamental, overtones, and harmonics to consider. You can really upset this critical balance, and that's where the beauty of music is.

Woram: If you equalize a mic by adding plus 8 at 100, you're affecting the low end, and the further the instrument goes from playing its lowest notes, the further it is from the equalization. If you can try a different mic instead of putting in all that comp, I'd think you'd get the quality you're looking for over the entire range of the instrument.

Fifteen feet away, and at variable height, the condenser microphone was set up 53 inches above the floor . . .

Schulein: Either way, you're equalizing. You're either adding externally, or choosing a different mic with probably a different response of its own.

Bradley: Except that the internal differences between two mics are usually more varied over the entire range than anything you could do with a console equalizer, unless you had a very good graphic.

Schulein: It would be interesting to find out what mics you fellows are now using, and then measure these mics in some way that would relate to how you're using them. Then, we might pick a reference mic and I'd give you some specific equalization settings to get it to approach the performance of the other mics.

Katz: I'd like to know what everyone uses on vocals.

Woram: I usually try an Electro-Voice 635a first.

Bradley: One of the best vocals I ever got was with a Shure 546. That's the kind of mic I usually use just for a reference vocal but at the time, our old studio was ripped apart, the console was half gone, and I was short of mics anyway. The singer happened to know how to work a mic, and it was really beautiful.

Katz: I've used 87's, the Sony C37, and for a full, round, rich sound, there's nothing like the 47.

Baxter: Sometimes, I've used the Sony C-22, and if I remember right, I think that John used an old RCA 44 a few years ago for Glenys, the female vocalist with *Four Jacks and a Jill*.

Woram: I remember that very well. We went through every microphone in the place before rediscovering the 44, which turned out to be just right. However, I don't think I've used it since. Steve mentioned the Neumann 47, which was a really fine microphone. It's too bad they're not still being manufactured.

Katz: It's a terrific mic for celli and trombones.

Baxter: We also used to use it a lot for low strings and brass, but I think most of us have switched to other mics as the 47's wore out and couldn't be repaired or replaced.

Zide: Bob, you talked about measuring the various microphones a little while ago. Maybe you could say something about how you go about evaluating a mic. Since you don't do as much actual recording as the others here, how do you compare one mic with another?

Schulein: Our main source of information is the anechoic chamber. Of course, this is not a realistic environment, but it is highly controlled and constant. We know we can repeat our experiments later and come up with the same results again. So, we can subject two microphones to the identical test and then compare our results. The performance of a mic in the anechoic chamber won't tell you whether you'll like the mic in the studio, but if you do like the mic in the studio, the chamber may be able to give us a clue as to why. Also, we can make accurate measurements of off-axis response and proximity effects, and these measurements are of value as is.

I'd like to know what everyone uses on vocals.

. . . you talked about measuring the various microphones . . . . Maybe you could say something about how you go about evaluating a mic.

PROXIMITY EFFECT, AND POPPING

Katz: What do you do on your SM 53 to minimize proximity effect.

Schulein: Well, it's a little involved—there's an AES paper on the design of this mic which I'll send you.[1] Basically, it's a two-entry uniphase system, and the low frequency entry is quite far from the diaphragm, which helps minimize any proximity effects. There are times when you can use the proximity effect to advantage. Let's say you want to minimize the pick up of unwanted low frequencies from off axis sources. So, you move in close with your cardioid until you have too much bass as a result of proximity. Then you roll it off until you're back to normal. In doing so, you cut down all the unwanted low frequency noise from the surrounding area.

Another point to keep in mind when using two-entry mics: if you're using a wind filter on the front, don't forget to put some kind of filter over the rear entry ports also. The microphone works on pressure differences, and if you reduce the pressure, or wind, at one end and not at the other, it may very well sound worse than with no filter at all.

Zide: I guess we all know what a pop filter does, but I'm not so sure I know how it does it.

Schulein: Well, think about what a pop is, from a physical standpoint. It's really not so much acoustical, because when I say a word like *Peter*, you don't hear the pop that you might get if I were speaking into a microphone. So, obviously the pop signal is somewhat different from the acoustic signal. The pop is literally a puff of air being forced from mouth to mic, whereas the rest of the signal is a vibration of air molecules travelling at approximately 1000 feet per second. The pop signal is moving much faster, and the pop filter materials' resistance is non-linear. For low-velocity signals, it's quite low, but as soon as the velocity goes up, it starts to get higher. In essence, you have a selective filter that minimizes the high-velocity part of the pop signal.

Colchamiro: Is there much of a sub-sonic component in the pop? I should think there would be.

Schulein: When we were running spectrum analyses, we were going strong at 20 Hz. And if you go down to 10 Hz or so, there would be a detectable output. So, some of it is sub-sonic and could therefore cause excessive overloading. You could get into a low frequency saturation problem on your tape.

Woram: Of course, one way to minimize the pop problem is to go to an omni microphone. Or, if you're working with a cardioid, working it slightly off axis.

Schulein: Sometimes having your performer sing over the top of the mic will cut down on popping *p*'s but may get you in trouble with *t*. The *p* signal comes out pretty straight, but a *t* signal is dropped down.

Another thing we've found is that three inches seems to be a very critical distance for mic popping. Oddly enough, as you get in closer on many mics there is apt to be less trouble. I know why it gets better at *more* than three inches, but I'm not sure why the sudden improvement at *less* than three inches.

DRUM MICING

Schulein: When you fellows are micing drums, do you find yourself using a lot of mics—more so than you did in the pre-multi-track days?

Baxter: For me, it depends on how many tracks are available for the drums. If I can put the drums on two tracks, I'll use extra mics and try for some stereo effect, depending on what the drummer is doing.

Woram: I've been averaging about five mics, if I can go onto two tracks.

Zide: How do you assign them?

Woram: I usually put two EV RE-55's (which are omni's) over the drum set and about two feet apart. I'm probably violating every rule in the book, but it just seems to work out very well for me. Then I put an RE 15 close to the top of the floor tom, and a condenser mic near the snare, and then a 546 in the bass drum. The RE 15 and one of the RE 55's goes to one track and the condenser and the other RE 55 to the other, with the 546 split to both tracks. I was using a Sony C-22 for the snare, but now I'm trying the Neumann 86.

Katz: For me, the greatest snare drum mic is the 83. It's very small so I can get in really close, and since it's omni I

A GLOSSARY OF MICROPHONES MENTIONED

Model Number	Manufacturer	Transducer type	Polar Pattern	List Price
RE 15	Electro-Voice	Dynamic	Cardioid	$ 159
RE 55	Electro-Voice	Dynamic	Omnidirectional	$ 132
635a	Electro-Voice	Dynamic	Omnidirectional	$ 52.80
666	Electro-Voice	Dynamic	Cardioid	discontinued
SM 53	Shure Bros.	Dynamic	Cardioid	$ 153
SM 60=	Shure Bros.	Synamic	Omnidirectional	$ 49.20
SM 76	Shure Bros.	Dynamic	Omnidirectional	$ 111
546	Shure Bros.	Dynamic	Cardioid	$ 142
U47	Neumann	Condenser	switchable Cardioid/Omni	discontinued
KM 83	Neumann	Condenser	Omnidirectional	$ 225
KM 86	Neumann	Condenser	both switchable Cardioid/	$ 327
KM 87	Neumann	Condenser	figure-8 omnidirectional	$ 342
44 BX	RCA	Ribbon	figure-8	discontinued

still get the attack and the snare ring. Sometimes people get into problems, trying one mic *over* and one *under* the snare. You place the 83 properly and you'll pick up the whole drum.

Woram: I agree with you about the 83. From time to time, I've borrowed one, and I've never been able to beat the sound I've gotten with it.

Colchamiro: We've gotten very good results with the 546 on snare.

Baxter: What about on bass drum? That's where I use the 546.

Colchamiro: Now, I'm using the 666, but I've gone through quite an assortment in the past few years.

Katz: Most of the time I use the 666. But if I were doing something like a jazz trio, I think I'd probably go for an 87.

Schulein: I think the main difference here is response. You all end up with more or less the same response, but get it in different ways—from different mics or with different equalization.

Bradley: Dick said he liked the 546 for bass drum. Isn't he just using the proximity effect as a tool here?

Schulein: Probably. You can get a bigger low end than you would with an omni.

Baxter: I find I have good control over the low end by moving the mic in or out until I get the right balance.

PIANO PICKUPS

Katz: What about stereo micing a piano? Would you put the mics in close and at right angles to each other?

Schulein: I'd probably want to do a little experimenting. Just like the drums, the piano source is spread out. It's not like a point source that you're picking up from two different places.

Bradley: If you get in too close, there's a hole in the middle of the piano. There's got to be a good amount of leakage or it will sound false.

Schulein: It's very easy to get into trouble if you double mic a piano and put the mics on separate tracks. It sounds fine in stereo, but there can be a lot of phase cancellations when you combine later to mono. If you are going to use two mics onto separate tracks, I think Steve's idea of putting them at right angles would be the safest bet.

Woram: Lately, I've gotten away from double micing the piano. I've got one of your SM 60's and it sounds much better than anything else I've ever come across. I use it really in close. Later on if I want a more open sound, I'll use a stereo reverberation room—not a plate, but a natural room—and feed the two microphones in the room to extreme left and right, with the direct piano output in the center.

THE CONDENSER SOUND

Schulein: This is something I've been interested in for a long time. I've made comparisons between condensers and dynamics, and the responses were practically the same. Yet, to the ear there was a subtle difference. The point is, if the lab says there's no difference, and the ear says there is, then we've got to find other lab testing methods that will reveal just what these differences are.

Katz: I've read that a typical dynamic rise time might be in the neighborhood of 40 microseconds. For a condenser, it's more like 15 microseconds. That's not much of a difference, but still, 15 is better than 40, and subjectively a condenser gives me the sharp attack—especially the 83 on the drums—that I like.

Schulein: Another thing you may like about the condenser is the slight high-frequency peak that some of them have. This could contribute to the sensation of sharpness of attack.

Zide: Before we break up for the evening, I'd like to propose that some of the mixing engineers here send Bob Schulein their favorite microphone for some specific instrument. He could run some controlled tests and compile data on each of the microphones. Maybe once enough tests were made, we could see some correlation between test results and personal preferences. And then, if any conclusions can be made, we'll include them in a later issue of **db**. ■

[1] Development of a Versatile Professional Unidirectional Microphone, Robert B. Schulein, Journal of the A.E.S., February 1970, p. 44-50.

ROGER ANDERSON and

ROBERT SCHULEIN

A Distant Micing Technique

Here is a study of a microphone placement technique which some engineers have already discovered but not fully utilized.

DURING an investigation of distant microphone pickup techniques, we were comparing two tape tracks recorded from the same source, one recorded at a near distance of one foot and the other at a far distance of fifteen feet. Compared to the near recording, the distant recording had a hollow quality, somewhat like short-wave reception when the signal is fading.

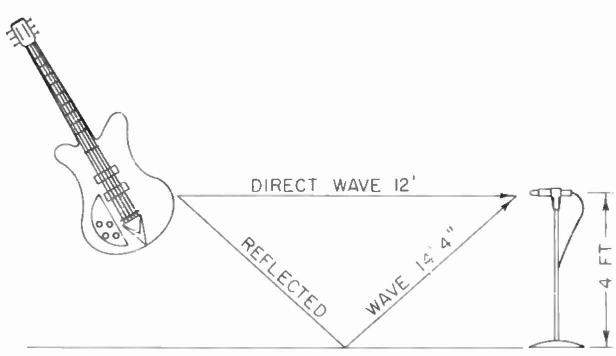
Another experiment pinpointed the cause of this effect. Here the distant microphone was moved vertically from a six-foot altitude down to the floor, keeping the source distance constant at twelve feet. Now the hollow effect varied in pitch, becoming higher until it vanished as the microphone approached the floor closely. With the microphone barely off the floor, excellent results were produced. The only difference between the near and far recordings was the greater reverberation and lower level in the far recording, as expected.

The explanation of this effect may be seen in *Figure 1*. Here a sound source (performer or musical instrument) is located four feet above the floor. The microphone is located twelve feet away on a floor stand, also four feet high. This arrangement might be used with a chorus or orchestra to maintain balance between performers and capture some natural reverberation; or in a singing/dancing routine where the stage area must remain clear.

The direct sound travels twelve feet; however a considerable amount of sound is reflected from the floor and up to the microphone again. This reflected sound travels a total of 14.4 feet, which is 2.4 feet farther than

With the microphone barely off the floor, excellent results were obtained.

Figure 1. The explanation of the quality change effect caused by moving the microphone vertically.



Roger Anderson is chief development engineer and Robert Schulein is a senior development engineer, both at Shure Brothers, Inc.

A unidirectional microphone used on the floor will retain most of its polar discrimination.

the direct sound. At a frequency of 233 Hz (wavelength 4.8 feet) the reflected sound will be 180 degrees out of phase with the direct sound, producing the phase cancellation effects associated with misaligned tape recorders or multiple microphones. This cancellation will also take place at all odd multiples of 233 Hz; *i.e.*, 699, 1165, 1631, 2097, etc. Actually, the cancellation is not complete because the reflected sound pressure is less than the direct sound pressure due to its longer travel and imperfect reflection. This is still sufficient to produce 15 dB dips in the response!

An objective experiment to document the effect was performed in an anechoic chamber. The results are shown in *Figure 2*. A one-inch condenser microphone and loudspeaker were set up four feet above the non-reflective mesh floor, separated by a distance of twelve feet. The resulting sound field experienced by the microphone is shown as curve (A). Next, a 4 x 12-foot sheet of plywood was placed on the mesh floor, between the microphone and source. Curve (B) is the result. The interference effects are quite noticeable. Curve (C) shows the result of a computer simulation of the setup, assuming 100 per cent reflection from the floor. The similarity of curves (B) and (C) show that the effect is real and predictable. The differences are due to the restricted size of the "floor" employed, and the absorption of the wood at higher frequencies.

The microphone was then lowered until it was barely clearing the floor. Curve (D) is the result, showing that the irregularities have disappeared and the level has nearly doubled. The high frequency roll-off occurs because the center of the microphone is still above the floor level.

An easy way of visualizing and explaining the situation is shown in *Figure 3*. Here, the floor has been removed and a mirror-image "virtual" source introduced which emits sound waves identical to the original source. A microphone located at A will receive the two sound waves somewhat out of step because the path lengths are not the same; consequently, interference effects will be produced. The only locations which are free of these effects lie along the perpendicular bisector of the line joining the two sources. Any point on this line will be equally distant from the two sources, and the two sounds will be exactly in phase. This line corresponds to the floor line in the real situation. Of course we cannot semi-sink the microphone into the floor, but using 1/16 or 1/8 inch clearance will insure that the lowest frequency cancellation is above 10 kHz.

To demonstrate the effect of a real, not anechoic, environment, a sound source was set up 53 inches above the floor on the stage of a high-school auditorium. Fifteen feet away, and at variable height, the condenser microphone was used to record the broadband noise fed into the loudspeaker. One-third octave analysis was later performed

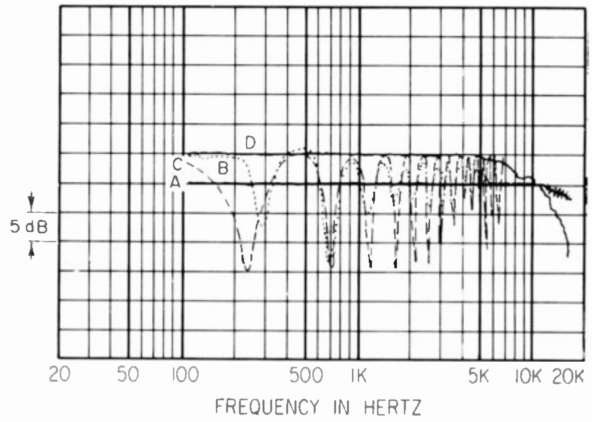


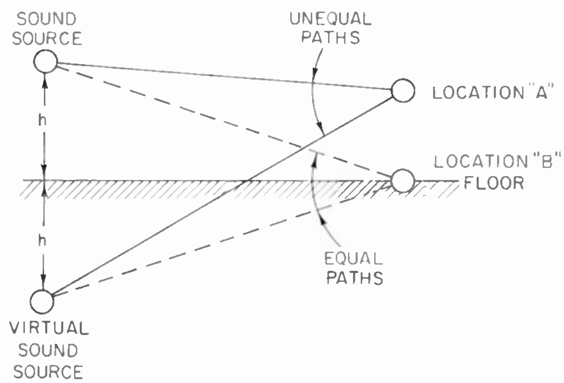
Figure 2. Anechoic chamber tests. At (A) we see the original sound field without a floor. (B) shows a 4- x 12-foot floor added. (C) is a computor simulation of (B). (D) is the microphone at floor level.

to yield the families of curves in *Figure 4*. The one-inch curve is similar to the response of the source measured in an anechoic chamber, and is the basis of comparison to the other curves. The two-inch curve shows a serious loss around 6 kHz. At twelve-inch spacing, the "hole" has moved down to 1 kHz, and some near relatives have appeared at 3 and 5 kHz. At the usual height of 53 inches, a serious dip occurs at 230 Hz, and even at 144 inches on an overhead boom some loss may be noticeable. The strong smoothing and averaging effect of 1/3-octave analysis makes the nulls less drastic than the sine-wave measurements, but they still are quite apparent.

Tests performed with unidirectional microphones have shown the same type of response-perturbation which the omnidirectionals exhibit. A unidirectional microphone used on the floor will retain most of its polar discrimination. Of course, when the microphone is close to the source, the intensity of the reflected sound is too small to have much effect, even though the path length is vastly different. In addition, the polar pattern of directional microphones will afford useful discrimination against floor reflections. If the floor is carpeted, the effects of reflection will also be reduced.

To effectively use this new position, the microphone must be very close to the floor and in a parallel orientation. The use of a desk stand places the microphone too high, or at an unfortunate angle to the floor. Overhanging the microphone head on the edge of a foam block is

Figure 3. A visualization of the effects described.



To effectively use this new position, the microphone must be very close to the floor and in a parallel orientation.

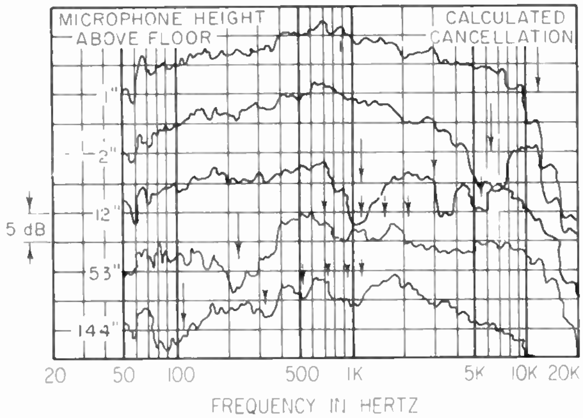


Figure 4. On-location tests. The curves have been displaced on this graph for clarity.

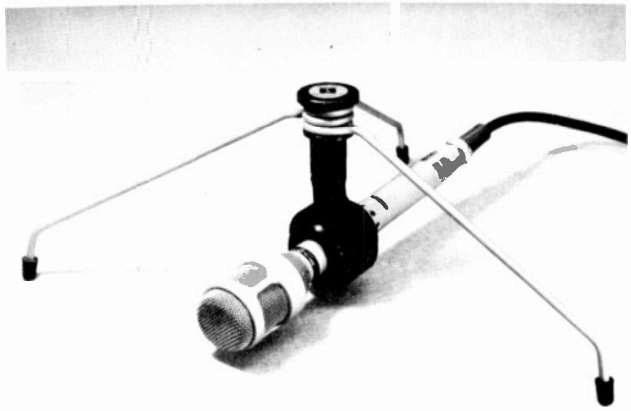


Figure 5. The mic stand that has been devised to take full advantage of the conditions described. It is commercially available from Shure.

acoustically usable, but mechanically unstable.

A practical solution to the problem is shown in *Figure 5*. This new stand has been designed to support the microphone properly and securely. In addition, it affords excellent shock isolation from floor vibrations. It is available in two models, the S53P and S55P, to fit 0.790 or one-inch diameter microphones respectively, and folds flat for storage.

Once the possibilities of reflections are recognized, other applications come to mind. For instance, a recording made at a desk from one or two feet away with the usual desk microphone stand will show the same type of interference effects noticed at greater distances. Similarly, when recording in auditoriums, putting microphones next to the side walls may be desirable. Many other examples will be apparent if the principle is kept in mind.

Our experiments have led to this general rule: *When the microphone-to-source distance becomes greater than one or two times the distance from the source to the reflecting surface, it is desirable to place the microphone next to the reflector.*

As a bonus, the sound level will be 6-dB higher than if the reflecting surface was not present. ■

JOHN EARGLE

How Capacitor Mics Produce Cardioid Patterns

In the studio, capacitor microphones are almost always used in their cardioid or (directional) pick-up pattern, but few engineers understand how this is accomplished. The purpose of this article is to show how this is done—with a minimum of mathematics and an abundance of graphics. But first we must touch upon a couple of basics. Just what *is* a cardioid pattern, and just how does the basic capacitor element function as an electro-acoustical transducer?

Figure 1 shows a capacitor microphone in its simplest form. The capacitor, made up of a fixed back plate and a moveable diaphragm, is in series with a battery and a resistor. The battery charges the capacitor through the resistance, and the steady-state condition which is readied after charging is given by the equation

$$Q = CE$$

Where Q is the charge on the capacitor in coulombs, C is the capacitance in farads, and E is the voltage of the battery.

When placed in a sound field, the diaphragm moves and the capacitance varies. This in turn tends to vary Q, but the high value of R effectively prevents the charge from leaving the capacitor. Thus there is a change in the voltage across C which is given by

$$\Delta E \cong \Delta C$$

We have made a few simplifications here, but the explanation shows essentially what happens.

A cardioid or heart-shaped, pattern is the sum of two simpler ones. *Figure 2* shows how a figure-8, (or cosine) pattern combines with a constant, or omnidirectional, pattern to yield the familiar cardioid. It's a pure and simple geometrical construction shown in polar form, where O represents the angle of incidence to the microphone. Obviously, if we can make a set of capacitor elements exhibit *both* figure-8 and omnidirectional response at the same time, then by linear combination the output will be a cardioid. We are just about to show how this is done, but first we must relate the familiar elements of electrical circuits with their mechanical analogs (*Table 1*).

John Eargle is chief engineer of Mercury Sound Production

TABLE 1

Electrical Quantity	Mechanical Quantity
	(Impedance Analogy)
Voltage (E)	Force (F)
Current (1)	Velocity (V)
Resistance:	Damping (dash pot)
Capacitance:	Compliance:
Inductance:	Mass:

A simple example of an electro-mechanical impedance analogy is shown in *Figure 3*. Here, the series resonant electrical circuit is likened to a weight bobbing up and down on a spring (the dash pot represents the normal viscosity of air acting on the spring and weight tending to damp out their oscillation). Note that at resonance a low force results in a high velocity in the mechanical circuit, while a low voltage results in a high current in the electrical circuit. Stated differently, both systems exhibit low impedance at resonance.

Both systems have the same resonance curve, as shown in *Figure 4*, for a constant applied force or voltage. Note that this curve has *three* regions of interest. Below resonance the response falls off at 6 dB-per-octave. Above

Figure 1. How a capacitor mic converts sound pressure to an electrical signal. The capacitor is charge by a voltage source through a resistor: Q (charge) = CE. Sound pressure varies C which tends to vary Q. However, the high value of R prevents the charge from leaving the capacitor. Thus there is a change in voltage on C which is ΔE = Q/ΔC. This voltage then appears as a signal across R.

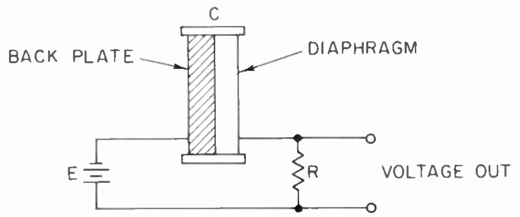


Figure 2. The construction of a cardioid from its two components. Thus, if we can make a mic element behave as an omni and a Figure-8 simultaneously, its output will have the directional characteristics of a cardioid.

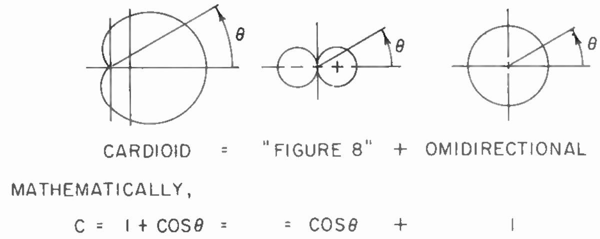
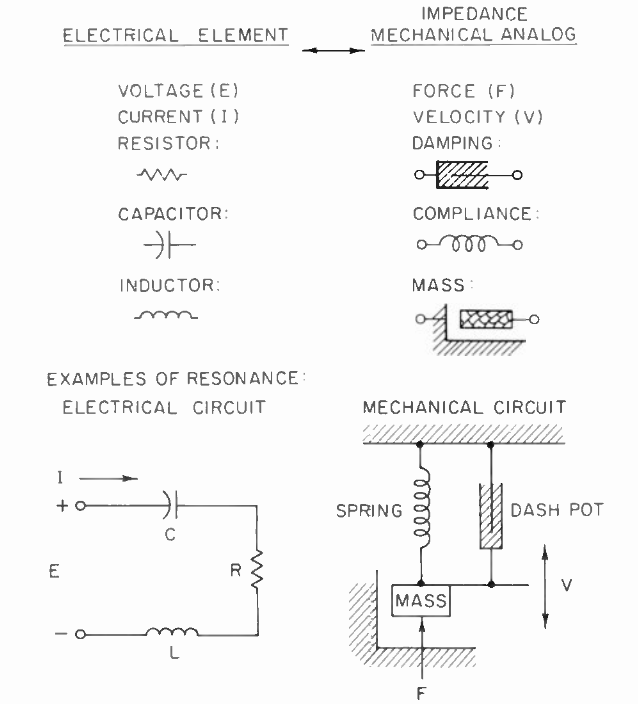


Figure 3. Examples of analagous electrical and mechanical circuits.

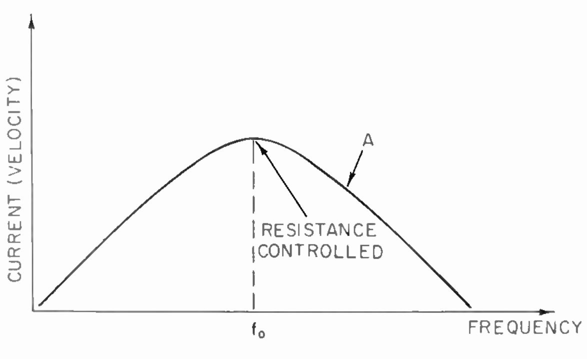


resonance the same thing occurs, while in the region of resonance, depending on how much damping there is, the response is fairly flat. There is another description of this; above resonance the systems are said to be *mass* or *inductance* controlled. That is, the reactance of these elements is far greater than that of the other elements. Below resonance the systems are said to be *compliance* or *capacitance* controlled. In the region of resonance (where the reactances largely cancel) the systems are said to be *resistance* controlled if there is enough damping or resistance present in the systems to flatten out the curve.

We are now ready to take a look at the compound diaphragm system which makes us a modern capacitor microphone (see *Figure 5*). Note the following characteristics of this arrangement:

1. There are two diaphragms on either side of the back plate.
2. The back plate is *perforated* by many fine holes.
3. There are additional holes on each side of the back plate (which do not go all the way through) which provide extra viscous damping for the diaphragms.

Figure 4. A resonance curve for constant voltage or force input for the circuits of Figure 3. The arrow at A points to the fact that above resonance, the system is mass or inductance controlled. However, below resonance the system is compliance or capacitor controlled. Both curves roll off at 6-dB per octave.



4. The two diaphragms and the back plate constitute *three* electrodes.

In the standard omnidirectional back-up, the three electrodes are powered as shown in *Figure 6(A)*. Here, the back plate is at ground potential, while both diaphragms are biased positively. Under this condition the microphone responds only to pressure variations as shown in *Figure 6(B)*, and this yields omnidirectional response. Furthermore, the system is operating well below resonance and there is a 6-dB/octave rise in diaphragm velocity with frequency as shown in *Figure 6(C)*. If the *velocity* of the

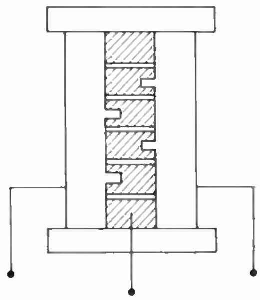
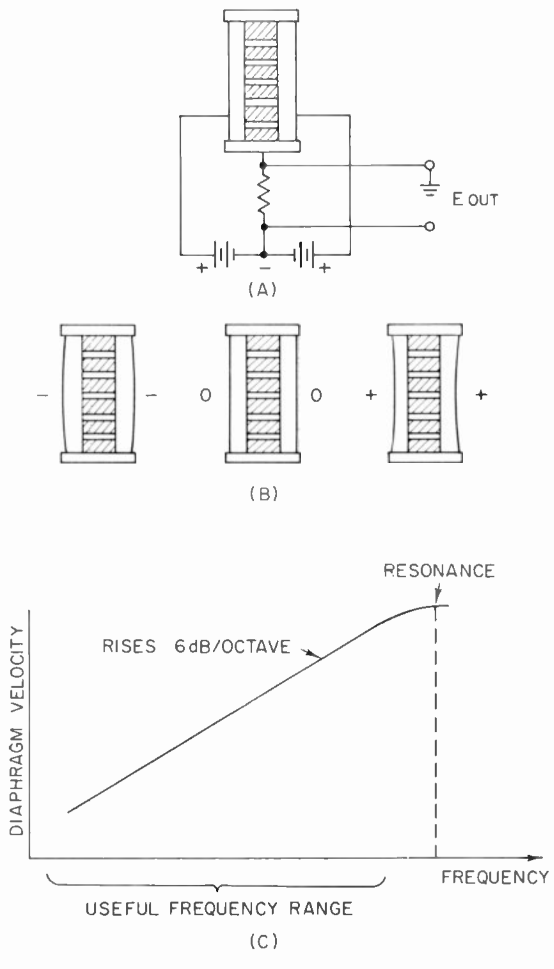


Figure 5. A standard capacitor mic (a U-67) looks like this. Notice that there are two diaphragms, one on each side; the back plate is perforated by many fine holes; there are additional holes provided in the back plate for purposes of damping; and the back plate and two diaphragms constitute three electrodes.

Figure 6. Here is the compound capacitor mic in its electrical connection for omnidirectional response. At (A) the electrical connection; at (B) this electrical connection is sensitive only to these motions of the diaphragms; at (C) diaphragms are compliance controlled.



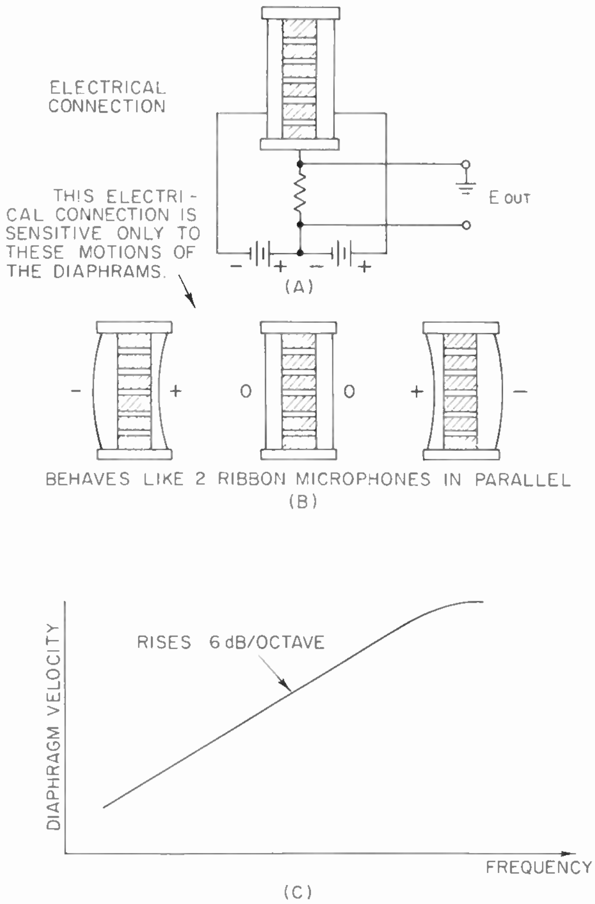


Figure 7. The same mic as in Figure 6 only connected in standard figure-8. At (A) electrical connection; at (B) this electrical connection is sensitive only to these motions of the diaphragms; at (C) diaphragms are resistance controlled in this mode, however, pressure difference between the two sides of the capsule rises with frequency.

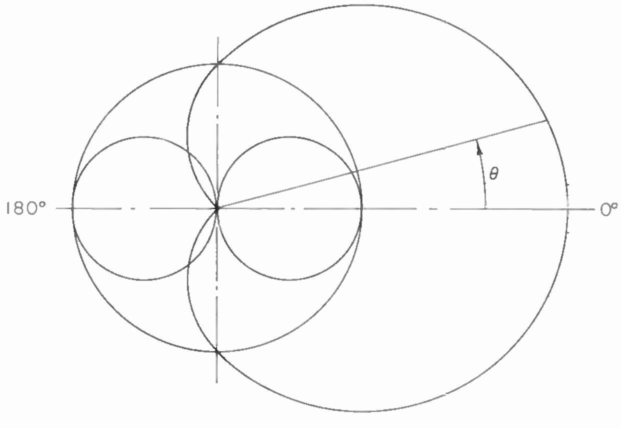
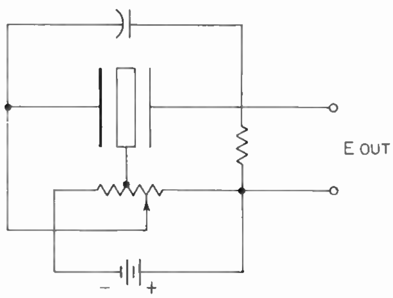


Figure 8. A summation of the two patterns of Figures 6 and 7.

Figure 9. A schematic of the variable-pattern Neumann M-49 microphone.



diaphragm rises with frequency at 6-dB/octave, then the *amplitude* of the diaphragm displacement is constant with frequency. This is an elementary fact of calculus, and further explanation is beyond the scope of this article. What is important is to observe that the flat amplitude response in this mode of operation results in a flat electrical output over the operating range.

By a simple change in the polarity of one of the voltage sources in the compound capacitor array, we get a figure-8 response out of the array. As shown in *Figure 7(A),(B)*, this array produces an output voltage *only* when the two diaphragams are moving in parallel. This motion resembles the action of a ribbon microphone; at right angles to the capacitor array, the two motions *cancel* electrically, while pressure variations perpendicular to the diaphragms produce a maximum voltage output. In this connection, the array is acting as a pressure gradient microphone; that is, it responds to the *difference* between the instantaneous pressures existing on each side of the capacitor array. Because of the finite acoustical path length between the two sides of the array, there will *always* be a pressure gradient or difference between the two sides, and this pressure difference is a direct function of frequency as shown in Figure 7(C). Because of the many fine holes in the back plate there is a high degree of damping on the motion of the diaphragms in this pressure gradient mode. The mechanical circuit is operating in its *resistance controlled* mode, and it is this, in combination with the frequency dependent pressure gradient, which results in a response which rises 6-dB/octave with frequency.

We have described two distinct motions of the compound capacitor array, one producing omnidirectional response and the other producing figure-8 response. Both of the motions are linear, and they can be added geometrically as shown in *Figure 8*. Since these two motions coexist in the compound capacitor array, we can combine them in any degree by simply providing a center-tapped potentiometer in the biasing network, as shown in *Figure 9*. This is a schematic diagram of the variable-pattern Neumann M-49. When the swinger of the potentiometer is at the positive end of the voltage source, the response is omnidirectional. When the swinger is at the center tap the response is cardioid, and when it is at the negative end the response is figure-8. There is a gradual transition from one pattern to another as the swinger moves from positive to negative.

I have described the fundamental workings of the compound capacitor array in modern capacitor microphones. Actual design work however goes far beyond these fundamentals. The effects of diffraction, material choice, and dimensions would all play a large part in determining the sound of a microphone. Their choice would in fact determine how well the principles I have outlined would work. ■

REFERENCE:

New High-Grade Condenser Microphones F.W.O. Bauch, Journal of the AES July 1953 Volume 1, Number 3.

40th AES Convention and Exhibition

SCHEDULE OF EVENTS

LOS ANGELES HILTON HOTEL

Monday, April 26, 5:00-7:00 P.M.—Welcoming Cocktail Party

REGISTRATION

Monday,	April 26—1:00 to 5:00 P.M. EXHIBITORS ONLY
Tuesday,	April 27—8:00 A.M. to 8:00 P.M.
Wednesday,	April 28—8:30 A.M. to 8:30 P.M.
Thursday,	April 29—9:00 A.M. to 5:00 P.M.
Friday,	April 30—9:00 A.M. to 5:00 P.M.

For Recording Studio Workshop Registration will be at the door.

EXHIBIT HOURS

Tuesday and Wednesday,
April 27 and 28— 1:00 P.M. to 9:00 P.M.
Thursday and Friday,
April 29 and 30—11:00 A.M. to 5:00 P.M.
PACIFIC, WILSHIRE, GARDEN AND SIERRA ROOMS

DEMONSTRATION ROOMS

Mission, Cleveland, Washington, Detroit, Boston, Buffalo St. Louis, Foy, New York, Dallas, Hartford

TECHNICAL SESSIONS

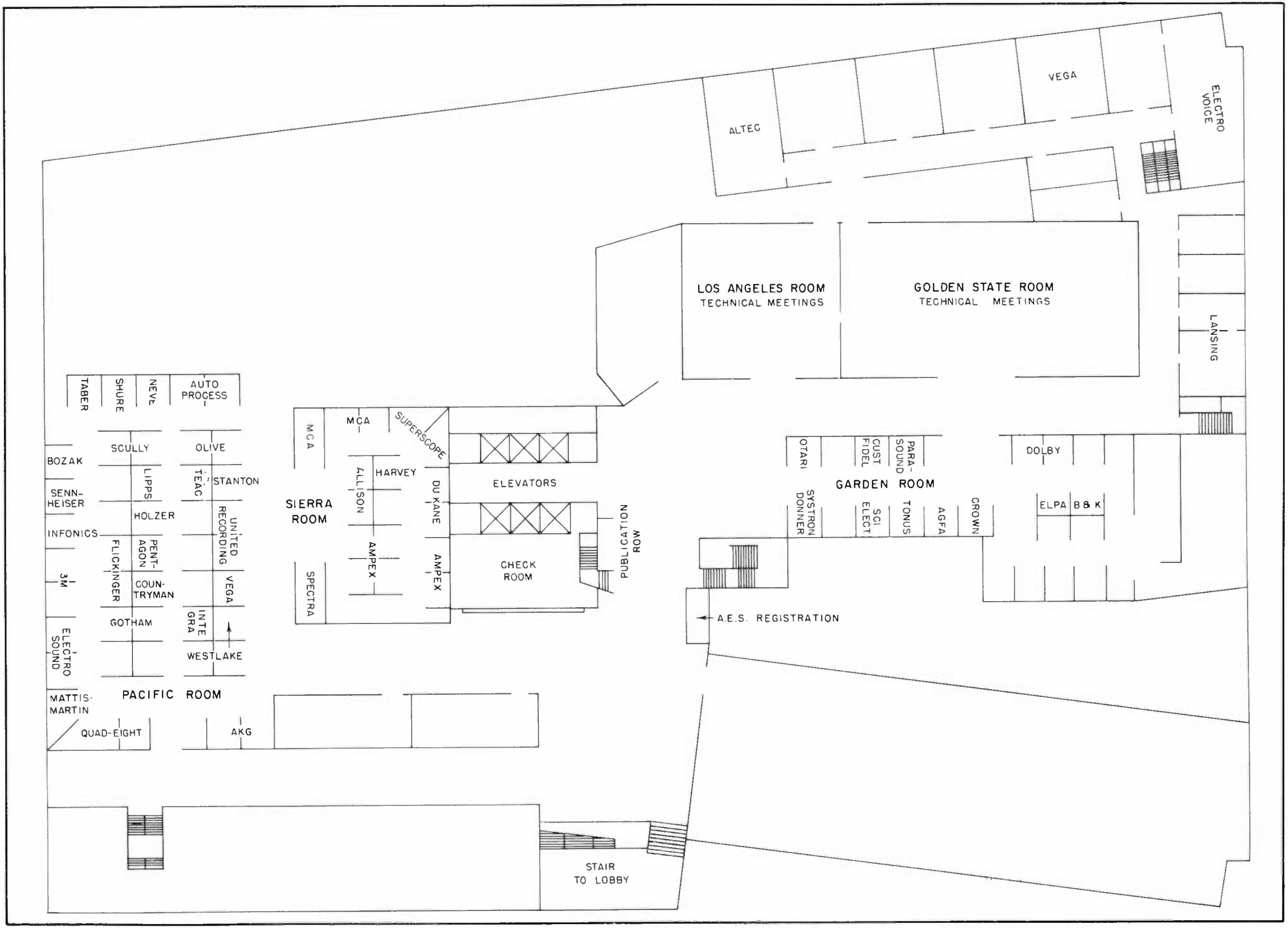
Tuesday,	April 27—	9:30 A.M.	A & B
		2:00 P.M.	C
		7:30 P.M.	D
Wednesday,	April 28—	9:30 A.M.	E
		2:00 P.M.	F & G
	No Sessions Wednesday evening		
Thursday,	April 29—	9:30 A.M.	H
		2:00 P.M.	J & K
Social Hour		—7:00 P.M.	Los Angeles Room
Awards Banquet		—8:00 P.M.	Golden State Room
Friday,	April 30—	9:30 A.M.	L
		2:00 P.M.	M
		7:30 P.M.	N

Note: Session N—Recording Studio Workshop
No Registration Fee

LADIES ACTIVITIES

A program of activities is being planned. Ladies may join the hostess and her committee at 9:00 A.M. each day for coffee and sweet rolls before commencing the day's activities. Suite number will be posted.

Ladies Committee: Mrs. William Brandt
Mrs. Hugh S. Allen, Jr.


ALTEC
VEGA
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GOLDEN STATE ROOM
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LANSING
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NEVE
AUTO PROCESS
SCULLY
OLIVE
BOZAK
LIPPS
TEAC
STANTON
SENN-HEISER
HOLZER
INFONICS
FLICKINGER
PENT-AGON
UNITED RECORDING
3M
COUN-TRYMAN
VEGA
GOTHAM
INTE GRA
ELECTRO SOUND
WESTLAKE
MATTIS-MARTIN
PACIFIC ROOM
QUAD-EIGHT
AKG
MCA
MCA
SUPERSCOPE
HARVEY
ALLISON
SIERRA ROOM
DU KANE
AMPEX
AMPEX
SPECTRA
ELEVATORS
CHECK ROOM
PUBLICATION ROW
OTARI
CUST FIDEL
PARA-SOUND
DOLBY
GARDEN ROOM
SYSTRON DONNER
SCI ELECT
TONUS
AGFA
CROWN
ELPA
B & K
A.E.S. REGISTRATION
STAIR TO LOBBY

SESSION A

Tuesday, April 27, 1971, 9:30 A.M.
Golden State Room

MAGNETIC RECORDING AND REPRODUCTION

Chairman: KEITH JOHNSON,
MCA Technology,
North Hollywood, California

A High Energy Cassette Tape With Compatible Magnetic Properties—D. A. Eilers and E. W. Reed, 3M Company, Magnetic Products Division, St. Paul, Minnesota

A Servo Controlled Recorder For Studio Applications—Harold T. Schneider, Philips Broadcast Equipment Corp., Montvale, New Jersey

Chromium Dioxide Audio Tape—Klaus E. Naumann, Memorex Corporation, Santa Clara, California

The Fringing Response Of Magnetic Reproducers At Long Wavelengths—J. G. McKnight, Ampex Corporation, Redwood City, California

Musicassette Quadrasonic: Tape Record Compatibility—E. R. Hanson, North American Philips Corp., New York, New York

SESSION B

Tuesday, April 27, 1971, 9:30 A.M.
Los Angeles Room

AUDIO MEASUREMENTS AND NOISE CONTROL

Chairman: CHARLES HORTON,
Altec, Anaheim, California

The Application of Impulse Measurement Techniques To The Detection of Linear Distortion—Alfred Schaumberger, Georg Neumann GmbH Electroacustic, W. Berlin, Germany (translated and presented by Stephen F. Temmer.

30 Band 1/3 Octave Spectrum Analyzer—Daniel N. Flickinger, Daniel N. Flickinger & Associates, Inc., Hudson, Ohio

A 1/3 Octave Real Time Analyzer Using Calibrated Meter Readout—Victor M. Hall and J. Earl Chapman, Communications Company, Inc., San Diego, California

Instant RT60—Victor M. Hall and J. Earl Chapman, Communications Company, Inc., San Diego, California

Airport Noise Management—John K. Hilliard, Ramberg & Lowrey, Architects (Acoustics and Noise Control Division), Santa Ana, California

Application of Acoustically Terminated Tube for The Measurement of Horn-Loudspeaker-Driver Characteristics and Comparison of Distortion Measurement Methods—Bart N. Locanthi, Ludwig W. Sepmeyer, Independent Consultants, Los Angeles, California

SESSION C

Tuesday, April 27, 1971, 2:00 P.M.
Golden State Room

DISC RECORDING AND REPRODUCTION

Chairman: STEVEN A. GUY
Location Recording Service,
Burbank, California

A New Dynamic Feedback Stereo Cutter-Head With Associated Solid State Driving System—Howard S. Holzer, Holzer Audio Engineering Company, Los Angeles, California

Further Improvements In Performance of the Westrex 3D-II Stereodisk System—Frank E. Pontius, Westrex, Beverly Hills, California, and John P. Jarvis, Consulting Engineer, Northridge, California

Groove Echo In Lacquer Masters—Daniel W. Gravereaux and Benjamin B. Bauer, CBS Laboratories, Stamford, Connecticut

Development of Skew-Sampling Compensator for Tracing Error—Shigetaka Washizawa, Tomofumi Nakatani and Takeo Shiga, Nippon Columbia Co., Ltd., Kawasaki, Kanagawa, Japan

Analysis of Crosstalk in Stereo Discs—Bernhard W. Jakobs, Shure Brothers, Inc., Evanston, Illinois

The Education and Tribulations of a Precursory Disc Recording Engineer—Robert Callen, Glen Glenn Sound Company, Los Angeles, California

A Console Approach to Quad-Sound Disc Mastering—Michael S. Levey, The Custom Fidelity Company, Inc., Hollywood, California

SESSION D

Tuesday, April 27, 1971, 7:30 P.M.
Golden State Room

STUDIO RECORDING TECHNIQUES TODAY

Chairman: HUGH P. STARK,
Elektra Records, Los Angeles, California

On The Processing of Two and Three-Channel Program Material for Four-Channel Playbacks—John Eargle, Mercury Record Productions, Inc., New York, New York

Studio Recording Techniques of a Small Recording Studio—Philip Kaye, ABC Dunhill Records, Los Angeles, California

Dual-Triphonic Matrix Stereo System—Takeo Shiga, Michio Okamoto, Nippon Columbia Co., Ltd. and Duane H. Cooper, University of Illinois, Urbana, Illinois

Two Ears, One Mind, and the Stereo System—David Thuesen, Poppi Recording Studios, Hollywood, California

Design Considerations for a New Studio Complex—John Mosely, Command Studios, London, England

A Stereo-Quadraphonic System—B. B. Bauer, Daniel Gravereaux and Arthur J. Gust, CBS Laboratories, Inc., Stamford, Connecticut

On The Acoustics of Multi-Track Recording Studios—Michael Rettinger, Consultant in Acoustics, Encino, California

SESSION E

Wednesday, April 28, 1971, 9:30 A.M.
Golden State Room

TRANSDUCERS

Chairman: AUSTIN J. BROUNS,
Advanced Technology Center, Inc.,
Dallas, Texas

A High Quality All Horn-Type Transducer—Raymond Newman, Electro-Voice, Inc., Buchanan, Michigan

Improved Measurement of Loudspeaker Parameters—J. Robert Ashley and M. D. Swan, University of Colorado, Colorado Springs, Colorado

A Mobility Analysis of the Closed Box and Reflex Loudspeaker Enclosures—Wayne M. Schott, Zenith Radio Corporation, Chicago, Illinois

Transducers and Industrial Espionage—Leo Jones, Saber Laboratories, Inc., San Francisco, California

Gradient Loudspeaker for Low Frequencies—W. L. Hayes, Altec, Anaheim, California

SESSION F

Wednesday, April 28, 1971, 2:00 P.M.
Golden State Room

SOUND REINFORCEMENT AND ARCHITECTURAL ACOUSTICS

Chairman: CHARLES A. STANDIFORD,
Altec, Anaheim, California

The World's Most Powerful Sound System—Robert E. Reim, Hannon Engineering, Inc., Los Angeles, California

Blossom Music Center—Daniel N. Flickinger, Daniel N. Flickinger and Associates, Hudson, Ohio

The Alteration of the Reverberation Times in a Small Theater and a Concert Hall Using Loudspeaker Equipment—Ernst-Joachim Voelker, Radio and TV Hessischer Rundfunk, Frankfurt, Germany

A Complex Sound System Equalization—G. R. Thurmond, McCandless Consultants, Inc., Austin, Texas

Sound Reinforcement Systems for the Modern High School and College Gymnasium Complex—Albert A. Huff, Hannon Engineering, Inc., Los Angeles, California

Acoustical Design of Poppi Studios—Ronald L. McKay, Bolt, Beranek and Newman, Inc., Canoga Park, California

SESSION G

Wednesday, April 28, 1971, 2:00 P.M.
Los Angeles Room

AUDIO IN AM, FM AND TV BROADCASTING

Chairman: DONALD C. McCROSKEY,
American Broadcasting Co.,
Hollywood, California

The Dorren Compatible Four-Channel FM Broadcast System—James Gabert, K101, San Francisco, California

A Tape Cartridge Recorder System Employing Integrated Circuit Logic and DC Servo Motor Drive—Ron DeBry, Garon Electronics (A Division of Visual Electronics), Sunnyvale, California

A Sound Augmentation System—Donald C. McCroskey, American Broadcasting Company, Hollywood, California

The Measurement and Control of Loudness Levels of Broadcast Sounds—E. L. Torick, R. G. Allen, P. Milner, B. B. Bauer, CBS Laboratories, Stamford, Connecticut

Panel Discussion: The Control of Loudness in Broadcasting—

Moderator: **Donald C. McCroskey**, American Broadcasting Company, Hollywood, California

Panel: **Kenneth Erhardt**, National Broadcasting Company, Los Angeles, California
Wallace Kabrick, Gates Radio Company, Quincy, Illinois
Joseph D. Kelly, Glen Glenn Sound, Hollywood, California
Emil L. Torick, CBS Laboratories, Stamford, Connecticut

SESSION H

Thursday, April 29, 1971, 9:30 A.M.
Golden State Room

SIGNAL CONTROL—SYSTEMS

Chairman: SHELLEY HERMAN
Allison Research, Inc.,
Hollywood, California

A Different Approach to Multi-Channel Home Recording Systems—John Mosely, Command Studios, London, England and Lou Lindauer, Automated Processes, Farmingdale, New York

A New Disc Mastering Console Designed for Flexibility—Robert M. MacLeod, Jr., Artisan Sound Recorders, Hollywood, California

Double Sound System—Stan Horobin, Supervisor, Audio Operations, Canadian Broadcasting Corporation, Toronto, Canada

Sound Effect Systems, Simple and Complex—David L. Klepper and Vincent Piacentini, Bolt, Beranek and Newman, Inc., New York, New York

A Functional Review of the New Automated 16 Track Recording Console at Capitol Records Studio A Hollywood—Deane E. Jensen, Quad Eight Sound Corporation, North Hollywood, California

Portable Mic-Mixdown Console Kit—B. J. Losmandy, Opamp Labs, Inc., Los Angeles, California

SESSION J

Thursday, April 29, 1971, 2:00 P.M.
Golden State Room

ELECTRONIC MUSIC

Chairman: PAUL BEAVER
Parasound, Inc.,
Los Angeles, California

The Electrical Design and Musical Applications of an Unconditionally Stable Combination Filter/Resonator—Dennis Colin, Tonus, Inc., Newton Highlands, Massachusetts

Synthesis of Moving Sound Sources—Robert B. Easton, Parasound, Inc., Los Angeles, California

SESSION K

Thursday, April 29, 1971, 2:00 P.M.
Los Angeles Room

AUDIO AND MEDICINE

Chairman: DAVID ANNETT
Stanford University,
Stanford Medical Center,
Stanford, California

The Origin and Power Spectrum of Fetal Heart Sounds—Dr. Louis Bartolucci, San Francisco, California

A Doppler Ultrasonic Method for Monitoring Fetal Cardiac Activity—Paul R. Goldberg, Project Manager, Ultrasound Instrumentation for Smith Kline Instruments, Palo Alto, California

A New Approach for Testing the Hearing of the Newborn—Clinton O. Jorgensen, Beckman Instruments, Inc., Fullerton, California

Spectral Analysis of Vascular Murmurs—E. G. Tickner and A. H. Sacks, Palo Alto Medical Research Foundation, Palo Alto, California

Characteristics of Acoustical Holography as Applied to Medicine—Byron B. Brenden, Holosonics, Inc., Palo Alto, California

SESSION L

Friday, April 30, 1971, 9:30 A.M.
Los Angeles Room

AUDIO INSTRUMENTATION

Chairman: BOB BEAVERS
Altec, Anaheim, California

Measurement of Microphone Characteristics—David G. Arnold, Shure Brothers, Inc., Evanston, Illinois

Wideband Microphone Calibrator—Ronald Brown, Advanced Technology Center, Grand Prairie, Texas

Low Power Drain Instrument Preamplifier—Robert F. Downs, OAS/Western, Ocean and Atmospheric Science, Inc., Santa Ana, California

Determination of Loudspeaker Signal Arrival Times—Richard C. Heyser, California Institute of Technology Jet Propulsion Laboratory, Pasadena, California

Group and Phase Velocity Requirements for Audio Systems—J. R. Ashley and T. A. Saponas, University of Colorado, Colorado Springs, Colorado

Oscilloscope Adaptor Presents Twenty-Four Simultaneously Different Voltages or Events for Comparison—J. Earl Chapman and Victor M. Hall, Communications Company, Inc., San Diego, California

SESSION M

Friday, April 30, 1971, 2:00 P.M.
Los Angeles Room

SIGNAL CONTROL—CIRCUITRY

Chairman: ROBERT A. BUSHNELL
Bushnell Electronics Corp.,
Van Nuys, California

A Variable Decay Reverberation System—Johan Van-Leer and John Windt, Quad-Eight Sound Corporation, North Hollywood, California

The Foster Freqy—A New Tool in Audio—Don Foster, Inventronics—Division of Amos Productions, Canoga Park, California

Electromechanical Line Transducer—G. Kirby Miller, GTE Sylvania, Inc., Mountain View, California

An Audio Delay System Using Digital Technology—Barry Blesser and Francis F. Lee, Massachusetts Institute of Technology, Cambridge, Massachusetts

SESSION N

Friday, April 30, 1971, 7:30 P.M.
Golden State Room

A RECORDING STUDIO WORKSHOP

Chairman: WILLIAM L. ROBINSON
Sunset Sound Recorders,
Hollywood, California

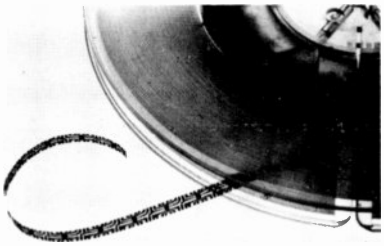
A Recording Studio Workshop—Bill Lazerus, Recording Workshop Participant, Senior Mixer, Sunset Sound Recorders; Brian Ingoldsby, MCA Recording Studios, Recording Studio Participant; another to be announced later.

Three of the top recording engineers will present a live recording session, a mix-down session from sixteen track. A detailed explanation will be given for the use of signal processing equipment, microphone techniques, and the use of specialized equipment. A question and answer period will follow each phase of the workshop.

PEOPLE, PLACES, HAPPENINGS

Friends of Rupert Neve of England will be interested to learn that they have formed a new company in the United States—*Rupert Neve Incorporated*. Their facilities are located within easy road or rail travel of New York City and they look forward to welcoming many old and new friends to their premises. A new Canadian Sales and Service Company has also been formed and is located in Toronto, Ontario. Their highly qualified professional design and audio engineers, now this side of the Atlantic, are ready to offer even more efficient service to their customers. For information contact Dave Neve, Rupert Neve Incorporated, Berkshire Industrial Park, Bethel, Connecticut 06801. Phone 744-6230, or Rupert Neve of Canada Limited, P.O. Box 182, Etobicoke, Ontario, Canada.

Ultra Sonic Recording Studios had an open house for the industry at their new Hempstead, N.Y. studios on February 26th. It was a gala affair attended by most of the recording and allied people in the area. The new studios are very impressive and include a number of firsts in the Long Island area as well as in the industry. **db** will have a story on this new facility shortly. One interesting note: the premises of the studio was formerly the main office of a large stock brokerage firm. Maybe it is trying to tell us something?



A look into the future of i.c. construction is provided by an announcement that **Philips Research Laboratories** in Eindhoven has a new process of i.c. mounting that is believed highly suited for automation. Contact patterns are first made on an inexpensive flexible plastic tape. Then flip chips are mounted on this tape and a final measuring station discards the poor chips. Finally a header with pins adapted to the product converts the i.c.'s to rugged components for use on conventional printed circuit mounting. Technical details are soon to be published by Philips.

An inexpensive method of background noise reduction said to considerably improve f.m. reception has been announced by Dolby Laboratories. Recent experiments in London and Chicago have estimated that the improvement is equivalent to that obtained by an increase in transmitter power of 5–20 times, while actual area coverage of a station can be more than doubled by use of the system. Implications for broadcasters and listeners are of extreme importance.

Derived from the Dolby B System, the technique is already in use for home tape recording and playback of commercially-recorded cassette tapes. At present, about a dozen companies in the United States, Europe, and Japan are licensed to manufacture products incorporating the system with many more currently making arrangements for licensing.

CLASSIFIED

FOR SALE

MODULES FOR ALL TYPES of applications. Discrete to thick film IC devices, Preamplifiers to power amplifiers. New Catalog with schematics. $1.00. **PM Electronics, Inc., Box 46204, Seattle, Washington 98146.**

AMPEX AG600 half track mono deck in custom Formica cabinet. Still in factory warrantee—$500. **L.A. Krause, 1324 N. Hwy 360 #249, Grand Prairie, Texas 75050.**

WHATEVER YOUR EQUIPMENT NEEDS—new or used—check us first. Trade your used equipment for new. Write for our complete listings. **Broadcast Equipment & Supply Co., Box 3141, Bristol, Tenn. 37620.**

COMPLETE TAPE DUPLICATING SYSTEM. Convertible 8-track/4-track. One inch master and bin with ten 1/4-in. slaves. Reliable, trouble-free operation. Some support equipment available. Greatly reduced price. **P.O. Box 65856 Los Angeles, Calif. 90065 Stn. 967**

WESTREX/SCULLY STEREO DISC MASTERING system complete including Scully variable pitch automated lathe, Westrex 3D system, two Pultec EQP-1A equalizers, two Pultec HLF-3C filters, Ampex 300-2 recorder with preview assembly, Fairchild Conax, Gates Limiter, McIntosh MC-275 monitor amp., JBL speaker systems (two), plus other miscellaneous cutting equipment mounted in four Bud Rack units. System is completely wired and ready to cut. Call or write for more information. **Frankford/Wayne, 212 N. 12th Street, Philadelphia, Pa 19107. Mr. Steele. (215) 561-1794.**

FOR SALE: NATIONALLY KNOWN RECORDING STUDIO. Eight-track operation (can be used for sixteen track). Located in Music City, U.S.A. (Nashville). Five going labels, two publishing companies. $100,000 gross annual custom business. Reason for sale: owner desires to devote full time to electronic manufacturing. Write **Box 5-A, db Magazine, 980 Old Country Road, Plainview, N.Y. 11803.**

AMPEX, ALTEC, McINTOSH, etc. Equipment for four track studio complete. Steal it!. First $4950. Investigate now. Evenings, **Stan Johnson, (319) 323-2873.**

SOLID-STATE AUDIO PLUG-IN OCTAL (1" Dia x 2" H) modules. Mic preamps, disc & tape preamp-equalizers, tape bias osc. & record ampl., power amps & power supplies. Send for free catalog and audio applications. **Opamp Labs., 172 So. Alta Vista Blvd., Los Angeles, California 90036.**

AMERICA'S LARGEST SELECTION of new and used broadcast and recording equipment! Latest bulletins available. **The Maze Corporation, P.O. Box 6636, Birmingham, Ala. 35210.**

WANTED

WANTED—DISC MASTERING—synthesizer recordings. **International Recording Company, 49 Desmond Avenue, Bronxville, N.Y. 10708. (914) DE 7-5726.**

EMPLOYMENT

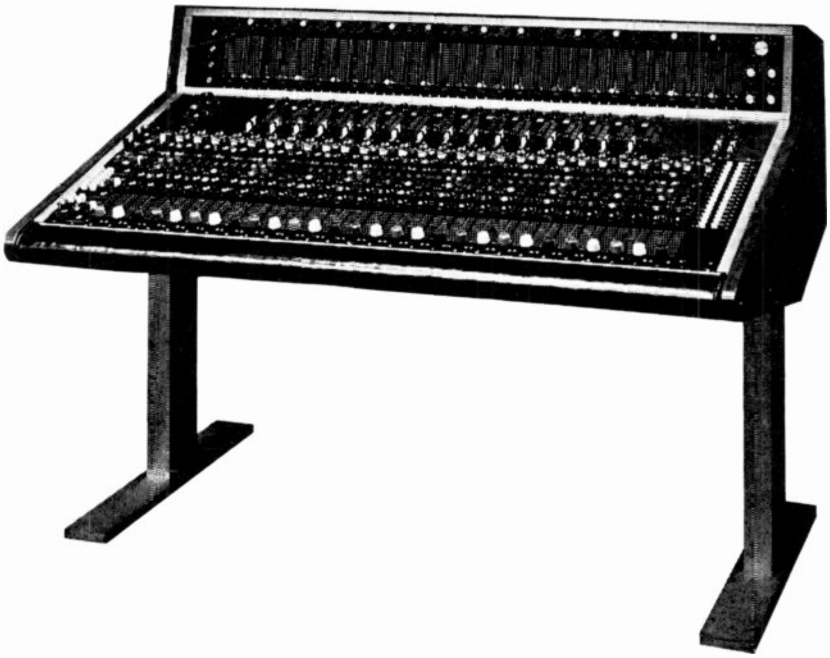
PROFESSIONAL RECORDING PERSONNEL SPECIALISTS. A service for employers and job seekers. Call today! **Smith's Personnel Service, 1457 Broadway, N.Y.C. 10036. Alayne Spertell 212 WI 7-3806.**

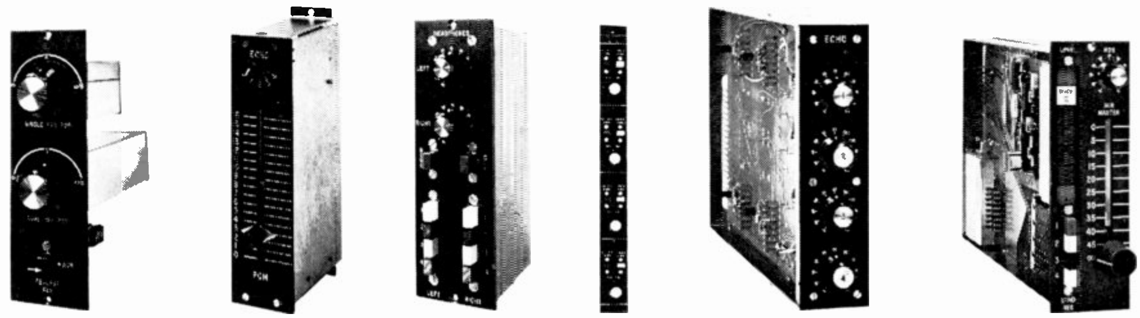
ENGINEER/CASSETTE SPECIALIST. Unusual opportunity for take-charge type of individual experienced in mechanics and electronics of audio cassettes to head new program for AAA1 rated prime tape producer. Send resume (including salary history and requirements) to **Robins Industries Corp., Attn: W. Meuselbach, 15-58 127th Street, College Point, N.Y. 11356.**

PROFESSIONAL SOUND SYSTEMS ENGINEER. Must be knowledgeable in communications audio field, major installations, and systems equalization. Contact **Bernard Johnston, DuKane Corp., St. Charles, Illinois 60174. Phone (312) 584-2300.**

Altec introduces a 4-foot control console with up to 28 inputs and 16 outputs.

It's built to your specs...delivered ready to use.





It's the all-new, all-solid-state Altec 9300A control console. Only 51½ inches long, it features direct-plug-in modular construction that lets you custom tailor your own board by simply selecting the specific modules you need.

The new Altec 9300A gives you up to 28 inputs and up to 16 outputs. And any input may be connected to any output by means of a switching matrix on each input channel.

Here are some exclusive features designed into the new Altec 9300A.

- **Channel Check provides an individual instant check of all input lines without interrupting the program.**
- **A Pre Cue pushbutton transfers signals from the output buss to the cue buss.**
- **A Modulite® Visual Volume Level Indicator on each module tells exactly how much level is being fed to tape machines.**
- **Echo Facilities permit selection of internal or external reverb devices and a bright or soft timbre.**
- **Color-coded knobs enable fast and easy matching of input channels with correct output selector modules.**
- **22 dB of headroom.**

Mail this coupon for all the details on the new Altec 9300A console.

To: Altec Lansing, 1515 South Manchester Ave.
Anaheim, California 92803.

☐ Please send me all the details on the all-new Altec 9300A control console – including information on how its unique modular design will let me simply plug in different modules as I need them.

☐ I'd like to hear more. Please get in touch with me.

Name__________ Phone__________

Position__________ Studio__________

Address__________

City__________ State__________ Zip__________

ALTEC LANSING®

A QUALITY COMPANY OF LTV LING ALTEC INC

See this new Altec control console in action at the AES Meeting, April 27-29, L. A. Hilton Mission Room.

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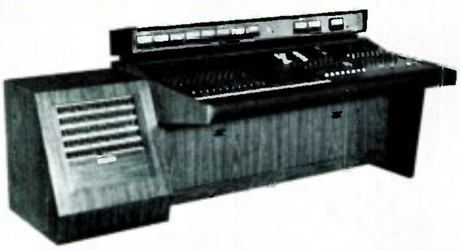

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