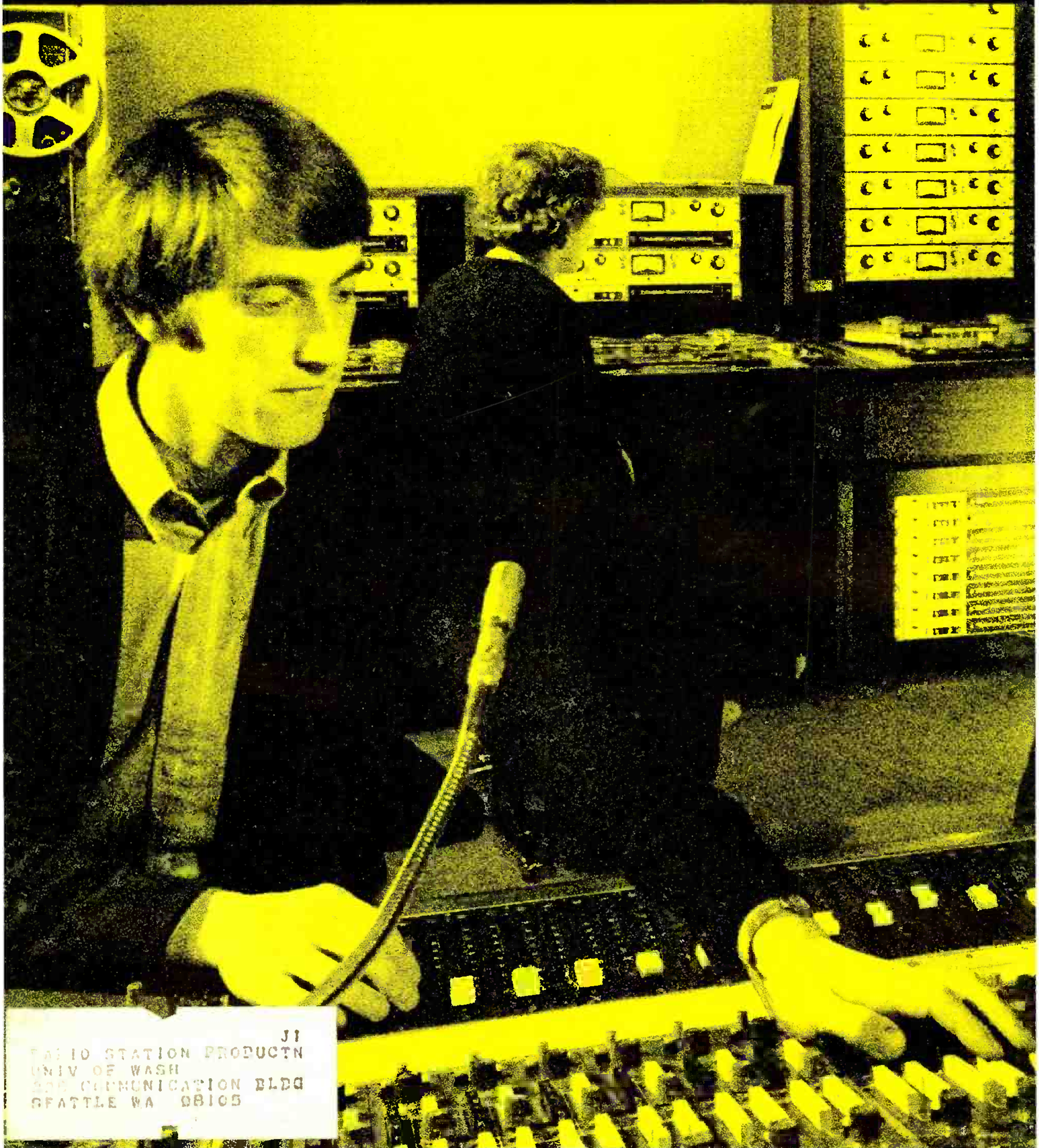
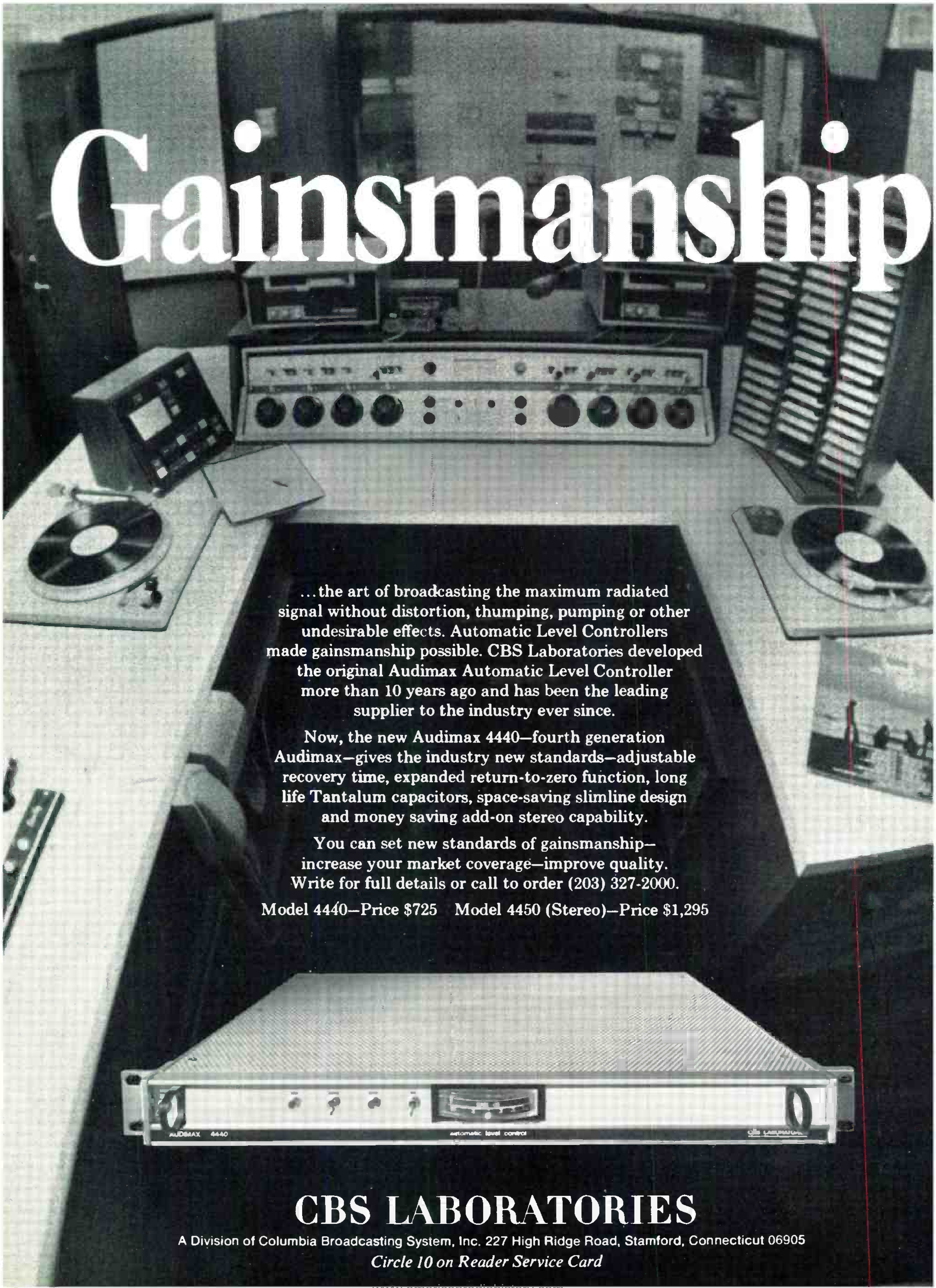
THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1971 75¢

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COMING NEXT MONTH

• Mel Sprinkle returns to our pages to begin a three parter entitled *Acoustics for the Audio Man*. In it he will detail those aspects of acoustics that apply to the audio profession.

Marshall King, who also has appeared in our pages before, describes the adventures of taking a sound department of a t.v. remote out into the field. It's great reading, and interesting, too.

The overworked and understaffed small radio station will want to build *A Transistorized Switcher for Stereo* as written by Ellwood W. Lippincott. This device can be the way to get those high-school kid announcers to get the right turntable or tape on the air.

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

• We are indebted to Dolby Laboratories for this wideangle view into the control room of London Weekend t.v. in London, England. Dolby 361 units may be seen between the Scully amplifiers on the two two-track machines, and below the transport, eight in a rack, on the eight-track Scully.

db

THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1971 VOLUME 5, NUMBER 2

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

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letters

The Editor:
Congratulations on the Teldec disc issue. I think it came out very well.

There are some significant factual errors in Martin Dickstein's article which I feel should be corrected.

1. In the second column on page 32 the reproducer is driven towards the center of the disc by a thin wire, not by a gear box and is returned (as of this time) by hand.

2. On page 34, the video disc is not 1 mm thick but 0.1 mm! That's a major difference.

3. On page 35, pulse-position modulation should be expressed as being not in the blanking interval between frames but between *lines*, a factor 525 more frequent!

4. On page 35 again; there is at this time no possible equipment for use by anyone but large-scale record companies for the cutting of such discs nor is such equipment contemplated, since the cost for producing equipment runs into many hundred thousands of dollars.

I think in all it was a very well prepared issue and the information with the exception of these few items, was most accurate.

Stephen F. Temmer
President
Gotham Audio Corp.
New York, N. Y.

The Editor:
My first thought on reading the December issue was that the half of my article on the t.v. disc should be entitled "Teld"—the second half to be called "Ec." The article split me right in the middle of a breath, as I was rising to a major point, that the Teldec disc offers a freeze action that does not compare favorably with that of EVR, which makes the still picture a major aspect of its new configuration. I offer this comment in case some readers are unable to patch together the December and January components of my phraseology.

Actually, I write with a different thought in mind, to say what perhaps the Editor has too modestly refrained from saying. Being a bit on the inside, I am aware that not only was db caught up in a printers strike last fall but—much worse—the printer in question simply went out of the publication printing business, thereby solving his labor problem on the spot. The Editor was left with a ghost magazine, for the time being, and hence all disruptions since, including those affecting my own writing. The fact that this letter is printed at all indicates that recovery is well on the way.

Edward Tatnall Canby
New York, N.Y.

Editor's Note
A massive attack of gremlins struck at the R. N. Andrews article in December to wit:

Figure 1 on page 23—the uoab's input polarity is reversed. The transistor's emitter and collector are reversed. Four points should show electrical contact. They are at the crossing just after C3; just below C11; just below C9; and between the 47 µf capacitors between the bases of Q1 and Q2.

Figure 2 on page 24—shows a four-position switch in the lower right. The +6 dB switch point should be labelled +16 dB.

Finally, a total of five lines of copy got misplaced. On page 24 the last three lines of the first column along with the first two lines of the second column should be repositioned as one block to fit into place three lines from the bottom of column 2.

We regret the confusion that these errors must have caused.

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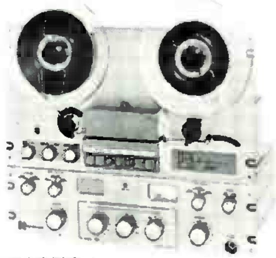

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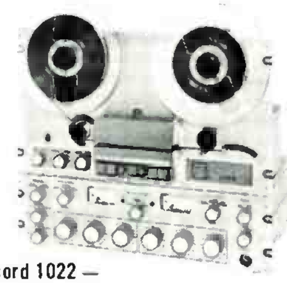

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George Alexandrovich

THE AUDIO ENGINEERS HANDBOOK

• In the past I have talked about the difference between echo and reverberation. This time I shall attempt to highlight the operating principles of the available equipment and devices which have appeared on the market since then—I would also like to mention a few ideas which seem to be practical enough for somebody someday to put into use.

Summarizing the difference between reverberation and echo—reverberation is very fast random repetition of an audio signal with decaying amplitude. Time between successive repetitions is normally less than 0.1 second. Echo is a distinct repetition of the same audio signal with decaying amplitude but with repetition intervals long enough for the human ear to distinguish each successive repetition. There is another kind of echo or sound-delay device which repeats sound only once (or several times) for use in sound-reinforcement systems in large auditoriums.

Techniques using sound-delay effects and reverberation are in wide use today. Equipment designed to generate signal delay and reverberation can be divided into two groups—electro-mechanical systems and strictly electronic.

Electro-mechanical systems are represented by the well-known EMT plate which is excited by an electro-mechanical transducer and where sound is traveling from edge to edge of the plate (similar to a ball on a pool table), until it is picked up by another transducer(s) located on the plate. Advantages of a plate are pleasant sound, relatively long decay, and an ability to have a stereo effect using two pickups. Disadvantages are size and susceptibility to ambient noises.

Spring-type generators consist of several springs stretched between the transducers, a pair of transducers for each spring. Excited from one end by a transducer into tortional mode, a spring delivers an impulse to the other end with a certain delay, then the same impulse is reflected back. This bouncing back and forth along the spring generates repetition of the signal in the pickup transducer. Since several springs are used each with different delay constants the over-all effect resembles that of sound reverberation in a hall, where sound reflections from the walls are arriving at different times to the listener. Advantages of spring-type generators are compactness, simplicity of mechanical design and low susceptibility to ambient noises when proper shock mounting is used in design.

One common mistake made by some is trying to feed the output of the reverberation generator back into its own input such as in tape delay units. Oscillation is usually the result of this attempt. Neither the EMT plate nor spring units can safely operate with positive feedback (since phase is changing all the time in reverberation devices one can assume that 50 per cent of the time the reinserted signal will be in phase with input signal).

Tape-echo units are usually tape recorders with a continuous tape loop which records incoming signal (or dry signal), then plays it back a fraction of a second later as the tape passes over the playback head. Part of this signal is fed back into the input for re-recording, creating a condition so the signal can be repeated an infinite

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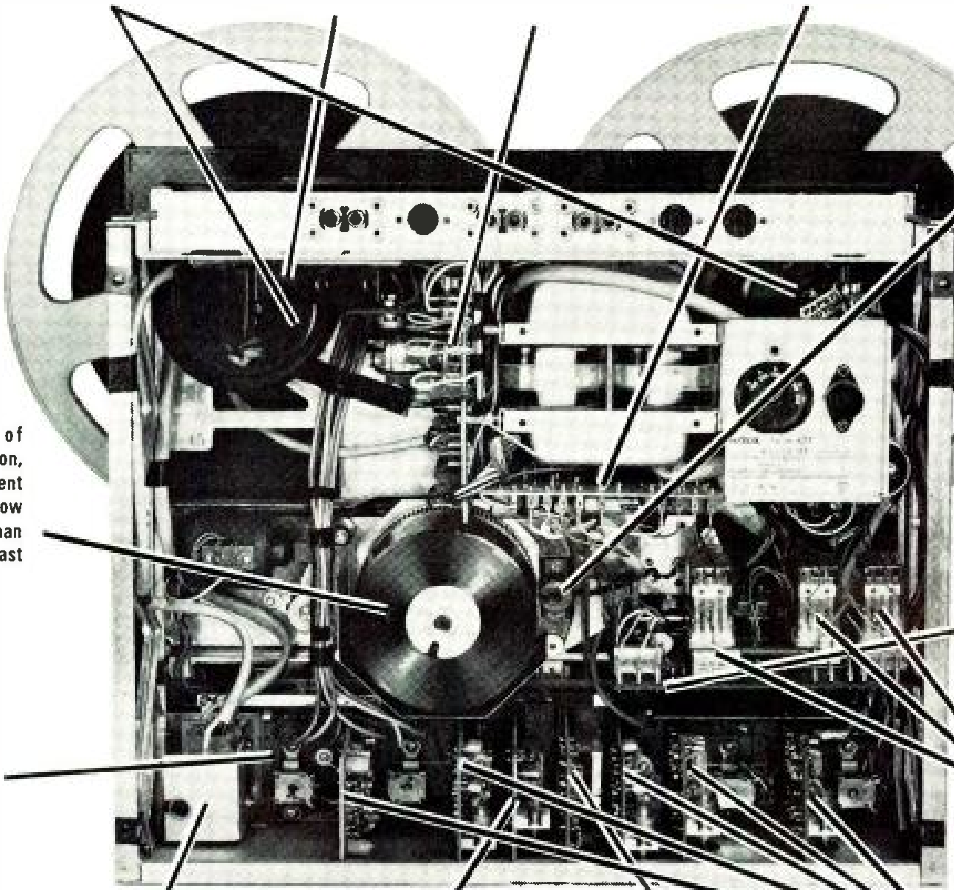

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amount of times (if re-recording process will not deteriorate the quality of the sound.) Speed of the tape and the distance between the recording and playback heads determines the time interval between the successive repetitions of the original signal.

Since a tape loop eventually wears out, attempts were made to substitute tape loop by rotating a disc covered with a thin film of oil. Instead of magnetic tape heads, a contact probe would electrostatically record signals on the film of oil—then as the disc rotates, charges would be picked up by other probes for reproduction. Difficulties of maintaining such a system have prevented it from becoming popular.

There have been other attempts to create sound delays such as using long hollow tubes. One end of the tube might have a small speaker-type transducer mounted on it, the other end a small microphone. The time it took the signal to travel through the tube was the delay. Advantages of this system are that there are no moving parts. However, absorption of certain frequencies by the pipe walls produces unsatisfactory frequency response. However, I think what with having so many new acoustic materials, plastics, and foams that this idea is worth looking into again, especially, if one considers small valves along the length of a tube to tap a portion of signal at different points of its travel for more realistic effects.

One of my friends—an acoustical and electronics consultant to architects—had an idea for building a long acoustically-isolated pipe into the walls of the newly-planned building so that signals from the stage or podium could be fed into this tube and picked up at any desired point for reproduction already with the required delay.

All the equipment described thus far relies heavily on transducers of one sort or another. 1970 saw the appearance of the first all-electronic delay equipment. No moving parts, no transducers, no gears, but plenty of I.C.'s and magnetic memory cores. Delay of the signal is achieved by converting analog information into digital and storing it in a memory-core system, then a fraction of a second later retrieving and reconverting into an analog signal, and remixed with dry signal or used all by itself as a delayed signal. In the meantime, memory cores are demagnetized and ready to receive new information for storage. This system is electronically complex and costly. Wide acceptance of such a device depends not only how soon one can get delivery but how reliable, economical, and flexible it will prove to be. All ingredients for successful approach seem to be there—sampling rate is high enough for low distortion and noise, it is compact enough, and performance specifications look good.

The popularity of signal-delay systems has increased with increased use of delayed sound-reinforcement techniques. In the large auditoriums, concert halls, or theaters if speakers in the back of the audience reproduce sounds originated on the stage without delay then listeners in the rear of the theater or auditorium will hear sound come first through the speaker, then a fraction of a second later the same sound will come directly from the stage. Acoustical delay has to be duplicated in a sound-reinforcement system so that both sound from the stage traveling through the sound-reinforcement system and directly will arrive at the same time to all listeners. This infers that every set of speakers equidistant from the stage should have the same specific delay time.

The closer the speakers are to the stage the shorter the delay required. Audio Instrument Company has for years been making a special closed-loop tape deck with one record head and several adjustable playback heads. Each playback head can be moved closer or further away from the record head. As the tape travels over all these heads a recorded signal is reproduced several times.

Presetting of delays or head adjustment is accomplished when the system is installed into a specific auditorium or theater and operating. Signal bursts are fed into the sound-reinforcement system, then delay adjusted at each speaker or group(more than one speaker can have the same time delay), until the acoustic echo effect is eliminated. Spring units are sometimes used to accomplish a similar effect, but the precision with which tape delay can function is not possible with springs. Spring delay units *are* very effective when the auditorium is acoustically dead and is in need of artificial reverberation and delay. *However, these techniques should not be used in motion-picture theaters.*

One of the little-publicized problems where delay systems are a must is in sattelite video links, when audio travels through cables and picture via sattelite. Distance traveled by the video signals is *so* much longer than the audio path that one has to delay the audio in order to lipsync signals.

Techniques of delaying signal and reverberating it artificially are heavily realized by today's technology. It is hard to predict over what route the development of new approaches will advance, but one thing we can be sure of is that this is only the beginning. ▪

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ARNOLD SCHWARTZ

THE FEEDBACK LOOP

• It's amazing how the human mind can complacently accept some concept or set of facts in the face of evidence which should at least set up some mental perturbations. One of these hard and fast concepts of mine has been the conviction that phase shift in a single channel of an audio system cannot easily be detected by the human ear, and that it does not cause any detectible degradation of the audio signal. An excellent article on the subject of the audibility of phase shift, entitled *Standardization of Monaural Phase*, appeared in the September issue of the *IEEE Transactions on Audio and Electroacoustics*. The author, David Stodolsky, concludes (on the basis of a survey of phase-shift studies) that phase shift can be detected by the human ear and that phase shifts do result in a degradation of the audio signal. Although a number of articles have appeared on the subject of phase-shift audibility for the past ten years, to date none had disturbed my belief in its inaudibility.

PHASE SHIFT EXPERIMENT

My attitude stemmed from an experiment I made a number of years ago. At that time I was working on a device that required the introduction of some drastic phase shifting within each channel of a stereo signal. However it was not known whether the introduction of a phase shift in excess of 45 degrees could be heard, and if so, whether it would result in a deterioration of the signal fidelity. Accordingly, an experiment was made to determine whether or not the ear could detect a phase shift of this magnitude. A network was used which had a phase shift of 45 degrees per octave over the entire audio bandwidth, was flat in the range of 20-20,000 Hz, and had unity gain. We then tried to determine whether the listener could detect any changes in a single audio channel while listening over a loudspeaker, when the network was switched in and out. About six or seven experienced listeners were used as subjects. Each was asked if they could detect any difference in the reproduced music when they switched between *A* and *B*. Each listener performed the test by himself in a "typical listening room". The subjects were not told what the two switch positions represented. None of the listeners could detect any difference between the in and out condition of the phase-shift network. I did notice, however, that setting up the experiment with earphones created a difference between the two conditions with certain types of sweep frequency signals. I rationalized this as not being significant in the face of the unanimous judgement (including my own observations) that no difference could be detected when listening to music over loudspeakers between phase-shifted and non phase-shifted signals.

In retrospect, I would hardly call this experiment conclusive; but I did conclude that if this large phase shift was not heard, that the much more modest phase shifts encountered in actual audio systems would have even less chance of being detected. One suggestion that Stodolsky makes may be relevant to the results of my experiment. He states that because we listen almost exclusively to recorded rather than live music that "many of us could have become 'phase' deaf . . . by continually listening to reproduced sound." In his article he cites cases where the acuity of human hearing of certain sounds is learned, and that some hearing is completely insensitive to certain sounds because the listener has little or no occasion to hear them.

QUANTITATIVE EFFECTS OF PHASE

Stodolsky attempts to make a quantitative measure of the effects of phase shift, and comes up with two equivalencies that relate phase shift to the more familiar quantities of frequency response and distortion. He states, "Our conclusion then is that a system which maintains a 3 dB tolerance in frequency response should also maintain a 17-degree tolerance in phase shift." His second equivalency relates to absolute phase distortion (polarity or phase reversal) and he derives a graph relating phase and both harmonic and intermodulation distortion. As an example of this latter equivalency he states, "Thus, high S.p.l. phase effects should be more detectable than 11.5 percent i.m. distortion."

PHASE SHIFT IN PRESENT AUDIO SYSTEMS

If indeed Stodolsky and others are correct in holding that the ear can detect phase changes, then it is time to carefully evaluate and correct phase distortion in audio systems. Here are just a few examples of where we can expect to find phase shift.

A 6 dB/octave slope will cause significant phase shift starting one decade from the 3 dB point. An equalizer with 6-dB attenuation or boost can cause a phase shift of up to 20 degrees.

In electronic equipment, frequency dependent amplitude changes, and the resultant phase shifts, are usually gradual in nature. On the other hand, electro-mechanical devices can often exhibit sharp amplitude changes over a narrow frequency range which can cause significant phase shifts. Examples of this are the stylus-groove resonance of the phonograph cartridge and the speaker cone resonance. A speaker system not only has cone resonances, but phase-shifting crossover networks as well. In the typical high quality three-way speaker system, there are three separate cone resonances with their accompanying phase shifts, and two crossover networks as well.

Recently introduced devices use large phase shifts for purposes of processing stereophonic signals. The problem of phase shift audibility should be carefully evaluated when these devices are introduced.

In this era of multi-channel recordings, the final master tape is a combination of many audio channels. Any one of these channels is a possible source of polarity reversal. If polarity reversal is a cause of distortion, as Stodolsky suggests, then care must be taken to maintain the phase integrity of all audio channels involved in processing the signals. ■

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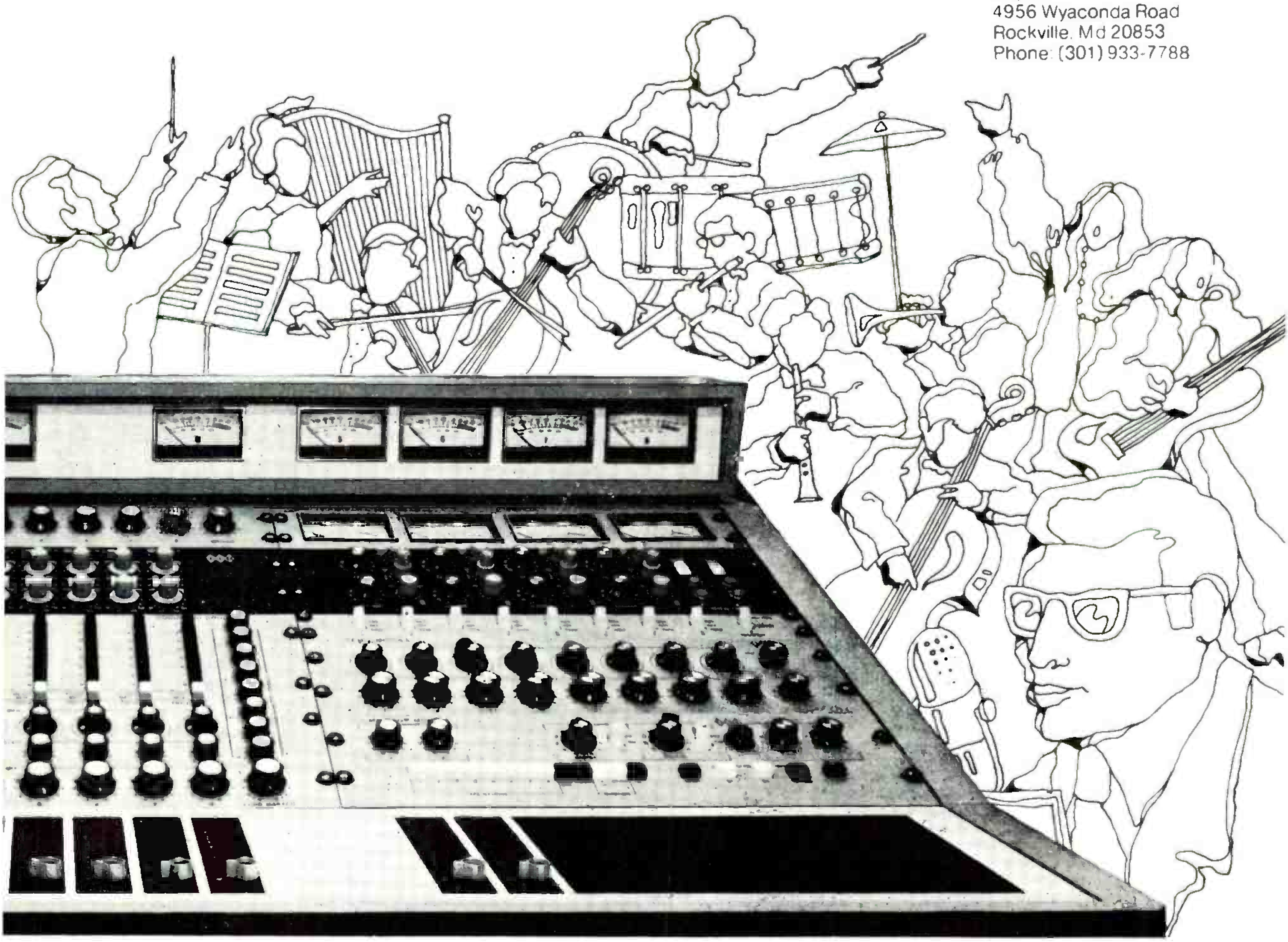

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John M. Woram

THE SYNC TRACK

The Problems of Phase

• At this stage in the state of the art (whatever *that* means), most, if not all, recording equipment is designed to preclude 180-degree phase shifts as various devices are switched or patched in and out of transmission lines. And, American microphone manufacturers have more or less arrived at a standard pin assignment. Usually, pin 1 is ground, and pin 2 is positive with respect to pin 3, when a positive pressure is produced at the diaphragm.

However, there is certainly more to correct phasing procedures than eliminating out-of-phase producing insertions, and making sure your microphone plugs are all the same.

When more than one microphone is used on a single instrument, there may be some acoustic phase distortion. If the two mics are feeding the same track, the distortion may be heard immediately, and the problem corrected. However, if the two mics are not being combined on the spot, the condition may escape notice until it is too late. To illustrate, let's jot down a few facts and an equation, and see how fast we can get into trouble.

The musical note, A, has a frequency, F = 440 Hz.

Sound travels at a velocity, V = 1100 ft/sec.

The wavelength, λ, of A = $V/F = \frac{1100}{440}$ = 2.5 feet.

So what? Well, since you asked, consider what may happen with two close up microphones on a piano. One mic is, say, x feet away from the A string, and the other is x + 1.25 feet away, as shown in *Figure 1*. From the illustration, we see that M1 reads the positive peak as M2 reads the negative peak, giving us a grand total of zero. From this, it appears as though the A string will disappear if the outputs from the two microphones are combined.

Of course, this is a laboratory type of condition and assumes the A string is a point source of sound and that the A itself is a pure sine wave. In practice, neither condition is true—the A is produced by several strings which are not in precise tune with each other, and the sound originates from a wide area. What with all the other variables of the piano keyboard, the end result of combining these two microphones together may be anything from a severely distorted frequency response to an over-all attenuation of the whole piano, or both. The wavelengths produced by the piano range from 40 feet to about 3 inches (27.5 Hz to 4,187 Hz) so no matter where the mics are placed, there is bound to be some cancellation due to acoustic phase shifts. However, by moving one or the other microphone slightly—maybe even reversing its *electrical* phase—the cancellations may be minimized.

Intentional phase cancellations may be used to produce unique equalization effects. For example, consider the two microphones, M1 and M2, in *Figure 2*. As in *Figure 1*, there will be cancellations. However, if the microphones are placed in front of a guitar amplifier, which is more or less a point source of sound, a little experimentation may yield some unique effects. Two electrically in-phase microphones, spaced at a distance, D, will cancel frequencies with a wavelength of 2D, and the uneven octaves (3F, 5F, etc.) above this frequency. If one goes much below this particular frequency, there will be little or no cancellation. However, if the mics are now switched electrically out-of-phase with each other, the low end will be substantially attenuated, and F and its uneven octaves will add rather than cancel out. Thus, two effects which would be all but impossible to achieve with equalizers are available to the engineer by taking intentional phase cancellation into account.

Getting back to unintentional cancellations, the leakage from the piano into, say, a guitar mic may be acoustically out of phase with the actual piano mics. If the leakage is appreciable, the over-all sound may be affected if the guitar is combined with the piano, either on the session, or later. Depending on the amount of out-of-phase information present, it may be advisable to move the guitar microphone slightly, or to switch it electrically out of phase at the console.

Another potential trouble spot is the area in the rear of a figure-8 microphone. The off-axis (90 and 270 degrees) rejection of a condenser figure-8 mic is impressive because the mic usually contains two elements that are connected out-of-phase with each other to create the figure-8 pattern. Signals originating off axis are thereby

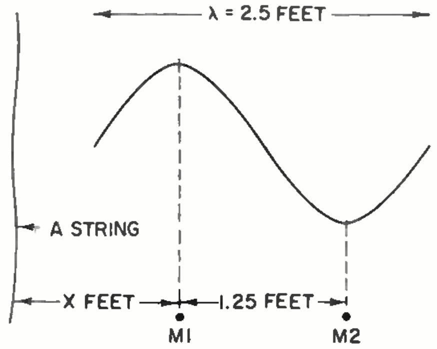


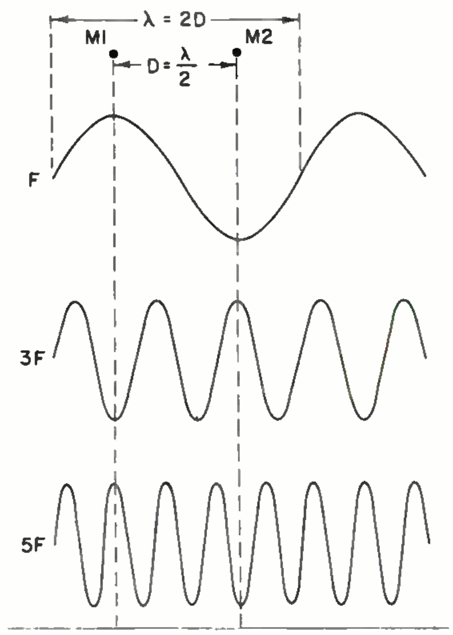
Figure 1. The wavelength of A-440 Hz with two microphones spaced at a distance of $\frac{\lambda}{2}$.

cancelled electrically. (The same elements connected in phase would yield an omnidirectional pattern.) In a ribbon microphone, pressure exerted from off-axis signals is equal on both sides of the element; hence no signal.

In either case, the out-of-phase condition between the front and rear of the microphone may usually be ignored *unless* another mic is being used in the vicinity of the rear of the figure 8. *Figure 3* illustrates this. Trumpets and trombones are being picked up by the figure-8 microphone, and everything is fine until the cardioid mics are brought in to pick up the clarinet and flute. Now, trombone D is also picked up by the flute microphone which, although it is in phase with the clarinet mic and the front (+) of the figure 8, is out-of-phase with respect to the rear (−) of the figure 8.

With any sort of luck, it would be very easy to spend most of the session checking phase relationships between each instrument and determining whether the shifts are acoustic or

Figure 2. Gain of any frequency, F, and the first two uneven higher octaves. These frequencies may be cancelled by combining mics M1 and M2.



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Nothing Stands Between You and the Sound

Both sections are powered by balanced positive *and* negative supply voltages (not just positive and ground), so there need be no coupling capacitors or interstage transformers between you and the sound.

Without them, the TA-1130 can extend its power band width down to 7 Hertz, and actually *exceed* its rated damping factor of 100 all the way down to 5 Hz.

An Abundance of Audiophile Conveniences

Of course, the TA-1130 has all the control facilities that you could ask for: low and high filters, tape monitor, a speaker selector, and even an Auxiliary input jack on the front panel. The selector switch is Sony's instant-access knob-and-lever system.

There's even provision to use the TA-1130's power amp and preamp sections separately, to add equalizers, electronic cross-overs, or 4-channel adapters to your system.

In fact, you can even get the power output section separately, as the model TA-3130 basic amp. It makes a great match for our TA-2000 preamp, too.

Your Sony dealer has both models available, and at prices—$359.50 for the TA-1130; $239.50 for the TA-3130. Sony Corporation of America, 47-47 Van Dam Street, Long Island City, New York 11101.

SONY® F.E.T. Amplifier

electrical—and what to do about them.

Some of this agony may be eliminated by making *sure* all your electrical connections are in order. It is remarkably easy to accidentally reverse polarity in a condenser microphone as the signal travels from the microphone to the cable—to the power supply to console. And, European manufacturers of dynamic mics don't necessarily conform to the general American practice. At least one manufacturer includes absolutely no clue as to where the pins at the base of his microphone are connected. Be careful—and make sure you standardize your microphones and cables *before* you put them into service.

Along with multi-track recording, there has come a tendency to monitor through more than two speakers. This is certainly very impressive, but in addition to sounding more spectacular than the eventual finished product, it also further obscures phase cancellations that may become apparent later on, when your master tape is heard over one speaker. At least once in a while, its a good idea to listen to the whole racket in mono, just to see if anyone disappears completely.

Two valuable electronic aids exist to help the engineer in spotting phase problems. Probably the most well known is the *oscilloscope*. With one signal applied to the vertical deflection plates and the other to the horizontal plates, phase and amplitude relationships may be observed on the 'scope face. Although any x-y 'scope may be used for this purpose, Gotham Audio recently announced a Stereo Monitor Oscilloscope designed expressly for this purpose.

Another device is the *correlation coefficient meter*, which analyses the compatibility between two complex signals. To get some idea of what the meter does, examine the two waveforms A & B shown in *Figure 4*.

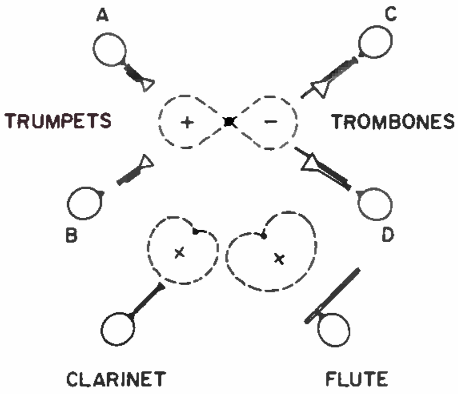


Figure 3. Microphone placements. Trombone D may be cancelled due to combining the flute mic with the rear of the figure-8 mic.

Since they are both simple sine waves of the same frequency, it is easy to see that they are 90 degrees out-of-phase. If they were more complex, their relationship would not be quite so obvious. But, regardless of how complex the wave shapes become, there is at least one measurable condition at any given time. Either;

Condition -A- Both waveforms are Positive (or negative) at the same time.

Condition -B- Both waveforms are NOT positive (or, not negative) at the same time.

Notice that during certain intervals (T1, T3, T5, T7) either wave A is positive while B is negative, or *vice versa*. During the other intervals (T2, T4, T6, T8) either A and B are both positive, or both are negative.

If a center-zero reading meter can be constructed to give a positive deflection when condition A exists and a negative deflection when condition B exists, the meter will give us an indication of the relationship between the two signals. Observing the square wave in *Figure 4*, we see that it is positive during condition A and negative during condition B. Since, in

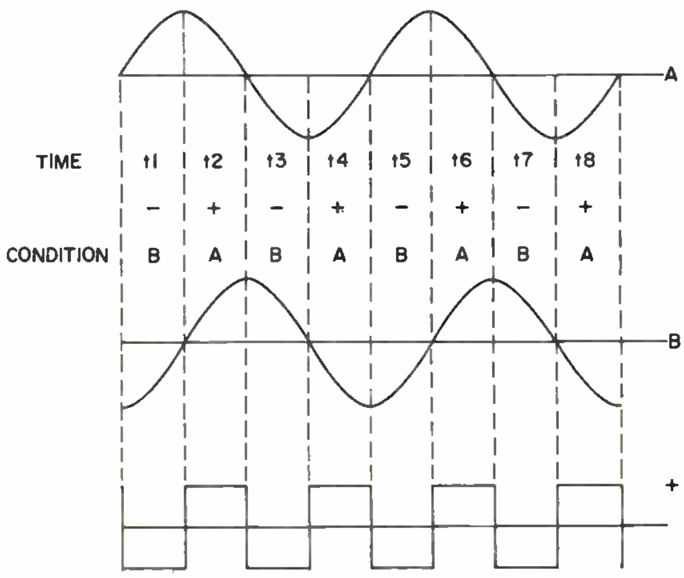
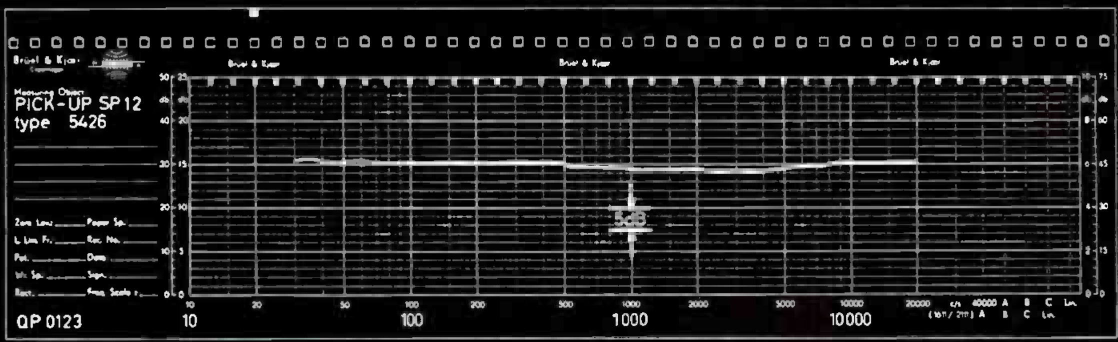


Figure 4. Waveform B is 90 degrees out of phase with waveform A. Below, the theoretical square wave representing the relationship between waves A and B are shown.


Brüel & Kjær
PICK-UP SP12
type 5426
QP 0123
5dB
10
100
1000
10000

this example, both conditions exist for exactly the same time durations, the average value is zero. If the waves were completely in-phase (0 degrees) the "square wave" would become a steady positive voltage. If the waves were completely out-of-phase (180 degrees) the "square wave" would be a steady negative voltage. Any condition between these two extremes would produce a square wave with some measurable + or − average, which, after filtering, could be indicated on a meter. This meter indication is an indication of the correlation between the two waveforms.

During condition A, the two waves are alike in sign and so, may be considered additive. During condition B, the waves are opposite in sign, and therefore subtractive. So, if the waveforms are to be eventually combined, Condition A should predominate. That is, the meter should read to the right of center. Readings to the left of center indicate the presence of out-of-phase, or uncorrelated information that may subtract when combined, or cause an excessive vertical component during tape-to-disc transfer. The member may be of particular value during multi-track recording sessions. New material may be compared with previously-recorded tracks and the amount of correlation noted as the additional tracks are recorded.

All you have to do is produce the square wave described above. If I can, I'll let you know about it in the next column on this subject. ■

NORMAN H. CROWHURST

THEORY AND PRACTICE

• First I would like to thank the people who have written to me congratulating me on my appointment as Assistant Professor at Teaching Research, which is located at Oregon College of Education in the town of Monmouth, Oregon. Do not let the fact that you may never have heard of it fool you. Things seem to be working out in a variety of ways.

Teaching Research is, in Oregon, something unique. Although the division is located at the College of Education, it is not under that administration, but directly under the Chancellor of Higher Education for the State. As such, it has acquired an interesting group of innovators, who have made quite a nation-wide reputation for themselves.

I was alerted to this quite quickly by the frequency with which various people were wanted on the telephone by "Washington," people in the United States Office of Education. At a conference in Seattle, about a month after I started work there, I met some of these Washington people, as well as some other researchers from all over the country, and learned a little more about the division with which I have now associated myself.

Many times during the last couple of months (as I write this, it will be longer by the time you read it) I have been reminded of my earlier career in audio. Then we were a bunch of young men who were, in one respect, perfectionists. And because of this, we were never satisfied with our work. In our own view, we "fouled up" something shocking.

We did get the job done, and we had a commitment that the show must go on. But some of that skin-of-our-teeth business by which it got on only just in time could get too close for comfort. The fact seems to be that this kind of "operation" is the heritage of people who accept challenges. For the people who do not periodically find themselves in this kind of corner, either do not take the risk, or else do not deliver the goods.

So working with Teaching Research is ringing a familiar bell, in a totally new context. Their hiring of me is evidence of this, in itself. According to the record, I have no background in teaching. But the record is not complete. I did considerable teaching in England, years ago, for which the record is not available, due to enemy action—we had a little war, you know.

The interesting thing is that my writings in db and other publications, including my book *Taking the Mysticism from Mathematics*, helped sell me to the Teaching Research management people. They recognize that most people raised in education do not possess the imagination and sense of challenge needed to solve today's problems in education. So "Here we go again!"

Now back to audio: a letter from a reader asks about the possibility of applying the function generator type of audio oscillator to the design of a sweep generator. The answer to this question depends on what the sweep generator is wanted for. And there are two parts to the answer. The other concerns design theory.

Taking that first, any sine-wave oscillator is difficult to sweep at a high speed with stability, because instantaneous frequency depends on one set of circuit values, while oscillation at a stable level depends on another set. Changing frequency involves changing the first set, and as this happens, an automatic circuit usually has to adjust the second set of values so the amplitude of oscillation is maintained constant.

It is true, the function generator approach avoids this interdependence problem. Amplitude can be absolutely controlled, independently of instantaneous frequency. If necessary, the generator can produce one hundredth of a cycle of a particular frequency, of just as definitely controlled amplitude (only you cannot see the amplitude of such a small piece, of course) as whole cycles of it can be produced.

The frequency can keep changing, without the amplitude changing at all. And the control of frequency can be achieved by a variable voltage, so that any desired rate of change of frequency can be designed into the sweep, without any serious problem.

With the sine-wave oscillator, to get a logarithmic sweep through frequency over, say, 3 decades, such as 20 Hz to 20,000 Hz, required a skillfully-designed logarithmic component that would sweep frequency, or some derivative of it, over that range, with logarithmic law.

It was extremely difficult to do this with r-c circuits, although Hewlett-Packard made such an oscillator for a while—it was later withdrawn from their catalog. The only type that has been consistently available uses the heterodyne technique of beating two radio frequencies together and producing the audio tone as the difference frequency.

Suppose one frequency is precision set to 200 kHz. To get a frequency of 20 Hz requires the other oscillator to have a frequency of 199.98 kHz. To change by an octave, which is one tenth of a 3-decade dial, the frequency needs to change only to 199.96 kHz. For the full change, frequency needs to change to 180 kHz, and the last octave, also one tenth of the dial rotation, requires a change in this frequency from 190 kHz to 180 kHz.

Thus at one end of the sweep, the change must represent 0.01 per cent of the actual frequency, while at the other end, the same movement must represent 10 per cent of the actual frequency. This is a difficult component to design, to say the least. But several good ones are on the market. They are mechanically swept, so it takes about a minute to take a frequency run.

The convenience of a high-speed display, such as is used for aligning radio and i.f. circuits, has often appealed to audio people. Why cannot the whole audio spectrum be scanned in something like 1/60th second?

Some of the sweepers made to work with function generators cover the same order of frequency range, such as from 10 kHz to 20 mHz, and sweep it at any desired speed, such as 1/60th second. Why cannot the same thing be done with audio frequencies?

Theoretically, it can. The charge-discharge values can be changed so that instantaneous frequency sweeps from 20 Hz to 20 kHz in 1/60th second, at logarithmic rate—or any other desired form. And it will do it perfectly, for whatever that means. So why is such a unit not available and in use?

Presumably the reason for wanting an oscillator or signal generator is to measure some kind of response: an amplifier, loudspeaker, acoustic system, or something that responds differently to varying audio frequency. To appreciate the problem, we must ask ourselves what *respond* means.

If voltage changes from zero to 1 absolutely linearly, in 1 microsecond, what frequency is this change a part of? If it is absolutely linear, then such a change continued indefinitely would go on forever, and the frequency is zero (d.c.). But assume this change is part of a 10-volt sinusoidal waveform, what then?

The 1-volt change is then about 1/60th of a period, so a full period is about 60 microseconds, representing a frequency in the region of 17 kHz. The nonlinearity, over this 1 volt step is 0.006 per cent.

If the 1-volt change is part of a 20-volt sine wave, it is about 1/120th of a period, representing a frequency of about 8.3 kHz, and the non-linearity is in the region of 0.0008 per cent. On the other hand, if it is part of a 5-volt sine wave, the frequency moves up to about 33 kHz, and the nonlinearity is about 0.05 per cent—almost measurable.

What this says is that whatever is being measured for response must have extremely linear response, for it to "know" what the segment presented to it, before frequency changes, constitutes a piece of. And as changing frequency will change the rate or slope of the voltage, this will make the frequency quite indeterminate.

In short, for the device to know what frequency it is momentarily responding to, it must be presented with a sufficiently large sample of that frequency to "recognize" as a frequency.

Assume we aim to produce an audio sweep, from 20 Hz to 20 kHz, in 1/60th second. Let us figure how long it takes to cover the first octave—a 2:1 frequency change. If the sweep is linear, it covers almost 20 kHz in 1/60th second, or 1200 kHz per second. The first octave is a change of only 20 Hz, which must thus happen in 1/60,000 of a second, or about 17 microseconds. Taking the average frequency (as it is linear sweep) as 30 Hz, a period of 33 milliseconds, the whole octave only contains about 1/20,000 of a cycle!

Change it to logarithmic, and the picture is a little better. Now the first octave will take 1/10th of the total sweep time, or 1/600th second: 1.7 milliseconds. So the first octave gets a total of about 1/20th of a cycle. Still not enough for the device you are testing to be able to tell the difference between one end of the octave and the other.

Human hearing requires several cycles of a frequency before it can identify pitch. And more physical devices still need more than a small fraction of a cycle to cover an entire octave.

Now suppose a sweep speed that scans the spectrum once every second is used. The first octave will get 1/10th second. With an average frequency of 30 Hz, this will be 3 cycles for the whole octave. Still rather fast, but definitely possible to get some kind of average response in this range.

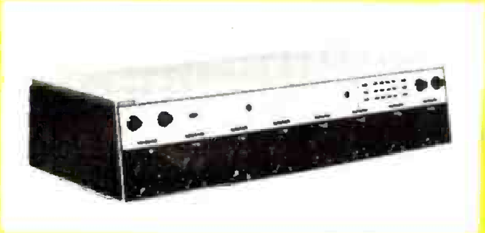
Actually, the adequacy or otherwise of a speed depends on the kind of response anticipated. Suppose, for example, the response is fairly uniform down there—no peaks. A fairly good response may be possible. But suppose there is a 10 dB peak at 30 Hz. Such a resonance will take about 3 cycles *at that frequency* to reach within 1 dB of the true response.

If the sweep only gets 3 cycles into the whole octave from 20 to 40 Hz, about 1/3rd of a cycle will be close enough to 30 Hz to contribute to response as the peak point. Thus, instead of registering 9 dB for a true 10 dB peak, you may only get a 1 dB peak reading, or perhaps for the whole octave, the reading may be about 0.3 dB higher than in the absence of such a peak.

However, if this frequency region is what you are interested in exploring, you do not need to scan the higher frequencies at all. So you could expand the frequency scale so you spend a whole second sweeping that octave. Then you would find the peak much better. And for this kind of flexibility in use, the function generator definitely has advantages, because all you have to do is scale control voltages up or down. ■

NEW PRODUCTS AND SERVICES

AUDIO CONSOLE



• A line of standard audio consoles for AM/FM/TV broadcast and recording studio applications has been introduced. The B-801 mono; B-802 stereo and B-803 dual channel models each feature eight input mixing channels with preselection of three sources for each mixer. Plug-in modules are utilized throughout. The standard complement of modules accommodate three microphone, four unbalanced high-level and one balanced high-level mixing channels. A total of twenty-seven input sources may be controlled. The modular, plug-in feature permits extreme flexibility in the console system since module connector wiring has been designed to permit interchanging of input modules. Thus to meet specialized requirements, all eight mixers can be assigned to microphone or high-level service or any intermixture desired.

Spare contacts on input preselect and lever-type bus selection key switches are provided to allow extension of studio speaker muting/warning light relay logic. Step-type, maintainable, channel attenuators are equipped with cue positions on all eight mixers. Headphone monitoring of eleven points within the console system is provided.

The consoles are completely self-contained and include complete, selective intercom facilities for three studios, four remote lines and a general paging location.

Mfr: McMartin Industries
Price: from $2350 to $3200
Circle 51 on Reader Service Card

COMPACT AUDIO AMPLIFIER

• Here is a replacement of its model SA 30-30 basic amplifier with the low distortion model D-40, a stereo amplifier that delivers 40 r.m.s. watts per channel. The D-40's compact design is only 1 3/4 inches high (less pictured walnut enclosure). Specifications are 40 watts per channel r.m.s. output into 4 ohms at 0.05 per cent t.h.d., 170 watts total IHF rating into 4 ohms. Intermodulation distortion is less than 0.3 per cent from 1/100 watt to 30 watts at 8 ohms. Power bandwith is ±1 dB from 5 to 50,000 Hz. Damping factor is over 200. Frequency response is ±1 dB from 20 to 20,000 Hz at any level up to 30 watts into 8 ohms, ±0.5 dB 5 to 100,000 Hz at one watt. A three year warranty is included on parts and labor.

Mfr: Crown International
Price: $229.00 (less enclosure)
Circle 55 on Reader Service Card

CONNECTION VERIFIER

• The Connection Verifier is a unique device that locates broken wires, bad connections, poor solder joints, and incomplete welds at the connector; all without disconnecting any wires or disassembling the connector. By varying the pitch of a continuously audible tone a good connection, a poor connection or a break can easily be determined. The unit weighs 10 ounces and measures 4 x 3 x 1 5/8 inches. It operates on a single 9-volt transistor battery.

Mfr: Electronic Tool Div.
C. H. Mitchell Co.
Price: $29.95
Circle 61 on Reader Service Card

"QUIET" LAVALIER MICROPHONE

• Annoying friction noise, usually a deterrent to the use of lavalier microphones, has been eliminated with the introduction of the new RE85 dynamic lavalier microphone. One design technique that allows this claim is the unique double-wall construction of the unit, with two separate cases being used. One case is nested inside the other and completely insulated from all shock and vibration with highly compliant rubber. The internal case is a capsule containing the generating element and isolated to a degree never before known in the microphone industry. Even cord-conducted noise has been eliminated with this sophisticated shock-mounting technique. Another factor contributing to the friction-free claim is the use of extra smooth microphone case finish and special, smooth coated cable. No grain or bumps are present to set up friction even when the microphone or cable is brushed against the user's clothing. It weighs 8 ounces, is 2-5/8 inches long, and has a diameter of less than one inch. When used with the supplied belt clip to hold the cord in a constant position and with even tension, it offers no hindrance to professional, hands-free use. A carefully designed integral windscreen and blast filter provides protection from wind noise, excessive sibilance, and pops. The outer case is turned from solid steel making it virtually indestructible. The RE85 is an omnidirectional microphone with frequency response 90 to 10,000 Hz, and it matches all low-impedance inputs.

Mfr: Electro-Voice Inc.
Price: $133.00
Circle 62 on Reader Service Card

NOISE EXPOSURE METER

• GR's 1934 Walsh-Healey noise exposure meter makes noise measurements leading to compliance with the Walsh-Healey Act without hours of spot testing, without calculation, and without error. The unattended 1934 automatically measures the noise exposure to W-H criteria and continually calculates the percentage of the daily maximum legal exposure accumulated during the work day. Since the 1934 is designed to measure the noise exposure in accordance with the applicable portions of W-H, ANSI, and IEC standards, all the data gathered from the 1934 are legally defensible. After initiation, the entire W-H measurement process requires no human intervention. Easy-to-use push-button operation allows the selection of the test period (time-on-the-job) in 8 ranges from 8 to 17 hours. Once started, the noise is sampled approximately twice a second and is automatically categorized and weighted according to its Walsh-Healey level. A digital counter accumulates the totals of both the weighted counts and the number of samples. At the end of the test period, the 1934 automatically stops but retains the data until asked to present the information on the high-intensity neon readout tubes; noise-level-*vs*-time data are also available for transmission to strip-chart or digital recorders. The maximum permissible exposure (100 per cent) is accumulated by the 1934 according to the conditions (or combination of conditions) detailed in Section 50-204.10 of the Walsh-Healey Act.

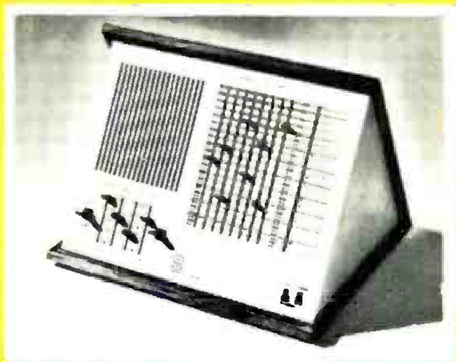
Mfr: General Radio
Price: $895.00
Circle 56 on Reader Service Card

AUDIO CONNECTOR ADAPTERS

• A new line of Q-G (quick-ground) audio connector adapters designed for use in situations requiring a transformer, attenuator, or other electronic circuitry in line with a microphone input, has been introduced by Switchcraft, Inc. The adapter is 4 inches long with 1 1/2 inches of usable space for extra circuitry between the unwired terminals of the male and female inserts at either end. Inserts are available in 3, 4 and 5 pin/contact configurations to accommodate a variety of standard microphones. The new line features ground terminal, ground contactors, and captive-design insert screw. These provide positive grounding/shielding of the shells, an insert screw that can't be lost, and quick and simple disassembly and reassembly. Hum and noise problems encountered at low signal levels are minimized by the use of insulation of high-impact, thermosetting plastic. They mate with Switchcraft Q-G cord plugs and receptacles and other connectors with similar insert arrangements and identical number of contacts.

Mfr: Switchcraft Inc.
Price: $4.50 (3 contact)
Circle 53 on Reader Service Card.

A COMPUTER FOR MUSIC



• A computer capable of composing and playing music for the home entertainment market may also find application in the professional audio field as well. The Muse uses a binary number system. Short tunes are created with the interval slides. The unit plays that melody and then plays variations based on the positions of the theme slides. This unit has application in certain areas of education to teach both music and logic and as a performance or compositional tool. Accessories include a light organ whose light patterns and rhythms are driven by the logic of the melody, and an external amplifier speaker with 20 watts of power.

Mfr: Triadex Inc.
Price: $300.00
Circle 57 on Reader Service Card

FREQUENCY EQUALIZER

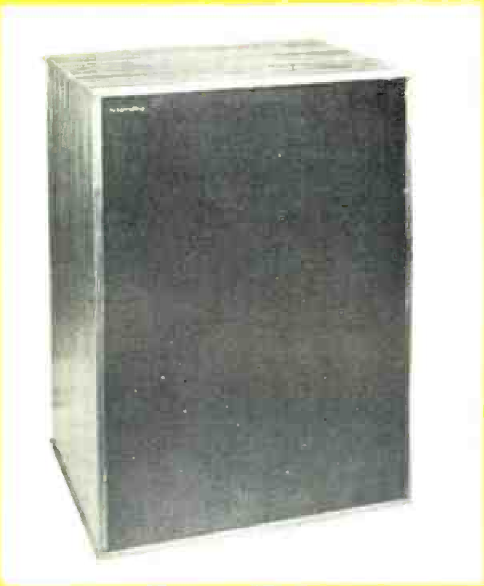
• The model 20-12 equalizer incorporates both active and passive circuitry to provide simplified adjustment of individual octaves for precise frequency balancing. Zero setting produces flat response. Response curves have been designed for optimum frequency control by octaves, with a broad skirt at low settings, and increasing sharpness as the control is cut or boosted to higher settings. In this way, a plus or minus 2-dB curve can be established easily in all but the sharpest and most critical peak and valley room conditions. A special equalization test record is supplied with each unit, and provides alternate pink-noise tones, alternating each frequency band with a 1,000-Hz reference tone, for rapid equalization of room conditions. This is claimed to be about a ten-minute operation. In addition to a 12-dB boost and 12-dB cut on each octave band, the unit also provides for adjustment of the total sum output of each channel, with an 18-dB range frequency-spectrum level control. Total available boost up to +24 dB and total cut down to -18 dB is claimed.
Mfr: Soundcraftsmen
Price: $259.50 (rack mtd. or cabinet)
Circle 50 on Reader Service Card

CASSETTE PROGRAMMING RECORDER

• A programming cassette tape recorder is now available from the exclusive national distributor for Coxco audio visual products. Known as the Coxco Series Y, the unit features two inaudible pulses. The first pulse is responsible for advancing a slide or filmstrip projector, while the second pulse stops the tape recorder automatically. A unique feature is that voice and pulses can be recorded either separately or in a single pass. Pulses can be erased independently and re-recorded at will. The Series Y can also be integrated with the Coxco student responder to provide step by step programmed instruction. This allows the student to progress through sound-slide or sound-filmstrip material at his own pace.
Mfr: Howe Folding Furniture Inc.
Circle 60 on Reader Service Card

STUDIO MONITOR

• The Studio Professional Monitor meets the needs for a monitor speaker for recording studios, auditoria, theaters, and discotheque installations that require extremely high-quality extended-range systems with a large power-handling capacity. The system utilizes a special version of the dual concentric loudspeaker which is combined with an enclosure incorporating a direct-radiating exponential horn and bass-reflex principle. It thus maintains its customary mid and upper-range frequencies while producing a strong bass. Frequency response is from 35 to 20,000 Hz and polar distribution for 60 degrees inclusive angle is -4 dB at 10,000 Hz. Cabinet size is 43h x 30 1/2w x 24d inches.
Mfr: Tannoy (America) Ltd.
Circle 59 on Reader Service Card



QUAD PANNER

• The Quad-Panner is a new mixing tool designed to provide quadraphonic panning in four unity gain channels. It is designed to enable any studio to do four channel positioning easily and inexpensively with its existing console. The four inputs are individually controlled with a joystick pan control, which positions the program material to any point in the quadraphonic area with constant acoustic level. It is balanced or mixed with a level adjustment control for each channel. A four-position blend control is featured which permits either extreme quad movement or three blended, limited spatial movement areas. These are available for mixing, when extreme ping-pong effects are not required. It uses active combining networks to provide built-in isolation and unity gain for ready patching in line level circuits. Input impedance is 600 ohms unbalanced and terminating. Output can be loaded with 600 ohms or higher. Max. level is +24 dBm and the noise figure is -77 dBm at the output. This is a s/n ratio of 80 dB when operating with a +4 dBm line level. The unit is battery operated.
Mfr: Quad-Eight Electronics
Price: $1250.00
Circle 54 on Reader Service Card

TAPE DECK

• A new three-motor three-head stereo tape deck has been added to the line of Sony professional-quality recorders. The model 640 incorporates many of the features found in the model 650 including a record equalization selector switch which optimizes performances with both standard tape and Sony SLH-180 low-noise, high-output tape; die-cast tape guide and head block mounting frame for perfect and permanent alignment of critical transport components; front panel sound-on-sound and echo controls; and microphone and line mixing. The 640 also has several ingenious features which set it apart from other low priced three-motor decks; such as Mechanical memory capability which permits timer-activated recording, playback, and shut-off; and positive-acting lever-type transport controls. Performance and tape handling are stated to be of professional standards with solid-state electronics, precise hysteresis synchronous capstan drive motor, and two outer-rotor torque motors for reel drive.

Mfr: Sony/Superscope
Price: $369.95
Circle 52 on Reader Service Card

COMPACT, WIDE-RANGE ATTENUATORS

• Easy to set and easy to read, two new attenuators, models 4436A and 4437A, cover the range from 0 to 119.9 dB and have a flat frequency response from d.c. to 1.5 MHz. Attenuation is set by front-panel controls in 0.1 dB increments, and the setting is read on a front-panel in-line readout. Maximum input power is 1 watt, which makes the attenuators suitable for a wide range of applications including power-level measurements, transmission efficiency tests, and gain or loss measurements of filters and amplifiers. Input/output impedance of the Model 4463A is 600-ohms balanced, and the 4437A is 600-ohms unbalanced. Both units use printed circuit switches, and a new drive mechanism designed to make them compact, light weight and rugged. The drive mechanisms, controlled from the front panel, are three rotary shafts on which are mounted four pairs of cams. Each rotary shaft drives eight printed circuit switches in the model 4436A and four in the model 4437A. These switches set up connections in the proper sequence to get the desired attenuation.

Mfr: Hewlett-Packard Co.
Price: 4436A – $300.00
4437A – $255.00
Circle 58 on Reader Service Card

Martin Dickstein

SOUND WITH IMAGES

The Beginning of Movies

ERRATA

In the story of the Mini-Fair at Burlington House in our November issue, we inadvertently reversed the captions of *Figure 11* with that of *Figure 12*. (or did we reverse the diagrams?).

• If you, or anyone you may know, had ever lost any money on "the sport of kings", consider it an investment in the beginning of motion pictures, or a contribution toward the development of the fastest racing animal in the world. Maybe this kind of logic won't make you feel a bit better about the loss, but you might be interested in some of the facts gleaned about the relationship between the race horse and the movies.

The story goes back to just a bit less than 100 years ago when a camera buff and a horseracing buff discussed some question about the race horse. One version of the question is whether the horse ever had all four legs off the ground during the race. To prove the answer, one way or the other, a battery of 24 still cameras was set up on one side of a track with a white reflector mounted on the other side.

A race horse was then run down the track and at each of the 24 camera locations the horse broke a thin string which had been stretched across the track. The pull on the string snapped the shutter of the corresponding camera and a complete record was then available in sequence. One of the two men (which one we do not know) won the discussion, but the biggest result of this run was that, shortly afterward, the inventor of the movie camera took up the challenge to devise a single unit which would be able to do the same thing that 24 did sequentially. Incidentally, the photographs could be seen in apparent motion in a device called a *Zoetrope*, a cylindrical affair with an open top in which the pictures were mounted around the inside. Spinning the cylinder gave the illusion of image motion. However, making a movie this way would have taken an enormous number of still cameras, even if the rate of motion of the film was 12 exposures a second, considered the minimum for a steady image illusion.

One of the biggest problems in the development of the motion-picture camera was the creation of the proper material for the negative. The still cameras used glass plates which obviously could not be moved quickly enough for motion pictures. First, unsuccessful attempts were made with cylinders like the original phonographs used. Different chemicals were used for emulsion. Images were to be very small, but in the enlargement for viewing, the coarseness of the coating was unacceptable.

Different mechanisms, various chemicals and larger images were tried. Still no good. Then came celluloid and at about the same time Mr. Eastman, of the now-famous film manufacturing company. His development of a narrow strip of sensitized film that would work in a roll-out manner helped immeasureably.

Meanwhile, back in the lab, the mechanism was being perfected. Attempts were being made to expose more than 40 frames every second. This meant that even with a hundredth-of-a-second exposure time, the rest of the second had to be divided evenly for smooth, continuous operation of the mechanism in pulling the film. Finally, the number of exposures per second was set at between 20 and 40 and the mechanism was developed to the satisfaction of the inventor. The film was made and delivered. Still no good. The film had been made half-inch wide but more than that was needed to allow for the room of the right size image and the sprocket holes for the pulling mechanism. The unit was modified, the film made and the first camera was finally operable.

To show the results of the camera, the pictures were presented in a cabinet with an electric motor, a battery, a fifty-foot film and a magnifying glass. The *Kinetescope*, as it was called, created quite a stir at the Chicago World's Fair of 1893.

When the inventor discovered that he had competition overseas because he had failed to patent his device there, he decided to invent a machine which would project the images large enough for an auditorium audience to see it on a screen. If the speed of the film was too slow, there would be a flicker. The thought of audience eye strain was even considered in the development of the projection device, and, of course, any changes made in the projector had to be reflected in the camera. Finally, the units were developed with 30 to 40 exposures per second as the film speed.

The inventor's noble concept of the use of motion pictures for making people happy, for educating them, for training purposes has grown far beyond what he could imagine. As early as 1902, Sears, Roebuck & Co., "Cheapest Supply House on Earth, Chicago," showed in its catalogue (recently published by Crown Publishers, N.Y.) a complete projection system for the price of $160.50.

The catalogue advertised "THE UNRIVALLED EDISON KINETOSCOPE, moving-picture machine, giving a pictorial presentation, not lifelike merely, but apparently life itself, with every movement, every action and every detail brought so vividly before the audience that it becomes difficult for them to believe that what they see before them can be other than nature's very self."

The unit itself is described by "Animated Moving Picture Outfits. Our complete moving-picture outfit, fitted with arc lamp for attaching to electric current, films, stereopticon slides and all other requistes for high-grade animated moving picture entertainment. 1 combined 1901 model Edison Kinetoscope with stereopticon fitted with arc lamp and rheostat. 1 12 x 12 screen. 1 set of stereopticon views, 25 in number, customer's selection. 300 feet of best selected film. 500 posters. 1,000 tickets. Rubber printing outfit, for filling in dates and places of giving entertainment."

Provision is also made for an additional $46.00 for substitution of calcium or oxy-hydro light instead of electricity. Then, at the bottom it says "DO NOT LET THIS MONEY MAKING OPPORTUNITY PASS AND WHEN IT IS TOO LATE' SAY, 'I AM SORRY I DID NOT TAKE HOLD OF IT.' "

Little did the inventor realize the type of films that would be made for public presentation. Oh, yes. The inventor. He also developed an elaborate improvement on the original "Mama" doll, a simplified stock ticker, revolutionized the portland cement business and developed a fast method for prefabricating houses by pouring concrete into reuseable forms, applied for a patent on the helicopter, devised the fire alarm and miner safety lamp, invented the gummed tape and waxed paper so commonly taken for granted today, and also has patents (for a total of 1,097 of them) on the electric light bulb and the phonograph . . . Thomas A. Edison. ■

CHARLES W. KING

Attenuation Pads

What audio man has not found himself in want of a particular attenuation pad. Here are the calculations necessary to solve these needs.

ENGINEERS and technicians involved in the design and use of attenuation pads in audio systems may find the tables presented here of practical assistance in their work. Only the more common types of attenuation pads (T, H, pi, O, bridged T and lattice) used in audio systems will be presented here. Many tables have been published over the years, but most of these contain the tabulated values for a specific impedance.

Charles W. King is a senior staff engineer with the DuKane Corporation of St. Charles, Illinois.

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Walter J. Gunn, Ph.D.

The Sound Comes From Up!

Scientific exploration sometimes finds creative possibilities for recording engineers. Are you recording in four-channel stereo? Try this effect and you may just have an interesting free fifth channel to use.

WHY do we hear mono material as a single source emanating from between two stereo speakers; Where do we localize mono material when played on a four-channel system? Before attempting to answer these questions, let me first review some of the basic notions about the localization of a sound source.

There are three basic cues for localizing the source of a sound in azimuth: the first of these is the difference in intensity of the sound at each ear. When a source of sound is not directly in front of the listener, there will be a difference in sound-pressure level from one ear to the other due to the presence of the listener's head between the farthest ear and the sound source. This difference is greatest for frequencies above 1000 Hz and can be as high as 20 dB. Lower frequency sounds would not be affected by the presence of the head and so there is another mechanism to handle localization of these sounds: this is the phase difference cue. Since the ears are separated by about six inches, the peak of the sinusoidal low frequency sound wave reaches one ear (the closest one) at an earlier time than it reaches the other ear. This is detected as a phase difference in the sound reaching the two ears, but is limited to frequencies below about 800 Hz. The third and final cue is the time of arrival of an impulse sound at each ear. An impulsive sound will reach one ear sooner than the other ear and this is used to localize the source of impulsive sounds such as a drum, a gunshot, etc. Since most of the sounds we hear are complex in nature and contain many different frequencies in their spectrums, all three mechanisms work together to give us an over-all impression of the location of the sound source.

All three of the above cues can be explained in terms of a single psychopsysiological mechanism, namely, "latency of neural response." By this, I mean the differences in the precise time the nerves in the auditory (hearing) system start to function determines where you hear the source of a sound. The more intense a sound, the faster the neural response. The earlier a sound reaches an ear, the earlier the neural response, and the earlier a peak of a sinusoidal sound reaches an ear, the earlier its peak neural response. It is the difference in the latency of the neural response of each ear for a particular sound that is probably used by the auditory system to determine the location in azimuth of a sound source.

It is thought that the pinna (external ear) is more than just an ornament for the head. It has been suggested that the function of the pinna is to cause phase shifts for various frequencies by means of its many curved channels, which when decoded by the perceptual system, indicate the direction of the source in the vertical plane.

When one listens to stereo material, he hears the various musical instruments spread out between the two speakers. When mono material is played on a stereo system, the listener hears a single source of sound located between the two speakers. This is because the only single source of

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Dr. Walter J. Gunn is with the School of Public Health and Administrative medicine of Columbia University.

sound which could create identical sensations of this nature in both ears would be a source located straight ahead, between the two speakers.

The next question which would naturally arise is, "Where do you localize a single sound source when it is presented by more than two speakers"?

I tried it in our lab using a four-channel tape recorder, two stereo amplifiers, and four speakers. Each of the four speakers was located in one of the four corners of our sound-proof room, which has dimensions of 18 feet by 14 feet. First I tried a mono voice recording and listened to it while standing in the center of the room. If the voice had said, "In the beginning, I created . . .", I would have thought that it was God, because the sound seemed to be coming from directly above me! Several other people were asked to listen and all heard it as coming from above. I then played a mono recording of an airplane flyover, and sure enough—it flew *over* the room rather than under or through it. Someone suggested that perhaps we hear it from above because we expect to hear airplanes up there, so we tried it with a recording of a motorcyle. It also flew overhead! The rather amazing aspect of this phenomenon is that the sound comes from above even when the speakers are at a lower level than the listener's head.

It occurred to me that one important application of this phenomenon would be in synthesizing the directional characteristics of jet flyovers for our research here at the lab, which is a NASA-sponsored project. The next step was to try to get the perceived sound source to move across the ceiling, from one end of the room to the other. This was successfully accomplished by starting the tape with the amplifiers for both rear speakers turned all the way down. Under these conditions, the sound (not surprisingly) came from between the front two speakers. Then, as the rear amplifiers were gradually turned up, the sound source rose from between the front two speakers until it was directly overhead when all four channels were balanced. I then gradually decreased the gain of the front amplifiers upon which the sound source continued its course across the ceiling, eventually winding up between the two rear speakers when the front two were all the way down. This was, of course, a crude way of controlling the levels, but the effect was successfully demonstrated. It would be possible and undoubtedly superior to control the recording level while taping the sound to achieve more consistent results. This, I will do in the future.

In an effort to try to determine the extent to which room acoustics account for the above, I listened to the same airplane tapes played on four channels in an open grassy field. The effect, although diminished somewhat, was still present. If we had used matched speakers (which we didn't), such as those in my laboratory, I suspect that the effect would have been even better.

I normally refrain from reporting such subjective results, but since this phenomenon seems to be so obvious, I am making an exception in this case and reporting on this prior to conducting a series of more exhaustive scientific studies dealing with the effects of frequency, bandwidth, intensity, duration, and meaningfulness of the sounds used. The results of these studies will be presented at the 81st meeting of the Acoustical Society of America in April, this year.

I think that an explanation of the phenomenon would run very much like that given for why we hear mono material as coming from between two stereo speakers: the only single source of sound which could create the particular sensations at each ear that we sense from four surrounding speakers would have to be directly overhead. If it were directly below, the body would interfere with the sound in some different way. Nonetheless, this area looks like fertile ground for research, which is my bag, and may have some important practical applications, which may be yours. ■

WALTER G. JUNG

A Transistorized Noise Generator

A broad-band noise source can be a useful tool to the audio man. Here is an inexpensive one you can make.

IT is a commonly known fact that a zener diode can be quite a healthy source of noise when biased near its operating knee. In fact, several manufacturers offer such devices specially as white-noise sources, and characterize their output spectrum in terms of amplitude flatness and voltage level. But for non-critical applications where an inexpensive noise source is desired, a reverse-biased transistor base-emitter junction makes an excellent low-cost substitute.

The circuit of such a device is in *Figure 1*, which in addition to the zener (Q1) contains a voltage amplifier and emitter follower to drive a 600 ohm line. Q1 and Q2 form a self-stabilizing d.c. feedback network by biasing Q1's base-emitter junction through R2 which is fed from Q2's collector. This negative feedback tends to hold the operating current of Q1 constant. For a.c., the minute current variations in Q1 (noise currents) are amplified by Q2 and appear as large noise voltages at Q2's collector. This signal, after being buffered by Q3, constitutes the output signal.

This simple circuit is easy to duplicate, but some comments about the components will be helpful. The transistors used are by no means critical and are suggested because of their low cost and ready availability. Q1 is the main determinant of noise output and should be selected from the Q1-Q3 trio (or those on hand) for the best noise performance.

Walter G. Jung is a frequent contributor to db.

Q1 should operate at about 50-100 μA and put out at least 2 volts p-p of noise at Q2's collector. Some transistors may generate more noise, but it appears "spiky" or non-uniform. Ideally of course, white noise should exhibit a flat frequency spectrum. The 2N2925's tested in this circuit all appeared quite uniform with no apparent predomination of any frequency. If the user feels inclined, he may want to experiment with other types in the Q1 socket.

This little device can be constructed for about two dollars in component cost and will serve as a useful source of white noise for special effects, amplifier testing or other applications. Although its output is uncalibrated, its economy and ease of implementation should make it attractive for many applications. ■

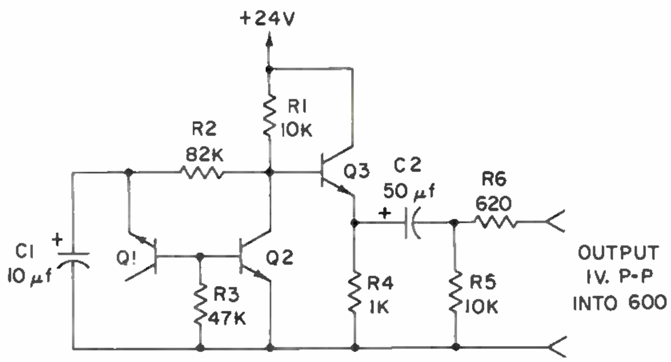

Figure 1. The noise generator described in the text. Note: Q1, Q2, Q3 = 2N2925; all resistors are 1/2 watt, ± 10%.

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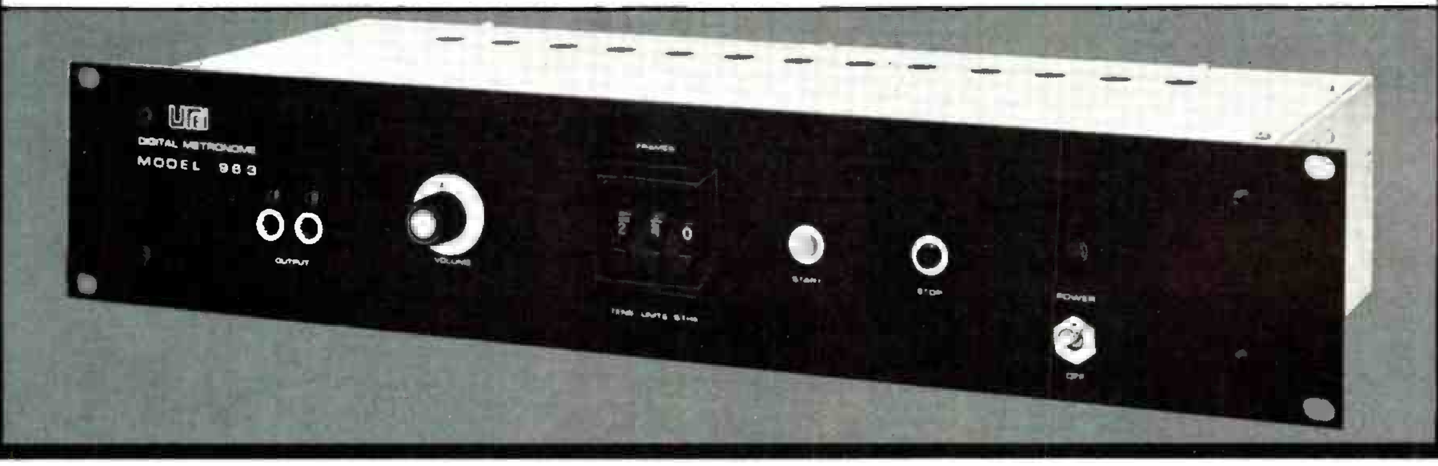

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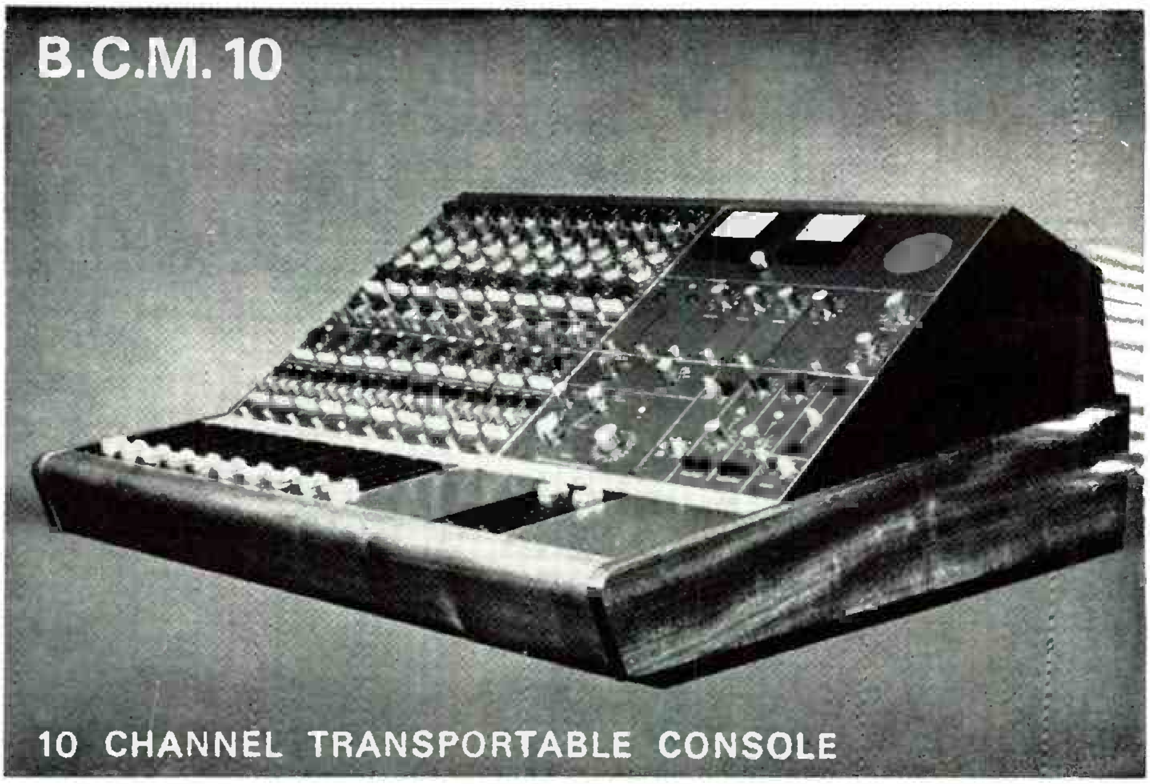

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PEOPLE, PLACES, HAPPENINGS


Chase

• Leslie H. Chase has joined Automated Processes' engineering staff. Les is well known throughout the industry, most of his career has been with RCA where he made the first stereo Red Seal recordings, and subsequently served as facilities manager.

• Public concern with noise pollution has received increased attention by both government and private organizations. The latest is a Columbia University laboratory under the School of Public Health and Administrative Medicine, Noise Reserach Unit which is headed by Professor Paul N. Borsky. Its aim is to discover detailed findings that will provide a basis for establishing administrative criteria of allowable maximum noise exposures. Samples will be obtained from residents in their homes as part of a regular community study. Details will be collected on personal variables and attitudinal and experience differences under different noise environments. The laboratory is located in suburban Long Island, N.Y. and is being conducted under an NASA grant.

• The new president of the Audio Engineering Society is George W. Bartlett, vice president for engineering of the NAB. This is the first time in the 25 year old Society that a broadcast industry representative has been made an elected official. Mr. Bartlett served for nine years as chief engineer at WDNC in Durham, N.C.

• Bernard Wise, president of CCA Electronics Corp. announce that it had completed the acquisition of radio station WABY, Albany, N.Y. FCC approval was granted to complete the acquisition as announced earlier. WABY will operate as a separate broadcast subsidiary with its own management.

• Robert Berkovitz has joined Dolby Laboratories, New York and London, as head of advertising and information. He brings a wide background to this position having recently been with Acoustic Research Inc. in a similar capacity and having served with Allied Radio Corp. where he was responsible for the development, design, and marketing of the company's branded products. He has also served in various capacities with Dynaco and Jensen.

• One of the newest and most up-to-date studios in London lead by Beatles producer George Martin has just added 18 more Dolby noise reduction units for their second 16-track recorder. The A.I.R. Studios is claimed to be "built by producers for producers" and boasts a full range of recording equipment and film dubbing facilities.

• Walter Goodman, president of the Institute of High Fidelity, Inc. has announced that the next IHF suburban show will be held in Palo Alto, California from April 1 to 4. Based on the success of previous suburban shows held in Westbury, N.Y. and Newton, Mass. it was decided to add another show on the west coast. Dealers will be welcome during the show, but a special dealer day has been set aside on April 1 from 2 to 6 p.m.

• Murray Goldman has been appointed sales manager, custom tape duplicating division according to an announcement by Joseph Tushinsky, president of Superscope, Inc. Mr. Goldman had been media director and director of marketing research during the past eight years with Superscope. The new division is involved in the production of cassette tapes tailored to individual business and industrial needs. A special series or research and educational tapes is being prepared by this organization. Raymond L. Jacobs will head the new Superscope Audio Visual Services, offering a new audio visual concept for sales, industrial and commercial training programs. This group will offer a complete program for training using audio-visual concepts and the sale and leasing of all hardware and software packaging facilities necessary the presentation. This unique service will be offered to all business, industry, religious groups, government agencies, civic agencies, medical institutions, and schools.

• CBS Laboratories confirmed that the United States Patent Office has granted a patent to Dr. Peter C. Goldmark and Dr. William E. Glenn, Jr., for a method of electronic color photography using black-and-white film. The patent describes a unique portable camera which makes it possible for the non-professional to take either still or moving pictures in full color. Up to 18,000 stills can be recorded on one reel of film with the system, which is the latest advance in video cassette technology for playing recorded programs over ordinary television sets. Dr. Goldmark emphasized that the system as described in the patent just issued is still in the research stage at CBS Laboratories, and that developments for production design and market feasibility studies still have to be undertaken.

"This new method of electronic photography employs the latest advances in electronics, in photography, and in television signal processing techniques," he said.

Dr. Goldmark emphasized that the system could be operated without any electric power. Color coding is accomplished through a unique arrangement of fixed optical filters placed directly behind the camera lens. Such a system would be equipped with a shutter lever for still pictures and a crank for moving film. When played through an EVR system over a television set, the EVR player retrieves the optical coding on the film electronically and registers them for color playback.


Grossberg

• Martin Audio Co., New York City manufacturing and sales firm servicing the professional audio field has announced the appointment of Larry Grossberg as vice president in charge of sales. Mr. Grossberg has had over sixteen years experience in the electronic industry. He was formerly general sales manager for the Sonocraft Corp. in New York. He follows a tradition in the electronic industry; his father, Louis H. Grossberg, an industry pioneer, founded Solo Electronics and co-founded Milo Electronics.

NEW PRO-4AA

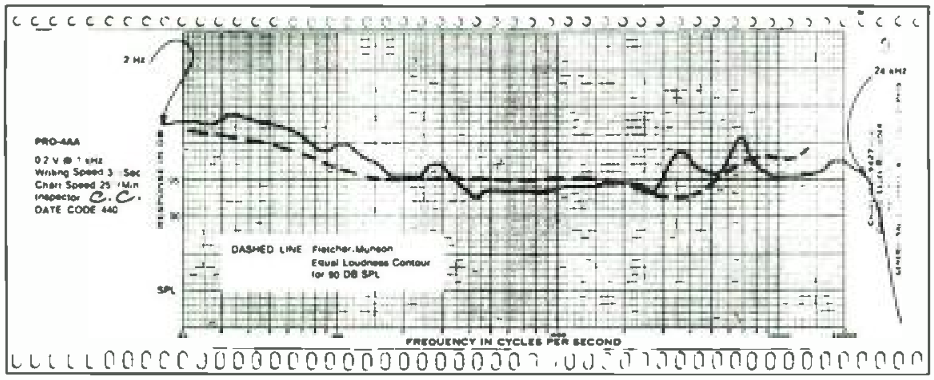
The good sound you monitor is the Sound of KOSS

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 Valtorta 21 20127 Milan Italy

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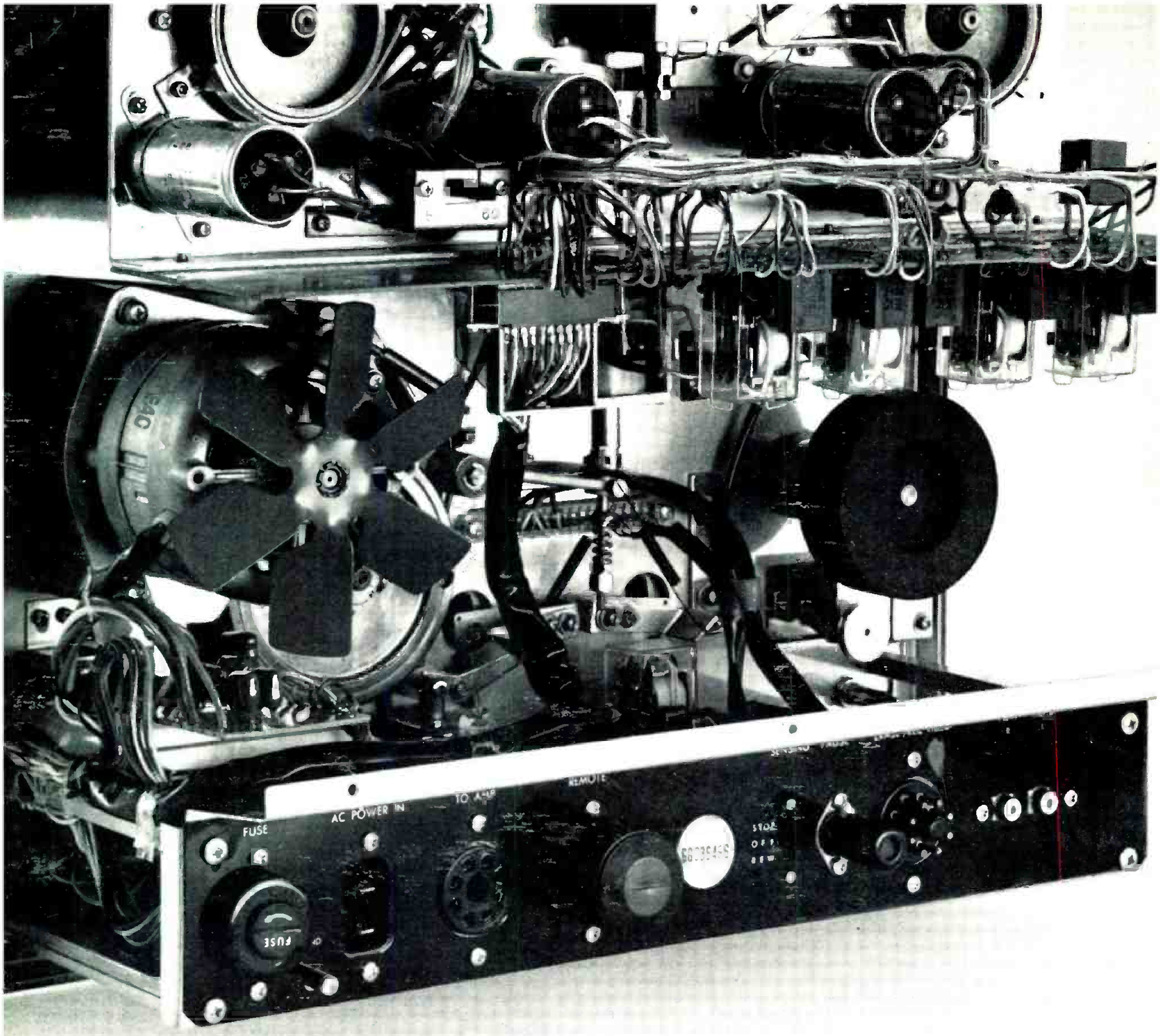

BACK TALK

When it comes to building sound equipment from the inside out, you could call us the component company. You see, we're one of the few tape deck manufacturers who make all our own critical components – from heads to motors and most of the electronics. After all, who knows better than we do what it takes to make a TEAC?

For instance, our heads are hyperbolic, not conventionally rounded. This means more intimate tape contact, less tape tension, better sound reproduction. Hyperbolic heads are the shape of things to come – and the only kind we'd think of using.

Meanwhile, we still buy outside parts for certain purposes. The ones we buy, we buy because they're the best. The ones we make, we make because they're the best.

And most of the time, we've got it made.

TEAC

TEAC Corporation of America • 2000 Colorado Avenue • Santa Monica, California 90404

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