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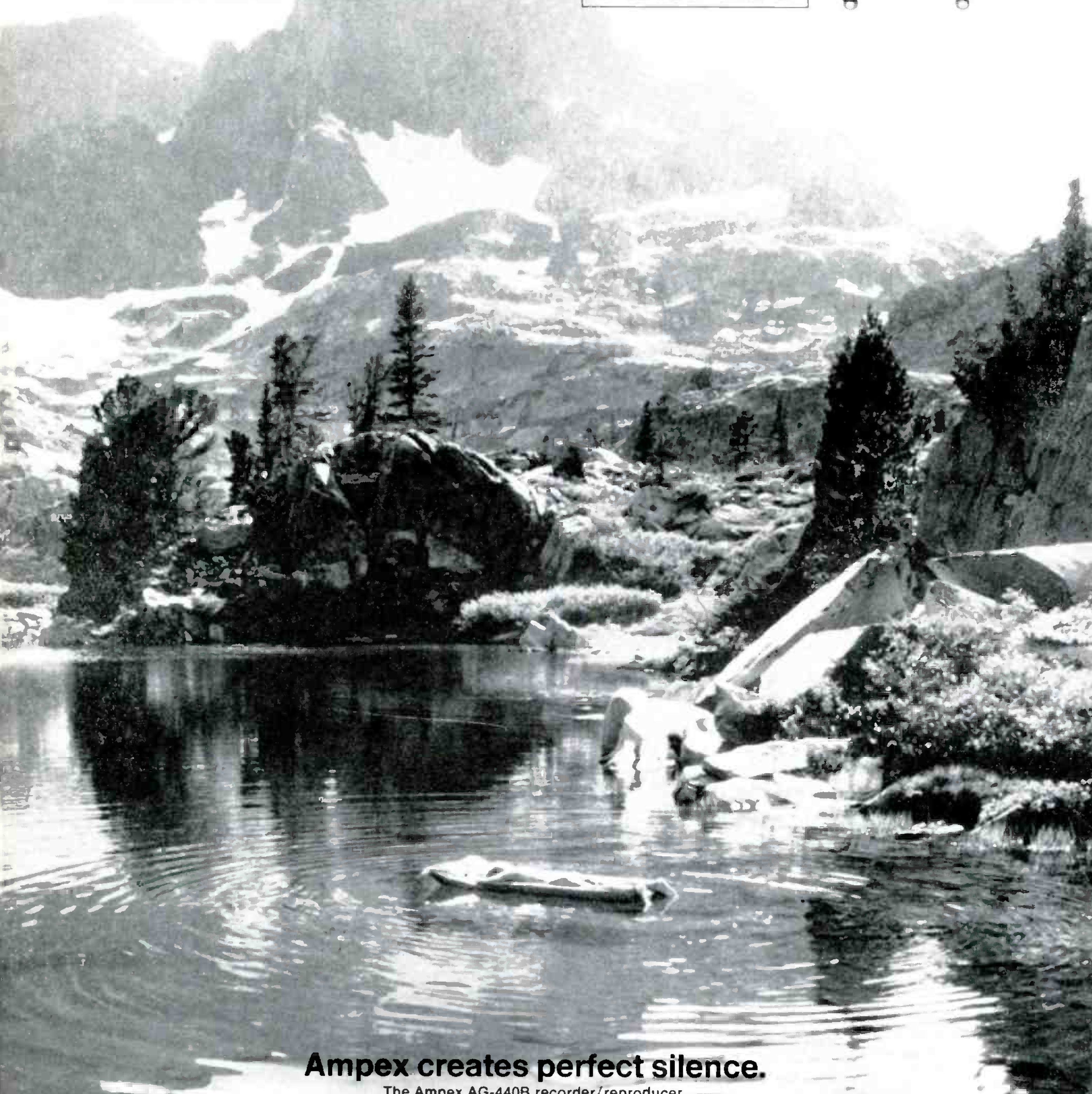

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Coming Next Month

A most vital subject to audio professionals everywhere is the topic of a multi-part article beginning next month. The author is W. Dixon Ward of the University of Minnesota and his subject is titled HEARING LOSS AND AUDIO-ENGINEERING. We do not wish to frighten you, but you may just reflect a bit when you learn the relationship between the volume levels sometimes found in monitoring rooms and irretrievable hearing loss. This is must reading for all audio pros.

Sidney Feldman himself an outstanding recording engineer reports on a meeting at which the topic was PHONOGRAPH RECORD COMPATIBILITY. He points out that a decade after the introduction of the stereo disc, the battle for compatibility still rages.

We will have a picture report on a DISC RECORDING SEMINAR Gotham Audio recently held in the lovely setting of Sterling Forest, N. Y.

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

About the Cover

A fish-eye control room view at Audiotek Recording Studios, Minneapolis, Minnesota. At the controls Dan Holmes, chief engineer. Audiotek uses such equipment as Scully, Dolby, UA, Langevin, and Lang, with a complement of Neumann and Sony microphones.

db

THE SOUND ENGINEERING MAGAZINE

APRIL 1970 • Volume 4, Number 4

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EDITORIAL BOARD OF REVIEW

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Letters

The Editor:
Mr. Smith's *A Phono Reproduction Preamplifier* (db Dec. 1969) deserves added comment.

The noise formula presented is valid for noise voltage from a finite source-resistance and clearly indicates that noise increases with source resistance. It is quite misleading as presented, however, since several very important factors are completely ignored.

It implies that noise is zero if source resistance is zero—quite so if the amplifier is noiseless. But real-life amplifiers are not noiseless and system noise is lowest when amplifier *and* source are optimized for the application. Mr. Smith gives sole credit to the low source resistance and fails to consider the intrinsic noise sources within the amplifier, such as 1/F or "flicker" noise.

Most cartridges are designed for high impedance with good reason—high output voltage. Certainly, thermal noise increases by the square-root of resistance, but so does the voltage! What better way to over-ride amplifier noise? Amplifier input impedance is not a problem since it is usually necessary to *add* resistance to properly terminate the cartridge.

Practical limits are determined by cable lengths and the capacitive loading of the source at high frequencies. Is there real disadvantage in locating the preamp within a few feet of the cartridge? It is interesting to note that Ortofon furnishes a step-up transformer with their cartridge. If the purpose of the article is to promote physical distance between cartridge and preamp, then it should be so stated.

The noise formula defines thermal noise from a resistance, not an inductance. How then, can we use the *impedance* of the cartridge? It's the *resistance* of the source, not the impedance, that determines the noise. If you rewind the coil to be non-inductive, the impedance will approach zero but the noise will remain the same—because the *resistance* is the same.

Mr. Smith refers to "response—to 300 KHz." (without equalization) and "down below 10Hz." Honestly, now, who can hear 300 KHz? What use is unequalized response in a phono preamp? And what "program" material is down at 10Hz? RIAA Response is quoted, yet RIAA also recommends rolloff below 30Hz. to reject sub-audible response.

". . .the true test of low-noise performance is to. . .listen to the surface noise. . ." I couldn't agree more! But previously he says "conventional" noise measurements are meaningless: I couldn't agree less! If phono preamps are designed to standards, then standard noise measurements made on them *are* conventional.

We are soon to release a new preamp with unweighted noise more than 80dB below program level, RIAA response within 0.5dB and thd below 0.035 per cent. Anyone who knows how can measure or compare. Mr. Smith's article would have been immensely more valuable had he included industry-recognizable performance specifications. As it now stands, it appears more a promo for Ortofon and Spectra-Sonics than a serious professional presentation. If the product is good, then why not show how good?

Roger K. Odom
Engineering Mgr.
Spectra-Sonics Corp.
Ogden, Utah 84404

Mr. Smith responds:
I would like to make the following clarifications concerning my article in the December 1969 issue of db Magazine. I believe my statement on page 25 concerning the 2-ohm cartridge impedance and the 3 ohm amplifier impedance was misleading. I had meant to say that the 2-ohm cartridge impedance and the 3-ohm output impedance of the 104 preamplifier card will permit long connecting lines without hum pickup and high frequency attenuation. Actually, the 2-ohm cartridge impedance works into a 600-ohm input impedance on the 104 card. The signal voltage is developed across the 600-ohm input impedance while the noise voltage is developed across the 2-ohm cartridge impedance.

The noise formula presented is for noise voltage from a finite source resistance. The Ortofon SL-15 cartridge is of the moving-coil type and is resistive in its impedance characteristic within the audio spectrum. While it is true that a transformer inside the cartridge will increase the cartridge output voltage to improve the signal-to-noise performance of the preamplifier input stage, the detrimental effects on ambient pickup, distortion, phase shift, and high frequency attenuation produced by the transformer must be considered. The prime intent in the design of the preamplifier described in my article was not only to improve signal-to-noise performance but to drastically reduce distortion. The low cartridge and amplifier output impedance permit long cable runs providing an added feature.

(*Continued on page 57*)

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ASST. CIRCULATION MGR.

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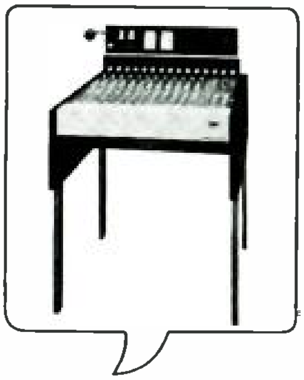

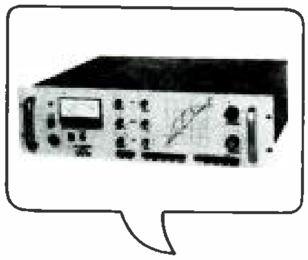

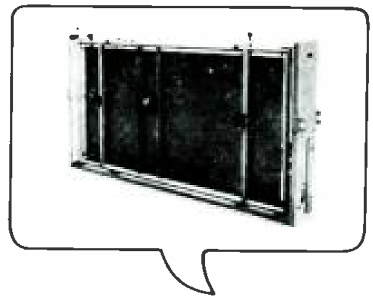

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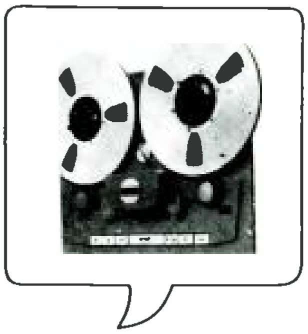

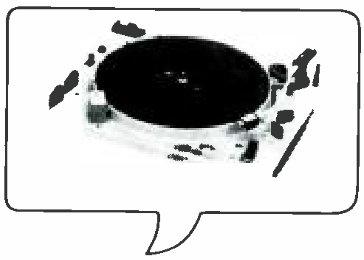

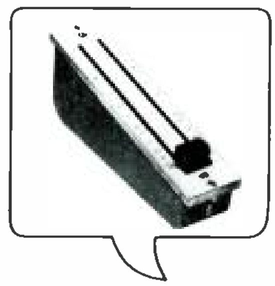

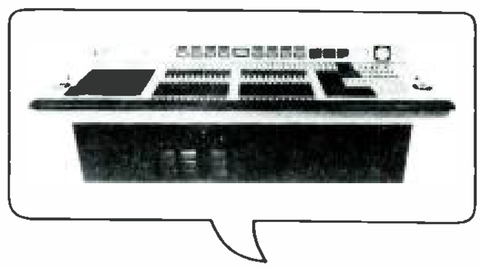

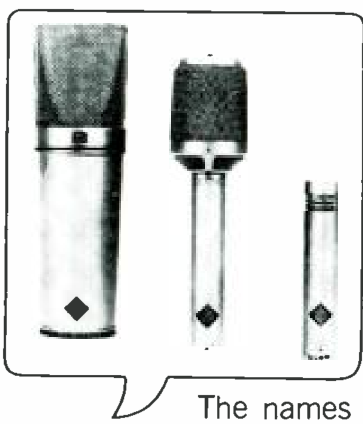

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

Filters in environmental equalizers

● The methods of equalization described up to now have dealt with the shaping of the extremes of the audio frequency spectrum. We have mentioned in general the role of narrow-band or high-Q filters. In now reviewing basic methods of equalization, we shall omit sharp cut-off or band stop filters, since they are more complex than ordinary single-frequency tank circuits we are to discuss this month.

We started discussion of the topic *equalization in sound-reinforcement systems* with the premise that we are dealing with an existing sound system that is permanently installed, and which is in need of response modificgtion in order to achieve higher reproduce levels before acoustical feedback sets in. In the case of sound reproduce systems we are seeking sound more faithful to the original. We have acoustically measured the frequency response and plotted the curve. Now we must design equalization for the amplifier system which should have a mirror image of the actual response of the system. The combination of both should approximate flat response. The advantage of such an approach is that you may end up needing fewer narrow band filters (which are harder to design and more costly) and make more use of standard tank circuit dip filters in equalizing gentle rises and dips in response. Inductors for filters should be selected according to the shape of the curve being designed for. This is suggested for purely economic reasons, because chokes with air cores are many times less costly than the torroidal inductors required for sharper filters.

In order to achieve more accurate equalization you should take as many readings as possible near frequencies exhibiting peaks when acoustically measuring a room. More accurately drawn curves will reveal more precisely what quality filter is required to remove the peak. Automatic plotting machines sweep through the whole spectrum, plotting an infinite number of points along the response curve. When measuring response using a manually-controlled oscillator, finding and marking the 3 dB points of every peak will indicate the Q of the peak and consequently that of filter for all practical purposes finding the 3 dB points means finding the frequencies two on both sides of the peak by adjusting the oscillator frequency until the reading is

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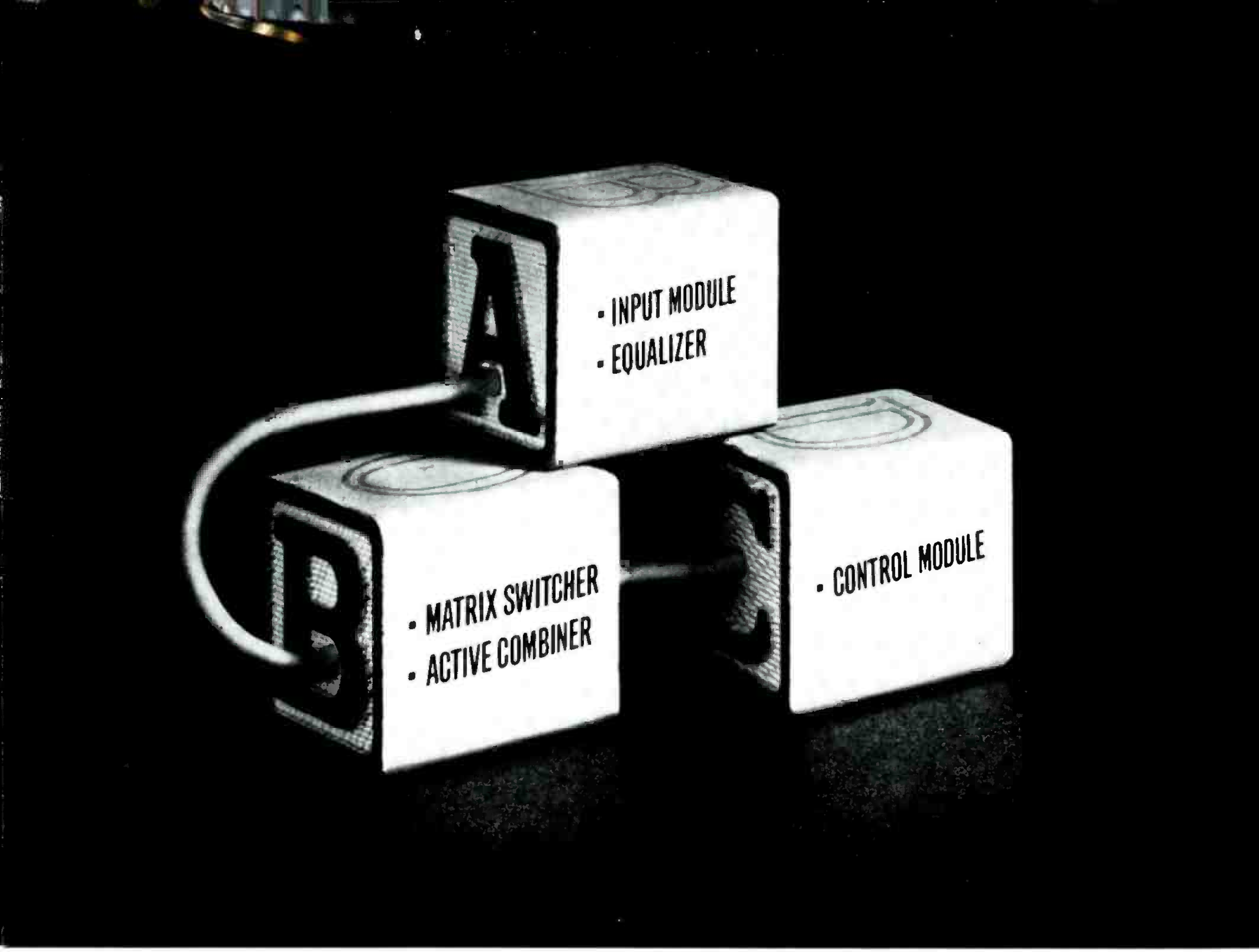

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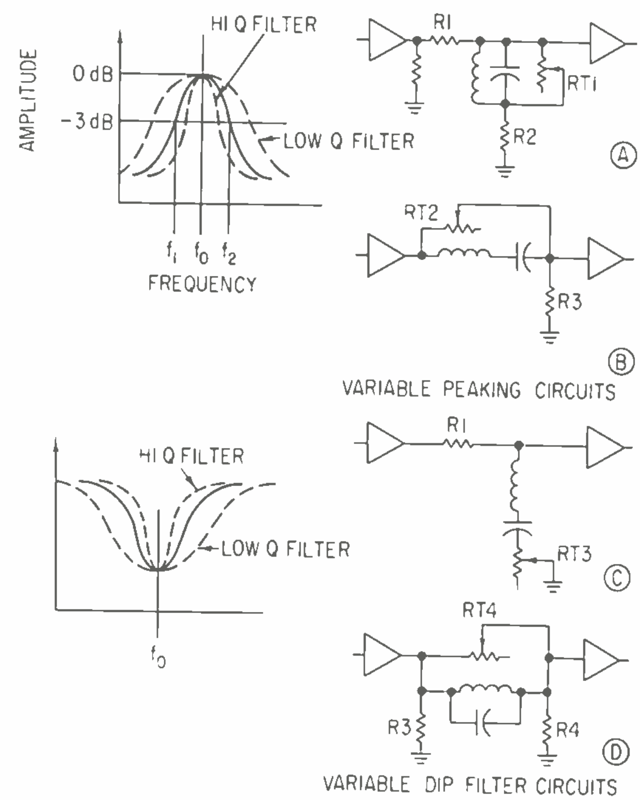

Figure 1. Variable dip filter circuits.

3 dB lower than the reading at the peak frequency. The difference in frequencies of the two 3 dB points is called the bandwidth of the peak (or the filter). The smaller the difference the smaller the bandwidth and the higher the Q.

Let us first discuss basic tank circuits for medium Q filters. A tank circuit consists of a capacitor and inductor or choke. The distinguishing property of a tank circuit is its ability to resonate at one certain frequency. This ability to single out one frequency out of the band is for equalization. Because a tank circuit can offer either very high or very low conductance at the resonant frequency (depending if it is in a series or parallel circuit) it can be used for either peak or notch filters. The height of the peak (or notch) can be precisely controlled by using variable resistance in shunt or in series with the circuit. Since a tank circuit consists of an inductor not easily adjustable to vary its inductance, it is suggested that capacitor(s) be adjusted instead so as to produce the required resonant frequency using stock chokes, rather than trying to obtain an odd value inductor.

For example, let us consider a peak in response at 2500 Hz which we would like to filter out. At hand we happen to have an assortment of chokes. Inductances range from 1 millihenry to 1 henry. In order to determine the required capeitance for the resonance of 2500 Hz we use the formula

$$f_o = \frac{1}{2\pi\sqrt{LC}}$$

where L is in henries and C is in farads

We solve the equation for C and substitute values

$$C = \frac{1}{4\pi^2 f_o^2 L};\ C = \frac{1}{4 \times 3.14^2 \times 2500^2 \times 0.01}$$

$$C = \frac{1}{2.46 \times 10^6} \text{ farads or } C = 0.40\ \mu F$$

Since 0.4µF is not a standard capacitor we parallel two capacitors in order to obtain the desired capacitance. Two values can be a combination of 0.22µF and 0.18µF or 0.33 µF in parallel with 0.068 µF which will be well within the range we are aiming for. Anyone who knows the prices for special inductors will appreciate the saving in using two capacitors instead of a special coil.

If the inductor in the tank circuit happens to be a plain coil that is wound on the ar air-core bobbin, you can expect a wide peak. If you are looking for a narrower peak or dip you will have to use a torroidal or powdered iron-core coil form industor. The circuit required to form a peak or notch in response is shown in FIGURE 1.

The four circuits shown have variable resistors to adjust the amplitude of the peak produced by the tank circuit. Let us analyze each circuit separately. Circuit (A) has a tank circuit in shunt with the input to the output amplifier. At any frequency other than resonant, a tank circuit is almost the same as a short circuit. Resistors R1 and R2 are as L pad, producing a certain level drop

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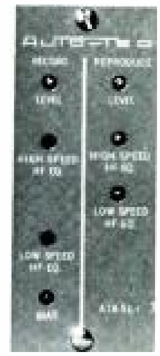

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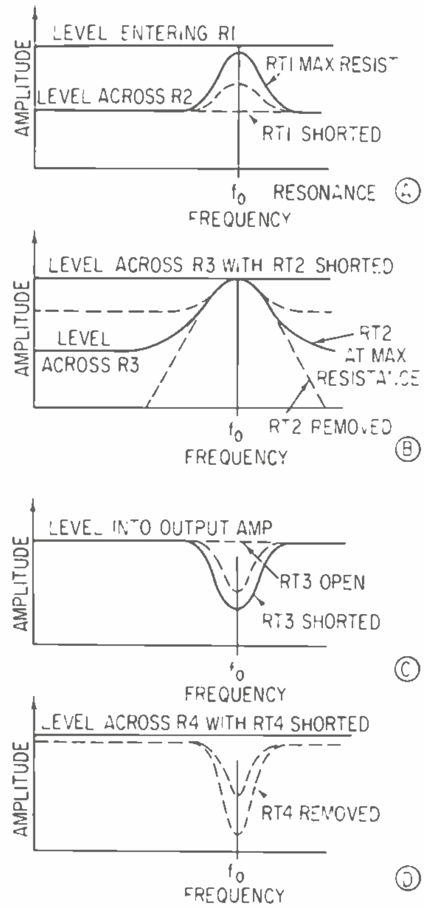


Figure 2. The results of L-pad manipulation.

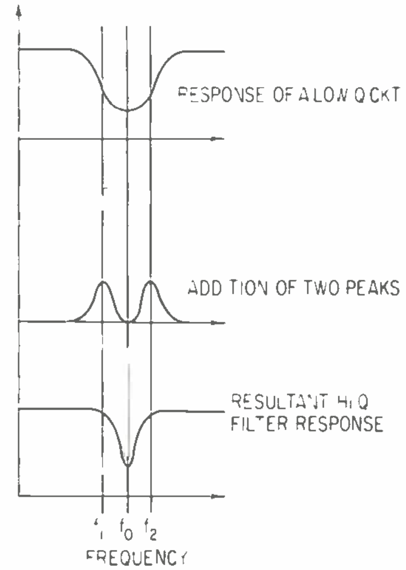


Figure 3. This is one of the approaches that can be used to obtain hi Q response.

At resonance, the impedance of the tank circuit becomes very high and produces a peak in response equal to the loss produced by the resistive L pad at other frequencies. Introduction of the rheostat Rt 1 negates the action of the tank circuit. The smaller the resistance of Rt 1 the less pronounced the peak.

In circuit (B) the tank circuit is a series one and presents very high series resistance at all frequencies other than resonance. Being a part of the L pad, the tank circuit produces very high insertion loss for all non-resonant frequencies, and the consequent voltage drop across R3 is very small. Using Rt 2, the action of the L pad is limited to the maximum loss produced by the L pad when Rt 2 is set for maximum resistance. Varying the shunt rheostat changes the loss at all other frequencies, but leaves the peak level of the resonance at the same level. This is demonstrated in Figure 2.

In the dip filters of Figure 1 circuit (C) is similar to circuit (A) except that the tank circuit affects loss of the L pad only at the resonant frequency. Series rheostat Rt 3 introduces finite resistance in series with the tank circuit. At the resonance, it softens the action of the tank circuit.

Circuit (D) is similar to circuit (B) but is useful because it varies the dip at the resonance using Rt 4, without introducing excessive loss into the circuit. This is contrary to circuit (C) where there always exists loss (because of R1 in series with the input impedance of the output amplifier introduces level drop) but circuit D presents less insertion loss. However, a lot depends on the amount of filtering it is necessary to apply and the impedances of the amplifiers. In addition, the Q of the coils may affect the loss because higher Q coils have smaller d.c. resistance.

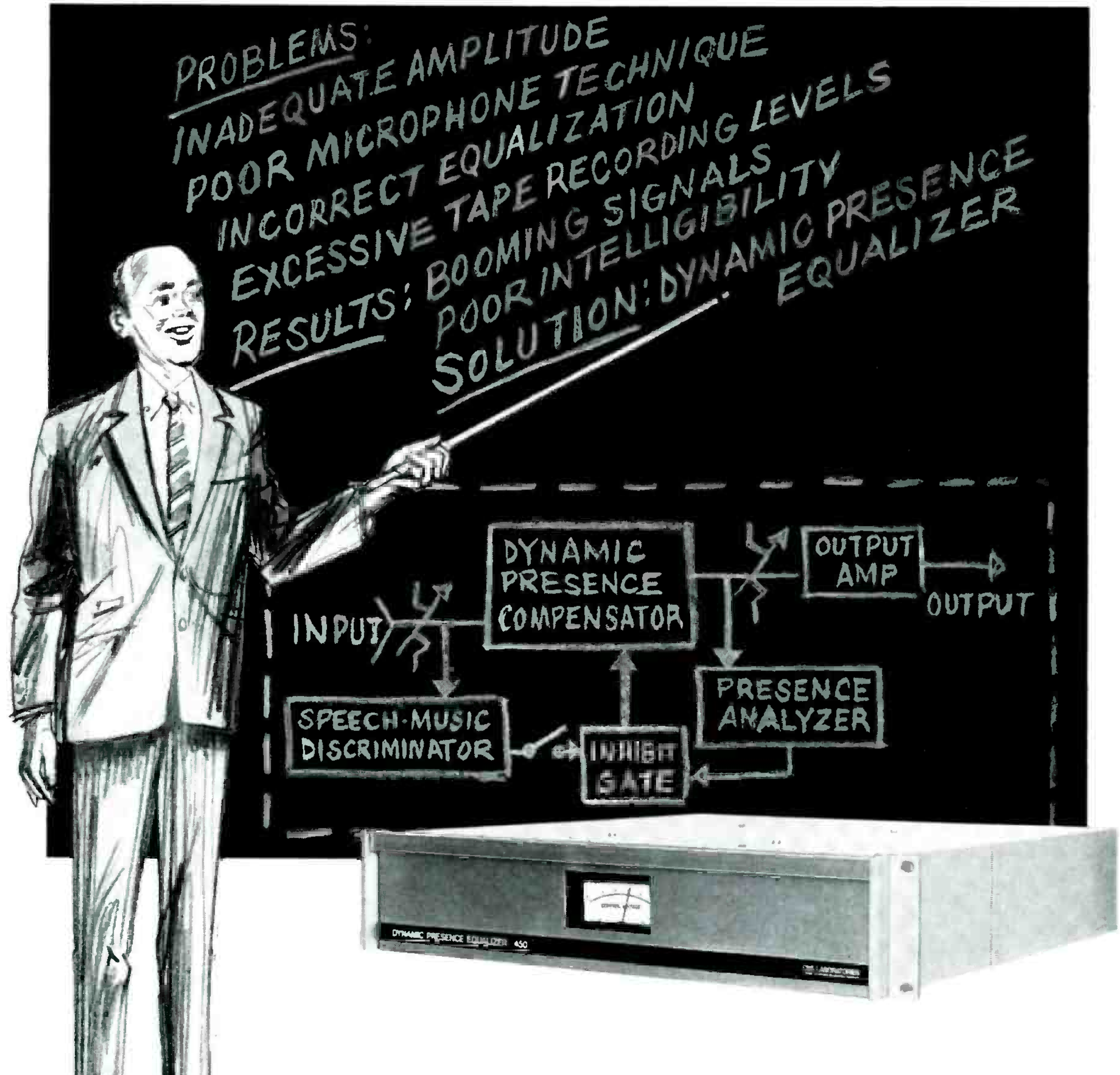
It is sometimes necessary to have a broad peak equalization capability. Insertion of fixed or variable resistance in series with the inductor would spoil the Q of the tank circuit and broaden the peak. If the job calls for a sharper peak than one possible with the highest Q inductor, we must look to new methods of sharpening the characteristics of the filter. One of the established methods of obtaining sharper filter characteristics is to use *several* tank circuits. The basic principle of this method is graphically demonstrated in Figure 3 but no detailed discussion is planned because (aside from the fact that this method is costly) components to do the job properly have to be carefully calculated and matched. Remember, values of inductors and capacitors almost never come out to be an easy-to-get standard value. There are so many modern approaches and solutions to this problem (using operational amplifiers and active filter circuits) that it doesn't pay to waste time rehashing circuits of yesterday. Next month we shall start a review of active filters using solid-state technology as the basis. ■

The Feedback Loop

ARNOLD SCHWARTZ

• It is an interesting comment on "entertainment technology" that when new media first appear they seem destined to drastically reduce the role played by existing forms, and possibly even to drive the latter off the market completely. Often the results are quite different from what is first predicted. The record industry appeared doomed by the one-two punch of radio and the depression. Record sales dropped off from a high of sixty-million dollars in 1927 to a low of six-million dollars in 1933. Nevertheless, records survived, and by 1947, the year before the long-playing record was introduced, sales were two-hundred-twenty-four million dollars. Current U. S. record sales are well in excess of one billion dollars. Radio in its turn appeared to be seriously threatened by television, but is now a booming industry; the number of stations has increased from less than three thousand in 1950 to over six-thousand today. Television also seemed to pose a serious threat to the local movie theaters, and many folded under the initial pressure of the ubiquitous home television set. The ghosts of many former movie theaters appear now as supermarkets, skating rinks, and decaying buildings. Despite the initial impact of t. v. local movie houses now show vigorous activity, and not only have old theaters revived, but many new theaters are being built.

All of this is not to say that each medium has been unaffected by the wave of competing forms. Radio programming for example, bears little resemblance to pre t.v. days. *Sexploitation* type films no doubt owe much of their existence to the strong competition of the television set. It seems



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that the establishment of a competing form has a strong influence on the established form, but it creates its own audience while the older medium continues to flourish.

In somewhat the same way tape recording technology appeared destined not only to replace the disc as a recording medium, but with the introduction of commercially-recorded cartridges and casettes also appeared slated to replace the disc as a play-back medium as well. Current commercially - recorded tape sales are estimated to be over a quarter of a billion dollars, but as we have seen, the disc continues to prosper and grow.

It is interesting to compare tape and disc from an engineering point of view to see just where we stand.

Disc recording, which is another way of saying mechanical recording, was invented by Edison in 1877—almost one hundred years ago. In 1887 Emil Berliner invented the disc record and this invention embodies the basic form of today's phonograph record. As things go, the disc has been around a very long time. The more recent technology of tape recording encompasses a vast area: from low-speed cassettes for audio playback to video recording. In the audio field, for reasons that are too obvious to discuss, tape is the outstandingly superior medium for recording. In addition, from the professional point of view tape is clearly superior for such playback applications as fast action radio formats that use cartridge machines, and automation. Does the disc present any advantages to the audio professional? In at least one area the answer is yes; the disc is the superior quality mass-produced playback medium.

This last statement came to mind recently when I came across an article in the May 1969 issue of *Broadcast/Management Engineering* which contained an interview with Everett B. Cobb of station KNEV-FM in Reno. Mr. Cobb states among other things that, "A disc, being a mechanical device, is subject to distortion, wear, and other deficiencies, even using the best professional turntables and pickups. Professionally duplicated tape, on the other hand, is acknowledged the finest source of mass-produced quality sound available." Instead of citing bandwidth, distortion, and signal-to-noise measurements to prove the superiority of the 33-1/3 rpm record over the 7½ ips recorded tape, I would rather take a somewhat different approach in comparing tape and disc.

Which medium has the greatest capacity for storing information? The greater the storage capacity of the medium the greater the capability for producing quality sound. On this basis we can compare two-track quarter-inch tape at 15 in./sec. to the 33-1/3 r.p.m. stereo disc. As a first approximation, we will say that both the tape and disc have the same bandwidth and signal-to-noise ratio. The maximum signal level in each case would be that level where the distortion reaches a predetermined amount. The medium which requires the least area and volume of recording medium to store the same musical selection has the greater information storage capacity. First we calculate the area required for one second of playing time.

Tape:

$(15 \text{ in. sec.}) \times (1\frac{1}{4} \text{ inch}) = 3.75$ square-inch area for one second playing time

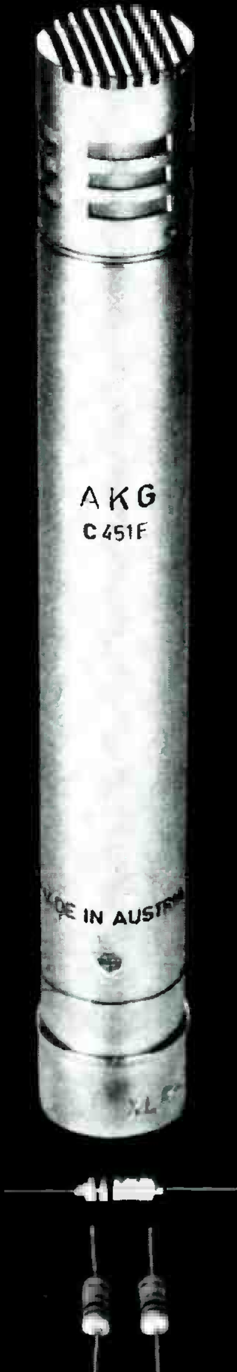
Disc:

assume 250 lines per inch at 11-inch diameter

$$\frac{(33\text{-}1/3) \times (\pi) \times (11 \text{ in.})}{(60) \times (250)} = 0.077 \text{ sq. in. area for one second playing time}$$

It takes fifty times more tape area than disc area to play one second of music. Now let us see what volume of recorded material it takes for one second of playing time.

AKG C-451E Microphone System



the common-sense approach to phantom powering.

Just one or two resistors! Simple, isn't it?

- No AC power supplies required
- No DC power supplies
- No "central" power supplies
- No special "cards"
- Operates off any DC voltage between 7.5 and 52v (including the standard 24v available in most equipment).
- Stabilizes its own operating voltage, to maintain constant output level.
- Optimum polarization voltage, for best signal-to-noise ratio.

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AKG-1 30

THE SIMPLICITY OF SIMPLEXING

The AKG C-451E may be powered at the cost of one or two precision resistors: No separate AC or DC power supplies required nor special "cards" or "central" power supplies at additional cost.

The microphone preamplifier requires as little as 7.5v DC and may be operated directly off the *standard* 24v B+ supply available in your equipment (or any other voltage between 7.5 and 52v). The current consumption is only 2.8 mA.

Other features unique with the AKG system are:

A) Stabilized operating voltage: The DC supply voltage to the microphone is not required to be particularly well regulated nor is it rigidly tied to a specific voltage. In fact, it may vary by ± 15% since the C-451E preamplifier will stabilize the operating voltage. There is no limit to the number of microphones to be powered off your console.

B) Constant 60 volts polarization voltage: 60 volts is the optimum polarization voltage for highest performance standards, specifically sensitivity; resulting in more gain without increase in noise level and better signal-to-noise ratio. The C-451E supply voltage is not simultaneously the polarization voltage (too low). The microphone preamplifier provides a constant 60 volts polarization voltage and fluctuation in the supply voltage will not change the output level of the microphone.

There are no short cuts in the AKG C-451E circuitry!

HOW DOES IT SOUND?

Interestingly enough, its pick-up characteristics are being compared to the quality previously obtained only with large diaphragm condenser microphones.

The newly developed CK-1 capsule incorporates a metallic alloy diaphragm (similar to the diaphragm material used in measuring microphones) and is absolutely smooth between 30-18,000 Hz with unequaled transient response characteristics and wide dynamic range.

The polar pattern is a true cardioid at all frequencies with linear off-axis acceptance and a front-to-back discrimination of 20-30 dB over the entire range.

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Circle 23 on Reader Service Card

Tape:
(3.75 in.²) × (0.001-in. thick) = 0.0038 cubic inches for one second of playing time

Disc:
assume a 0.05-inch thick disc, and since there are two sides to the disc we allow 0.025 for each side
(0.077 inch²) × (0.025 inch thick) = 0.0019 cubic inches for one second of playing time

It takes almost twice the tape volume as it does disc volume to play one second of music. If we take into account the exact signal-to-noise ratio and bandwidth the disc would be shown in a somewhat more favorable light. Mr. Cobb was actually talking about 7½ in./sec. tapes (not the 15 in./sec. tape used in our comparison above) which will degrade the tape performance by a 3 dB reduction in signal-to-noise ratio, and decrease the bandwidth. From this comparison we can conclude that the disc has the greater information storage capacity—and hence has the capability of producing higher-quality audio.

There are more direct approaches to evaluating the tape and disc which would yield similar results. Today tape recording is in the forefront of modern technology in many areas. However, in the field of mass-produced high-quality audio playback it is still the disc, after an amazing ninety-three year history, that comes out on top. ■

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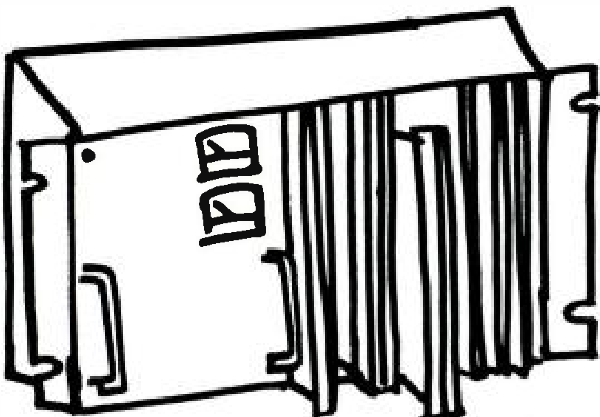
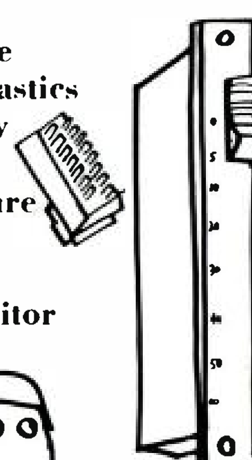
Down with dB's!



Harvey's has a new motto: "Down with dB's." Of course we're referring to those unwanted dB's of hiss, noise and the like.

And to help fight the battle against those evil dB's, we now carry a rather large arsenal of noise-reduction equipment.

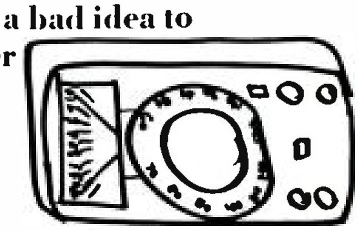
For example, no well-equipped studio is complete without a Dolby noise-reduction system (Harvey's is one of the few places you can get one). The Dolby system is compatible with any tape recorder, no matter how many tracks it has.



Speaking of well-equipped studios, many of them have installed banks of the new Penny and Giles faders (available only at Harvey's). The faders (designed with conductive plastics technology) are super-narrow, super smooth, and extremely quiet (there's minimal extraneous electrical noise).

Harvey's also has a compact sound-level meter to measure the noise level in a studio or room (take it home and try it out on your wife, if you like).

It's extremely useful for adjusting the level of, say, monitor speakers in separate studios. And it isn't a bad idea to take a measurement with this Scott meter when you're in the middle of a critical recording session. That way, you can duplicate the exact monitor level again, should you ever need to.



Of course, the Scott meter can also be used to measure dB's of noise in a room.

As Harvey's always says, "a good soldier always knows his enemy."

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Circle 24 on Reader Service Card

The Sync Track

JOHN M. WORAM

●db readers certainly don't have to be told of the many effects of exposure to sound. At one extreme there is soft harmony to sooth savage breasts—at another, nervous tension and eventually deafness. In between, lie all the emotions known to man. According to the ads one reads, we have discovered, and recorded, sounds for working, eating, making love, and anything else you may want to do with accompaniment.

Further afield, the scientific community has begun accumulating data which may prove that in addition to influencing human behavior, exposure to sound affects the growth rate of plant life. In a series of experiments, corn grown under continuous exposure to sound exhibited a significantly higher rate of growth than a control crop grown in a quieter environment. In another experiment, polygraphs have recorded what would appear to be emotional reactions in plants that were exposed to situations in which their physical well-being was threatened. These experiments, which sound like pure science fiction, have been described in reliable journals. (see the references).

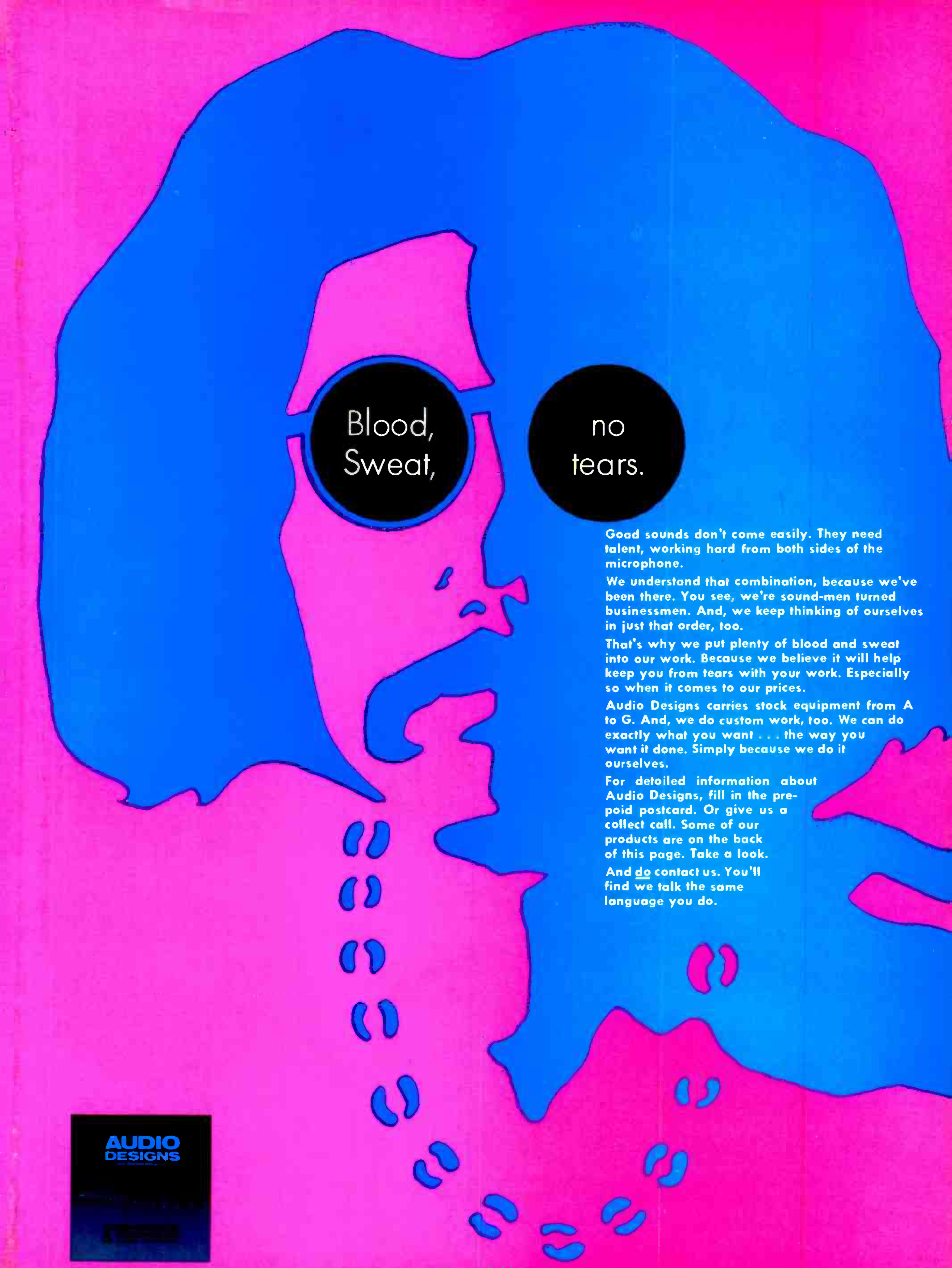
Although at first it may seem that these experiments—though fascinating—have no particular significance to the recording industry, careful thought will reveal implications of staggering proportions.

To cite one obvious application of this new knowledge, consider the now time-consuming task of mixing down a multi-track tape. One of the reasons this takes so long is that engineer and producer are rarely confident that they have finally made the ideal mix-down. Often, after long hours of mixing, a theoretically impartial third party will be asked which mix he prefers. I say theoretically impartial, since we know that no one is truly impartial. A whole complex of variables such as environment, education, sex, (the noun, and sometimes the verb) mood, health, and so on influence all our human tastes. Obviously then, no one is truly impartial, and the danger is that the third party will choose the improper mix—one that does not have the best commercial potential. Due to listener fatigue, the engineer/producer team will not trust their own judgements and a wrong decision may easily be made.

Now, to return to the plant experiments mentioned above. Apparently 1. plant growth is influenced by sound waves. 2. plants are capable of reacting to specific types of sound.

Yet what could be more impartial than a plant? No plant has ever suffered from a broken home, chronic unemployment, bussing, commuter tie-ups, and any of the other tribulations of the human experience. Presuming a healthy root structure (easily accomplished under lab conditions), a plant may safely be regarded as an absolutely impartial source of opinion, now that science has discovered a reliable read-out system.

The next step is obvious. Before the final mix is decided upon, several alternative versions are played in different listening rooms. In each room, a plant is placed near the speaker. Plant height is carefully measured before playing the tape, and again after playback. The different growth rates during the playback cycle will accurately indicate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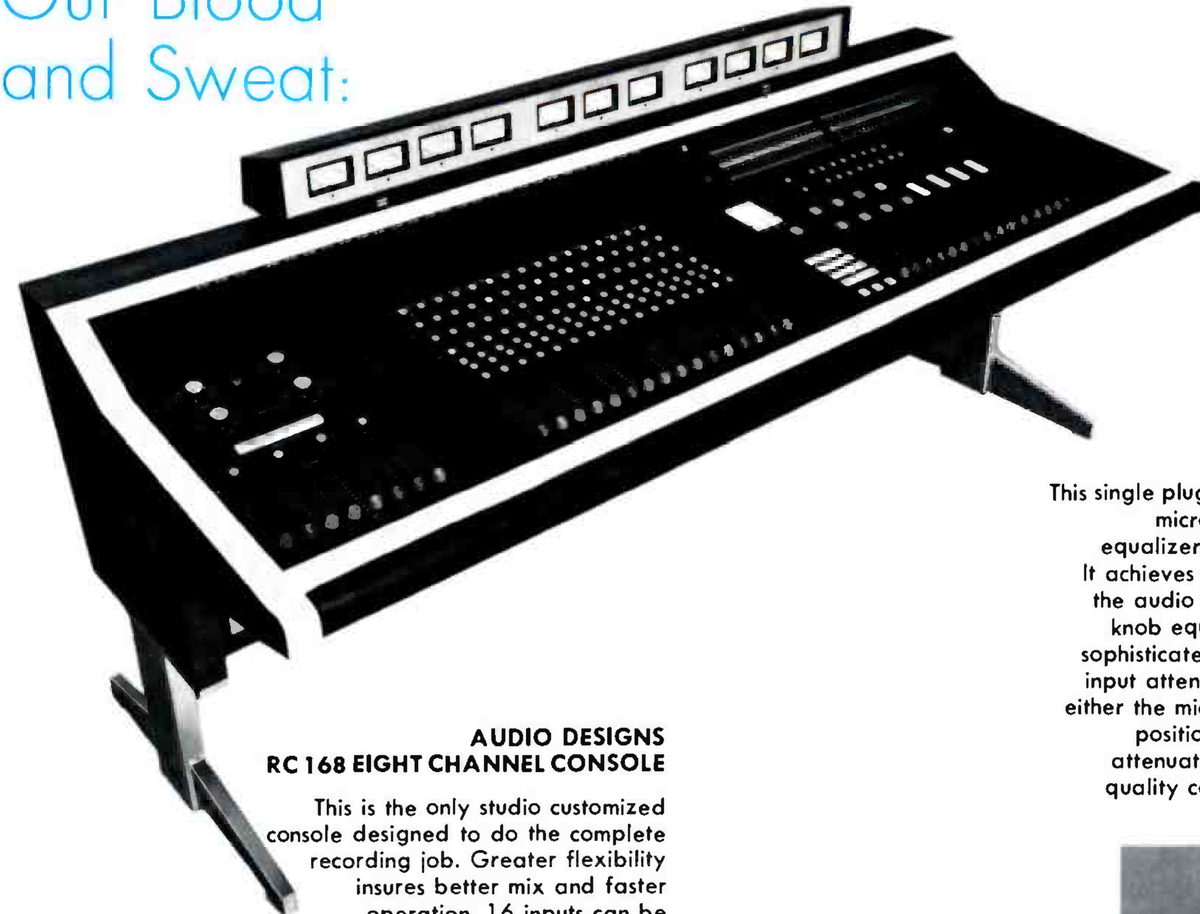

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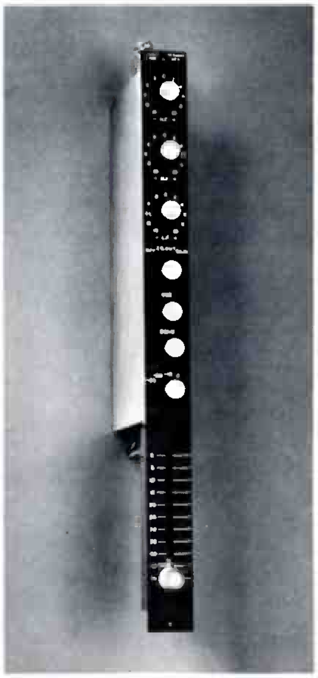

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which mix is preferred by the plant life. Since the plant has no comprehension of budget, artist preference, release dates and what not, its decision may safely be considered as absolutely impartial.

To insure maximum accuracy, certain precautions must be observed. First, of course, is plant suitability. There is a growing suspicion in learned circles that plants of the *Lycopersicon Esculentum* family may be notoriously tone deaf. Until further studies have been made, they should be avoided. At this time, indian corn seems to be eminently suitable for all-around use. Some strains exhibit a sensitivity to apparent loudness, which may be an asset in evaluating rock singles, especially where variations in limiter settings are being considered.

Unusual plants must be avoided at all cost. In one mis-guided experiment, a Venus-fly trap (*Dionnaea Muscipula*) was chosen to audit an electronic synthesizer album. Unfortunately, the tape deck had a severe flutter problem, and the plant became so dis-oriented that it began snapping at the monitor speaker, eventually electrocuting itself when it severed the speaker leads during a complex passage of modulated square waves.

Naturally, even the most sound-sensitive plant will not exhibit a measurable growth rate during the time it takes to play a top-100 type single. For that reason, the tape must be replayed continuously until a measurable change has been made. The most practical way to do this is with a continuous tape-loop system since this avoids the problem of rewinding. If you must fast rewind a tape, be sure the speaker is turned off. Many plants appear to be direction sensitive, and if they audit a fast rewind, their growth rate will become inverted. In severe cases, plant death may result.

When evaluating a stereo product, care should be taken that the plant is placed precisely midway between the two monitor speakers. An interesting phenomenon occurs if this precaution is not fully observed. Botanists have shown that plants will grow towards a source of light, either the sun or an artificial lamp. If the lamp is moved from time to time, the plant will change its direction of growth accordingly. By extension, one might assume in our work that a plant would incline toward the louder of two speakers. The opposite has been shown to be the case. In an apparent attempt to balance itself within the sound field, most plants will grow in the direction of the quieter speaker. Yet, since the louder speaker exerts the most influence on the rate of growth, some involved mathematical calculations become necessary before valid conclusions may be reached. It is much simpler to determine before the experiment that the plant is perfectly centered.

As mentioned earlier, experiments have shown that plants react to a hostile environment. Whether actual growth rate is affected is not yet certain, and further experiments are indicated. For the moment, lyric content should be considered as a weighting factor in analyzing plant evaluation of final mix-downs. Songs containing references that might be construed as threats to plant life in general should (for the time being) be judged by more conventional means.

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← *Audio Designs—Circle 30 on Reader Service Card*

Circle 40 on Reader Service Card

Theory and Practice

NORMAN H. CROWHURST

● Although this column has devoted some 10 of its 24 issues to some aspect or other of transistors, it seems I have never discussed the mysteries, as one reader expresses it, of modern power amplifier output circuits. As he correctly points out, in the "good old" tube days, we had some pretty fancy circuits going, that were thoroughly explained several times over, but nobody seems to have done the same for the new transistor circuits.

When transistors first arrived, they could amplify, but it was a while before big enough types were developed to handle the kind of output power that audio amplifiers need. At the time, while there were some pretty fancy circuits, they could be broadly divided into two groups: one group, somewhat of a minority for practical reasons, favored the abolition of the output transformer; the other accepted it as a practical necessity.

Tubes, operating with plate supplies in hundreds of volts and currents in hundreds of milliamps, naturally needed loads in thousands of ohms, however the circuit was arranged. As voice coils with that high an impedance are not practical, at least one transformer seemed a virtual necessity. The abolitionists were struggling.

When transistors eventually got big enough to handle the power, they required a collector voltage well below a hundred, and used peak currents around ten amps: much better for direct matching to voice-coil impedances. It looked as if the abolitionists were in business at last. But habits die hard. The majority were accustomed to using transformers, and remained convinced they are inherently more efficient.

The transformer users ran into two problems, for which they devised more or less successful solutions—aggregate being less. The two problems were mutually aggravative, in the technological sense.

Transistors, as well as possessing all kinds of parameters relating input and output voltages and currents, have properties that depend on temperature, which lead to a nasty little habit called *thermal runaway*! After you have found yourself a nice stable operating condition that will sit there all day, you maybe push the circuit a little harder and get the transistors warm. Then rising temperature results in rising current and *vice versa*, until very quickly your transistors pop.

Along with that is the fact that, to achieve their vaunted superior efficiency, transformer windings must have resistance values much lower than the impedances they represent. If the load reflected by the secondary back to the primary looks like 10 ohms, the pri-

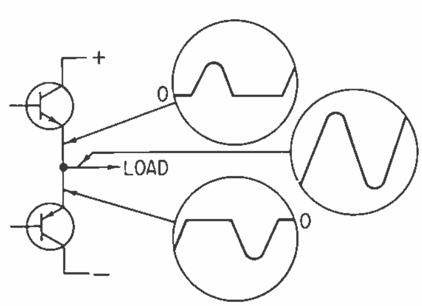

Figure 1. The basic concept of complementary transistors working in class B.

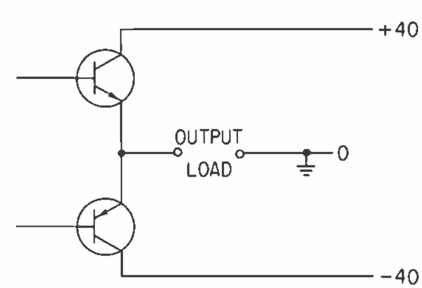

Figure 2. The more obvious way to supply the circuit, with two separate voltages, positive and negative of ground.

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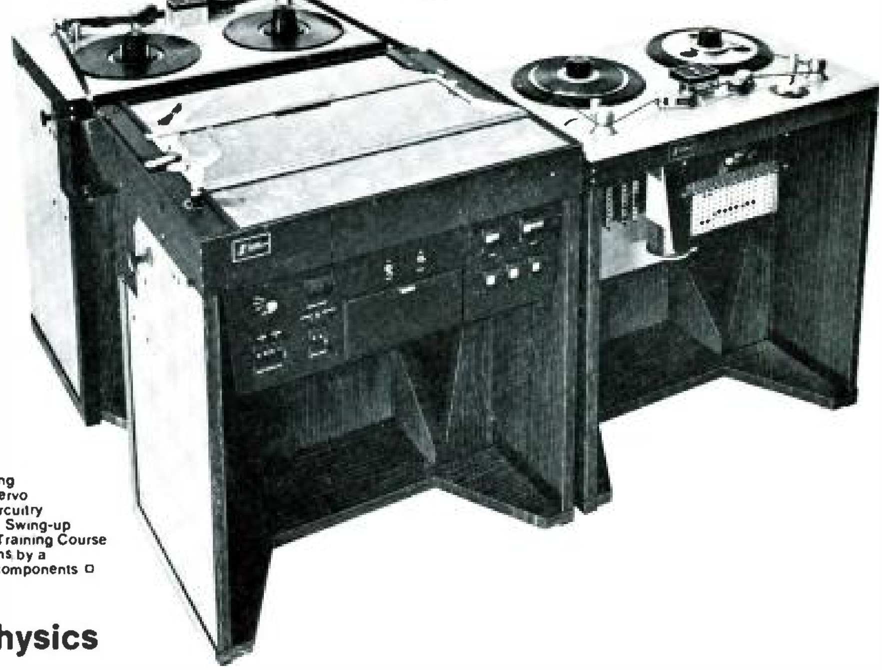

The basic Series 1200 System consists of a Model 1260 Loop Bin. Model 1210 Master Reproducer. and a Model 1220 Slave Recorder. Each slave in the system will produce 100 C-30 cassettes per hour at the standard Gauss 32:1 duplication ratio. The master tape is reproduced at 240 ips and the duplicate copies are of peerless fidelity.

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American Sound Corporation, Warren, Michigan
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2309	horn for 2390
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2328	2" to rectangular throat adaptor
2329	2" dual to rectangular "Y" adaptor
2330	1" to rectangular throat
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2341	right angle horn, 2" throat 60° x 40°, 800 Hz.
2345	radial horn, 800 Hz. 90° x 40°
2350	radial horn, 300 Hz. 90° x 40°
2355	radial horn, 300 Hz. 60° x 40°
2356	long throw radial, 150 Hz. 20° x 40°
2360	1 x 2 multicell, 300 Hz.
2365	2 x 4 multicell, 300 Hz.
2370	2 x 5 multicell, 300 Hz.
2375	3 x 5 multicell, 300 Hz.
2380	3 x 6 multicell, 300 Hz.
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2395	horn/lens, 140° x 45°

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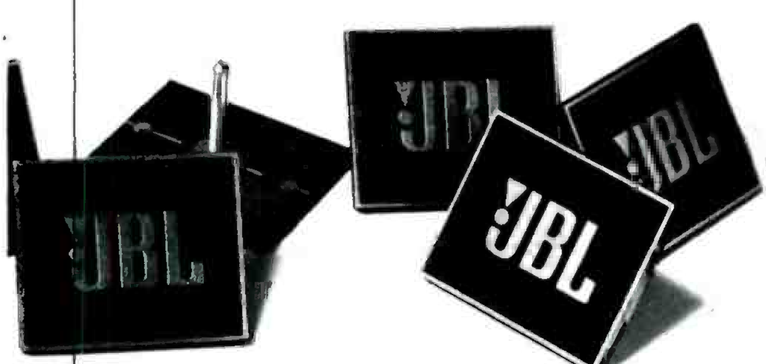
2470	50 W. driver 1" throat phenolic diaphragm
2480	120 W. driver 2" throat phenolic diaphragm



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3115	500 Hz. (except 2220)
3120	1200 Hz. (for 2205A, 2220A)
3125	1200 Hz. (for 2150)
3150	500 Hz. high power (except 2215)
3180	800 Hz. high power (except 2215)

ENCLOSURES AND SYSTEMS

4310	control monitor – gray
4310 WX	control monitor – oiled walnut
4320	studio monitor – gray
4320 WX	studio monitor – oiled walnut
4370	slant-front paging radiator 180° x 90°
4375	vocal column (speech range) (line radiator)
4380	(extended range) column (line radiator)
4503	utility cabinet
4520	utility dual rear loading L.F. horn
4530	utility single rear loading L.F. horn
4550	utility dual front loading L.F. horn
4560	utility single front loading L.F. horn

MIXER POWER AMPLIFIERS

3101	10/15 watts, 2 channel
3202	25* watts, 5 ch. max#
3204	40* watts, 5 ch. max#
3206	60* watts, 5 ch. max#

POWER AMPLIFIERS

4004	40* watts, with PRO-GUARD
4006	60* watts, with PRO-GUARD
4010	100* watts, with PRO-GUARD
4015	150* watts, with PRO-GUARD
4030	300* watts, with PRO-GUARD

PREAMPLIFIERS

5100	1 mic preamp – self powered

MIXER/PREAMPLIFIERS

5300	5 channel max #
5600	8 channel max #

ACCESSORIES

7001	PRO-VOX
7102	Speaker Monitor Panel
7103	VU Meter Panel
7104	PRO-NOI
7105	PRO-PLUS
7106	PRO-COM Compressor
AMC-2	2 Mic Preamp Expander
EPC-10	Precedence Circuit, Plug-In
MBT-10	Matching/Bridging Transformer, Plug-In
MPT-1	Pre-Amp for Magnetic Phono or $3\frac{3}{4}$ & $7\frac{1}{2}$ Tape Head, Plug-In
RM-6	Rack Mount for 3101
RM-7	Rack Mount for 3202
RM-8	Rack Mount for 3204, 3206
RMP-1	Blank Panel $1\frac{3}{4}$" x 19"
RMP-2	Blank Panel $3\frac{1}{2}$" x 19"
RMP-3	Blank Panel $5\frac{1}{4}$" x 19"
RMP-4	Blank Panel 7" x 19"
RMP-5	Blank Panel $8\frac{3}{4}$" x 19"
RMP-6	Blank Panel $10\frac{1}{2}$" x 19"
SPT-1	Pre-Amp for Tape Head $1\frac{7}{8}$, Plug-In
UH-1	Pre-Amp for High Impedance, Plug-In
VU-10	VU Meter – Rectangular Edge Reading
XE-10	"T" Pad Converts Mic Input to Unbalanced 50K Input, Plug-In
XT-10	Microphone Transformer, Plug-In

Notes: * All amplifiers are rated by RMS Professional Standards.
Ask for specification sheets for higher commercial rating. # Using AMC-2 expander module.

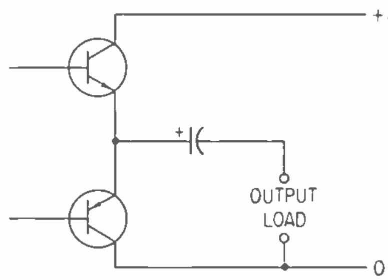

Figure 3. An alternative that enables a single supply to be used, at questionable component economy.

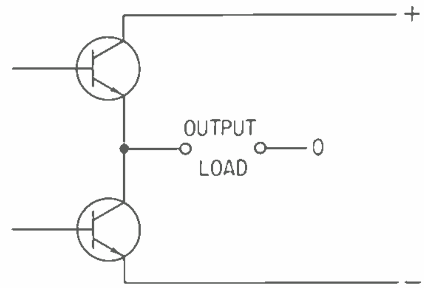

Figure 4. To achieve audio power, complementary symmetry of the output transistors is not possible; identical types must be used.

mary resistance should be less than half an ohm.

Now the 10-ohm load may be just what the transistors need to make them deliver the power. But when they are driving all out, each transistor momentarily becomes virtually a short-circuit—saturated. If a real 10 ohms is there, it limits current to the rated maximum. But when the only real resistance in circuit is half an ohm, the current can, under some unforeseen circumstance, run as high as 20 times the rated maximum.

Those two problems combine to make a good formula for failure. Meanwhile the transformerless advocates were faring a little better, if not quite out of the woods. In comparison with the problems associated with transformers, obviously the nearer we can get to direct coupled, so the load itself limits current, the less likely we are to encounter runaway current problems.

Transistors are inherently a highly efficient form of current modulator: their condition changes all the way from open-circuit to short circuit. This means that operation in class B is particularly attractive. The theoretical efficiency of a class-B modulator is 78.5 per cent as compared with the class-A's 25 per cent. So a realistic efficiency with class-B transistors is 75 per cent.

This means that, to deliver 100 watts of audio power, the transistors dissipate 33 watts with a class-B circuit, as against 300 watts if class A is used, because 100 watts is 75 per cent of 133 watts and 25 per cent of 400 watts.

Without a transformer, the nice way to use class B would be to have two complementary transistors deliver opposite halves of the waveform (Figure 1). The obvious, direct way to do this is to use two supplies for the output transistors (Figure 2). An alternative is to use one supply for the whole voltage, and capacitance couple (Figure 3).

What does the alternative actually save? Because of the relatively low load impedance, to pass the low frequencies a very large electrolytic capacitor is needed for output coupling. In effect, we need just as much electrolytic capacitor, either way. Rectifier and power transformer requirements are not very different (cost-wise) either. And putting the big capacitor in series with the load may achieve low-frequency coupling just as well, but all that capacitor is also hanging on to the output circuit at high frequencies as well.

So the over-all advantage goes to the double supply circuit.

That this idea works could be shown with complementary transistors, of the biggest ratings available, but they would not deliver the kind of audio power needed. By now quite a variety of power transistors, big enough to handle these wattages, is available, but they just do not come in complementary pairs.

There are pnp and npn; there are germanium and silicon types. But nothing that looks like matching characteristics that could be used in complementary fashion, combining pnp and npn. The only way to combine transistors in the output circuit itself is to use two (or more, if you double them up, to get more power) transistors of the same type (Figure 4).

Fortunately there is relatively simple drive circuit, that enables a complementary pair of drive transistors to be used (Figure 5). This gets back to the point where the voltage driving the two

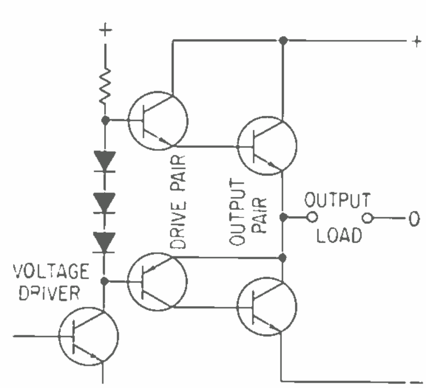

Figure 5. Combining the two of Figures 3 and 4: complementary symmetry drive with identical outputs, and a high voltage drive. Quiescent condition is controlled by the diodes.

THANK YOU WILLI STUDER

Seldom has a tape recoder captured the immediate and whole hearted acceptance of the audio fraternity as has the Revox A77.

Willi Studer has been designing superlative magnetic recording equipment for the broadcasting and recording industries for years. The A77 is his latest—a great machine that outperforms recorders costing up to three times as much. Specifications and performance are a joy. Want details? Write us for full particulars.

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Circle 38 on Reader Service Card

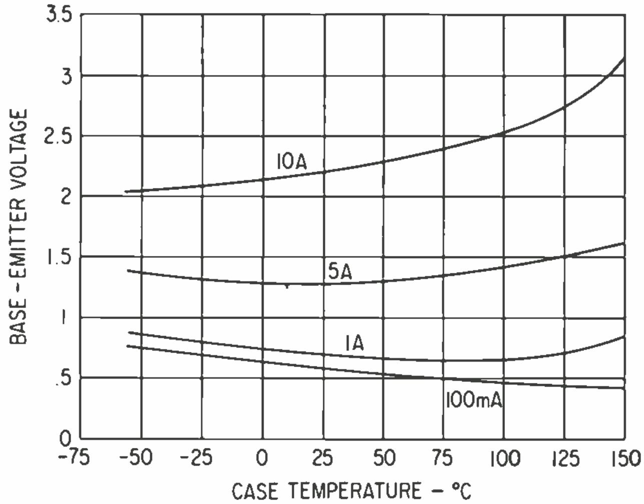

Figure 6. Typical base-emitter voltage characteristics for a silicon transistor. The curves represent different collector-current values. Note that the low-current condition reduces voltage with rising temperature, resulting in a tendency of the operating point to run away, although the high-current region is protected by a rising volt drop.

bases is identical for signal, and spaced apart by a d.-c. voltage that is quite small.

But that d.-c. voltage is critical, because it makes up the sum of the base-emitter drop of the two drive transistors, plus the base-emitter drop of one output transistor. And this is complicated by the fact that base emitter voltage drops depend not only on current, but also on temperature. FIGURE 6 shows typical characteristics for a silicon type transistor.

In all these transistors, base current controls collector-emitter current, so if base current rises, transistor dissipation rises. The primary purpose of controlling the difference in voltage between the drive transistor bases is to control quiescent current. If the circuit was not required to deliver class-B power, a larger voltage difference could be used with series resistors to control base *current.*

If a resistor large enough to control quiescent base current were used, the drive voltages to provide signal-power base current through the same resistors would be enormous: resistance control of quiescent operating point is not feasible.

The method universally used is selected diodes that will provide the correct voltage difference when at the same temperature as the transistors they are driving. The drive voltage is derived from a high-voltage drive stage (also shown in FIGURE 5), with a collector resistor from supply to the "top" drive base, then the diodes controlling base-voltage difference, then the collector of the stage providing the voltage swing.

This circuit lends itself very readily to the provision of over-all feedback, making it possible to get very linear amplification at quite high power levels. However, there are more problems that we will take up next time. One is matching to various impedances. At least one early amplifier did this by designing to match 8 ohms, using the same terminal for 16 ohms, and putting a 4-ohm wire-wound resistor in series with the 4-ohm terminal.

The disadvantage of that method is that the amplifier delivers its rated power only into 8 ohms. Into 16 ohms it delivers the same voltage, which is half the power. And into 4 ohms, the internal 4 ohms limits output to the same current, which again means the load only receives half power.

But there are other problems. Although the circuit described above will deliver its power beautifully into an ideal resistance load, practical loudspeaker-type loads can lead into a variety of dangers: over-current, over-dissipation and reverse voltage. There is enough meat for at least another go round! ■

"The Dolby System is essential

for the recording of chamber music,"

notes Leonard Sorkin, First Violin of the Fine Arts Quartet.

The music of the string quartet is, by its nature, small-scaled and intimate. Unlike the symphony orchestra, the string quartet can actually perform in an average living room. Much of the scoring is open and exposed, with extreme pianissimos and passages of great delicacy. Thus, when recordings of string quartets are played in the home, listeners are acutely aware of any intrusions of tape hiss or print-through. The Dolby System effectively supresses these distracting noises.

For the recording of the Karel Husa Quartet No. 3 (winner of the Pulitzer Prize for music in 1969) on Everest Records, Leonard Sorkin felt that it was especially important that the unusual and subtle timbres demanded by the composer should not be marred by tape noise. According to Mr. Sorkin, "The Dolby System was the solution to this problem."

String quartet, symphony, opera, multi-track pop/rock . . . whatever your recording endeavor, you can make a better recording with the Dolby System.

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Circle 36 on Reader Service Card

Sound with Images

MARTIN DICKSTEIN

MORE HOME ENTERTAINMENT

● It did not take a long time for the audio cassette tape recorder to make its entrance on the home scene. This device, with the availability of commercially-recorded cartridges and the extreme simplicity of playing back or recording literally everywhere, soon stole the scene and became not only a household item but was carried all over by teenagers, reporters, students, and business men.

Similar steps are now being followed in the video field. With at least one TV set in nearly every household, and many more being carried almost everywhere the portable radio and cassette recorder can go, it is only natural that a cartridge or cassette device would be developed which can feed video and audio to a TV set. Recently there was an announcement that CBS had invented the e.v.r. (electronic video recording) system with its cassette film recorded by a special CBS process for later playback via the antenna connections of a home TV set (see **db**, September, 1968).

Next came the announcement that a different type of unit developed by RCA would also play video programs through the home TV set using a cartridge recorded by RCA. The principle behind this unit is the application of a relatively new development—holograms. This device, it was then expected, would be on the market for home use in the early '70's (see **db**, December 1969—this column.)

Near the beginning of last year, an announcement was made, and toward the end of last year, a third entrant appeared in the field of video-cassette machines. This unit, introduced by Sony, makes use of a most recent development in the video field—video tape, similar to that used in semi-professional and home video tape recorders—rolled into a cartridge just as done with audio tape. This makes a race out of the rush for the consumer dollar.

The *Color Videoplayer* (FIGURE 1), as the unit has been named, uses a *Videocassette* capable of providing up to 90 minutes of program material. As presently envisioned, the unit should be on the market in Japan at the end of this year and the ultimate price in the U. S. would be in the vicinity of $350.00. The cassette will be recorded, and purchase selection is to be made from a library of material which will be collected in a duplicating center. As the cartridge uses video tape, the previously recorded material can be erased any number of times and new program material put on the tape.

Thus, the customer will be able to return his cartridge to the manufacturer or the nearest duplicating center

Figure 1. The Sony Videoplayer. The tape cassette will hold up to 90 minutes of videotape.

INTRODUCING THE UNDER $75,000 TAPE DUPLICATING SYSTEM*

* The same high volume system that produces GRT's stereo music tapes.

The GRT 260 Tape Duplicating System

Sound as it was originally recorded

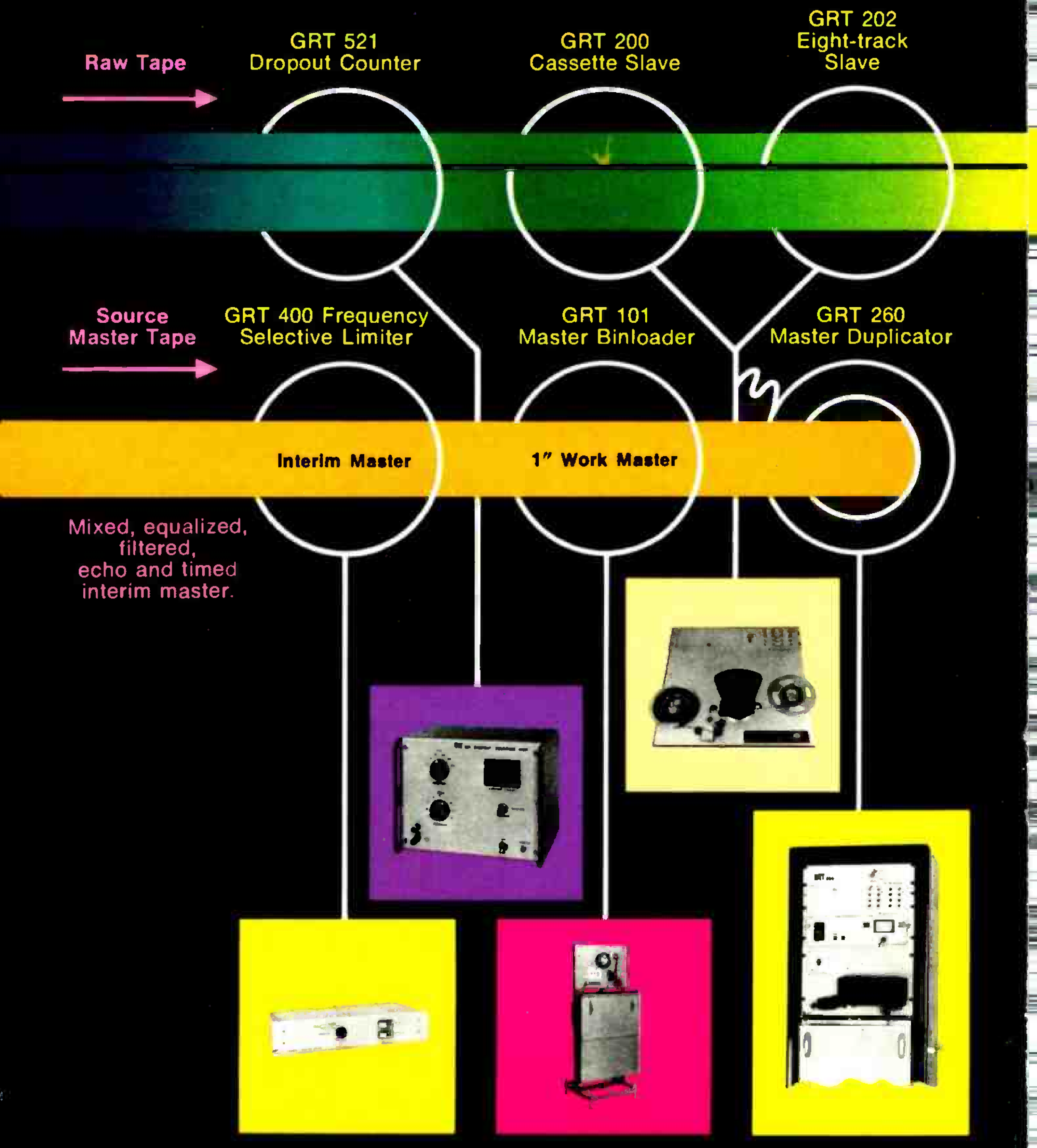
The new GRT 260 system offers you higher quality, higher yield at a lower cost. Duplicated tape at playback speeds of 1⅞ ips (cassette) and 3¾ ips (8-track cartridge) has negligible generation losses in fidelity.

Production yield from the 260 system is over 6,000 cassettes or 12,000 8-track albums in an 8-hour shift. Unit cost is under 8 cents per copy, including labor and overhead, excluding materials.

This type of performance comes naturally with the 260 because all components are production oriented.

For example, the Master Duplicator employs a simple dial for the number of copies. With the push of a "start" button, it permits transport systems to

The components that make up the 260 System



stabilize, automatically leader, duplicate, place test tones for quality control and stop on completion. The Master Duplicator is maintained with pull-out drawers, interchangeable circuit boards and front panel monitoring.

Highly stable tape transports and interchangeable circuit boards are key features on the 200 and 202 Slaves. Slaves may be added or deleted from the line with no change in slave bias.

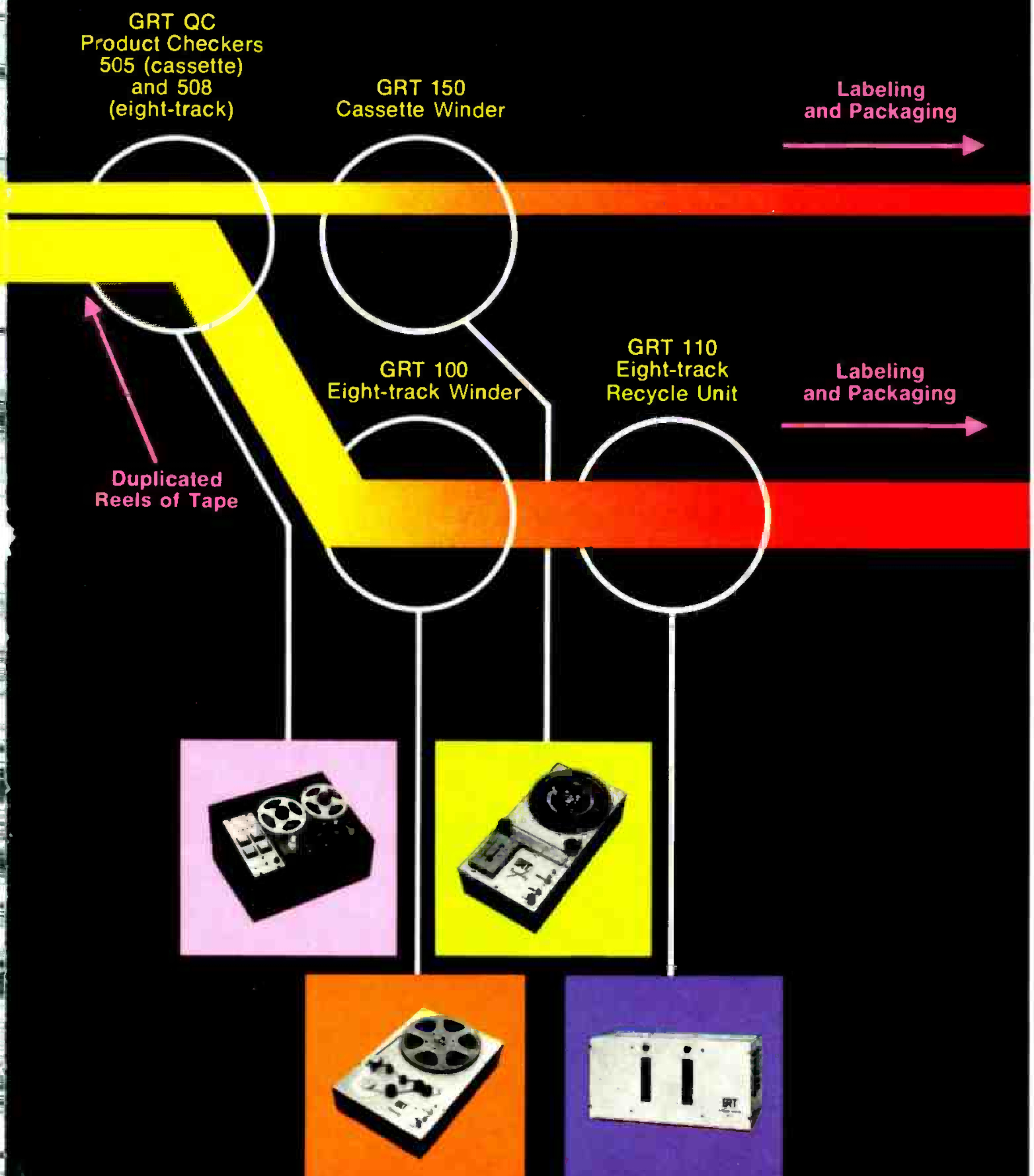
The 521 Dropout Counter controls incoming tape for instantaneous amplitude variations (dropouts) resulting from inconsistencies in the oxide coating.

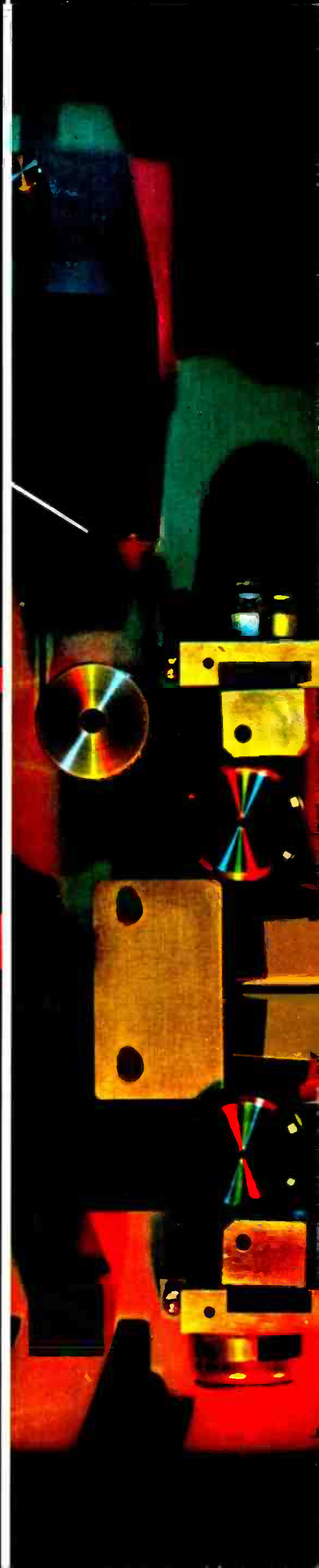
For an audible check of material before assembly, the 505 and 508 QC Product Checkers permit monitoring of the bulk duplicated material and auditing of the test tones placed on the duplicated reels.

Assembly production is speeded by the 100 and 150 Tapewinders which respond to inaudible tailoring tone and stop and cut tape between albums.

Further quality control on 8-track cartridges is accomplished with the 110 Recycle Unit. It checks mechanical stability by running the tape a complete revolution and automatically stopping.

All GRT components are available individually or as a complete 260 system. Complete details on the total system are covered in our new catalog. Write for it today.





Here are just some of the specifications that make the GRT 260 Tape Duplicating System a top performer

GRT Master Duplicator and Slave

Format: 7½ ips NAB, four or eight 70-mil tracks on one-inch tape

Type: Continuous loop, 1200-foot capacity, remotely loaded (1800-foot bin under development).

Speed: 240 ips

Duplication ratio: 32 to 1 (slave speed 120 ips for 3¾ ips copies. 60 ips for 1⅞ ips copies). (Approx. 2-sec duplication time per minute of program)

Frequency response of copy (Deviation from master): ±2 dB 50 Hz–10,000 Hz

Distortion: Reproduce and record electronics each generate less than ½% THD at any frequency from 1600 Hz to 400 kHz at any level up to 15 dB above operating level. Bias system generates less than ½% even order distortion in recorded signal

Signal-to-noise ratio: System contributes less than 3 dB weighted noise over that of bulk erased tape

Flutter: .25% rms, 0.2 to 200-Hz unweighted

Master/copy speed error: 0.5% max.

GRT 400 Frequency Selective Limiter

Frequency response: (below limiter threshold) ±1 dB (25 Hz–25 kHz)

Output clipping level: +24 dBm

Noise: (unweighted 20 Hz–20 kHz) = −70 dBm

Input impedance: 200 k ohms, single-ended

Output impedance: 12 ohms, single-ended

Attack time: Full limiting within the first half-cycle of input signal

Release time: 50 ms maximum

GRT 521 Dropout Counter Unit

Threshold: Adjustable in 2 dB steps, −12 dB, to +4 dB

Dropout duration: (selectable) 2.5 ms., 5 ms., 10 ms., 20 ms., 40 ms., 80 ms.

Input signal: Any "carrier" between 1 kHz and 100 kHz may be employed

Counter capacity: Six digits with push-button reset

Maximum counting rate: 10 counts per second

GRT 100 8-track Tapewinder

Max. reel size: 14" dia. NAB hub

Winding speed: Approx. 100 ips

Tailoring tone requirements: 10–20 Hz at approx. operating level on both channels of a single stereo pair

GRT 150 Cassette Tapewinder

Max. reel size: 10½" NAB hub

Winding speed: Averages 120 ips

Winding time: Approx. 17 seconds for average length cassette album

Tailoring tone requirements: 10–20 Hz at approx. operating level on both channels of a single stereo pair

Blank tape winding feature: (optional) precision timing circuit with 3 pre-settable winding times for C-60, C-90 and C-120 cassettes

GRT 110 8-track Recycle Unit

Tape speed: 60 ips

GRT 508 QC Product Checker

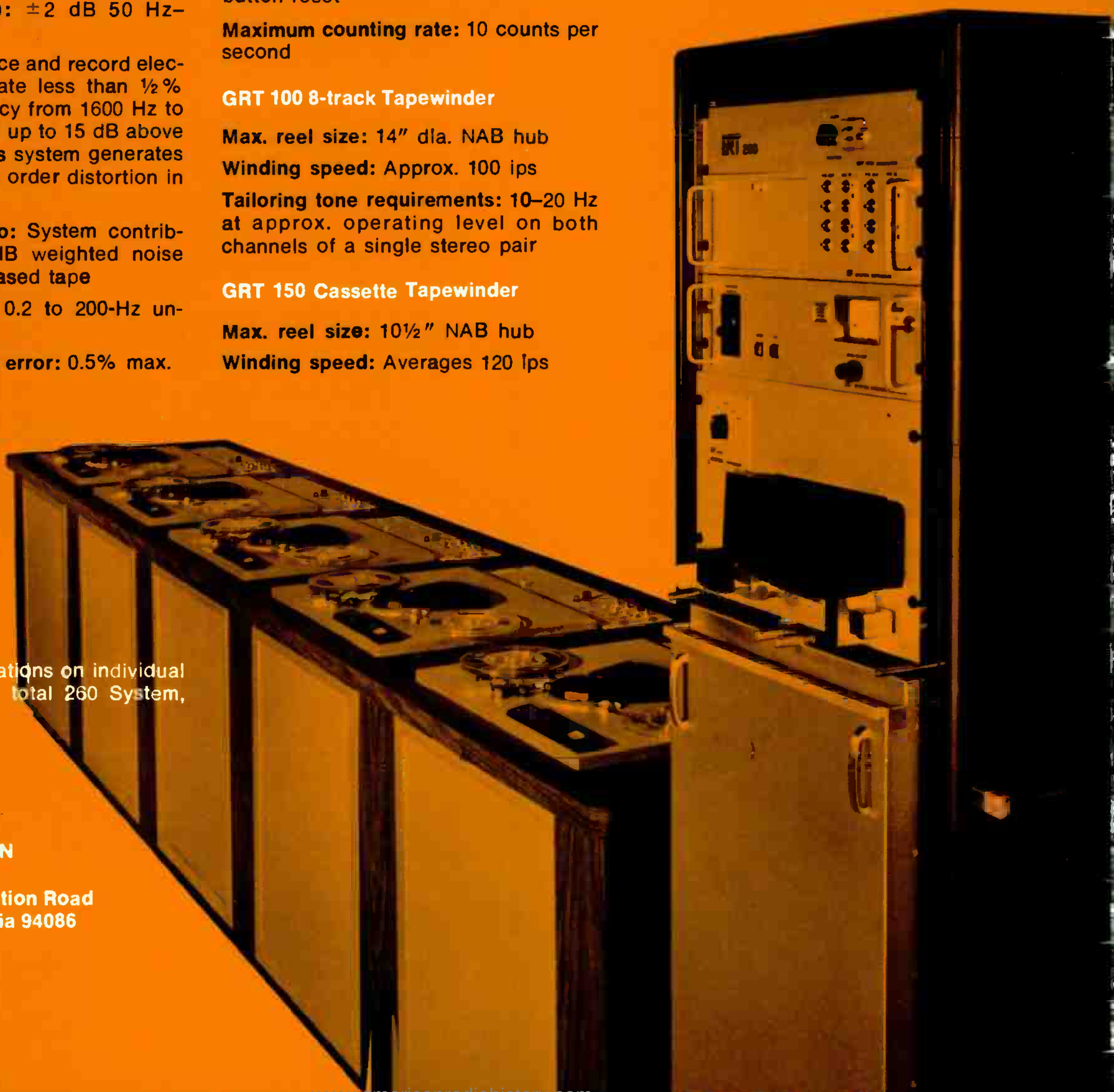
Tape width: (nominal) ¼ inch

Tape speed: 3.75 ips ±0.3%

Frequency response: 50–10,000 Hz ±2 dB

Signal-to-noise ratio: Peak record level to unweighted noise, 20–20,000 Hz 52 dB

Output: 0 dBm operating level, clipping greater than +20 dBm



For further specifications on individual components of the total 260 System, write:

GRT

GRT CORPORATION
Industrial Division
1286 Lawrence Station Road
Sunnyvale, California 94086

E-104 2-2-70 15M

for a new program when the old material has had its full time on the owner's TV screen. The cost of the non-recorded cassette is presently expected to be about $20.00. The cartridge will have a counter on it and a program fee will be charged according to the number of times the material on the tape has been played.

To meet worldwide standards, Sony worked with Philips and others and hopes to develop the technology necessary to achieve this aim. The unit will be small enough to mount in a hi-fi cabinet, is compatible to color and black-and-white TV sets, has two audio tracks so it can be played back on stereo systems, and has the capability of having the cassette removed before the material on it has been played through to the end (and without rewinding). Thus, when the cartridge is again started it will pick up at the stop point.

Each of the various systems indicates that their cartridges will be available with recorded material on the tape (or film). However, Sony will also have an adapter available (which will cost about $100.00) that will permit the consumer to record on non-recorded tapes directly from the home TV set, in color or black-and-white.

It is expected that the recorded library will be opened to motion picture and TV companies as well as recording companies, educational institutions, sports promoters, and industrial film producers. They will be able to transfer their material to the tape masters for reproduction in cassettes so that remote locations where TV is not normally available may be able to receive their share of entertainment and education via the TV tube. In all of these applications the second audio track can be used for added commentary over the video sound track or for dubbing on the language in which the tape is to be marketed or used. It seems that the uses and applications are innumerable. Can commercials be far behind?

The basic specifications of the Sony unit are:

- Horizontal resolution . . . Monochrome-300 lines; Color -250 lines
- Audio frequency response (2 channel stereo) . . . 50-12,000 Hz, plus 1.5 dB minus 3.5 dB
- Audio s n ratio . . . . . . More than 40 dB
- Playing time . . Maximum of 90 mins.
- Dimensions:
  - Videoplayer . 15 x 16 x 8 inches
  - Cassette . . . 8 x 5 x 1¼ inches (Uses ½-inch video tape)
- Weight:
  - Videoplayer . 32 lbs.
  - Cassette . . . . 1 lb. ■

Many readers do not realize that they can also be writers for db. We are always seeking good, meaningful articles of any length. The subject matter can cover almost anything of interest and value to audio professionals.

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Studio Construction Techniques

WILLIAM R. GRAHAM

Simple building procedures are described that will enable tight-budgeted builders of sound studios to achieve their goals.

TODAY'S TYPE OF BROADCASTING calls for studios that are really announce booths or control rooms where the announcer is also the operator. Naturally the acoustics of these rooms must be good, but with close microphone procedures it is more the purpose of these rooms to stop external noise and to provide damping of clicks and taps made by personnel working in front of a live mic. than to provide music-room acoustics.

William R. Graham, MAES, MIEEE has designed broadcast studios. He is based in Kitchener, Ontario.

It is the purpose of this article to show that a small radio station can achieve isolation by using simple modern building techniques. If you are leasing space, the landlord will prefer this type of construction because it is easier to remove when you leave. The fire inspector will certainly appreciate the method over anything but concrete blocks. Your building costs will be lower because the contractor and his staff are familiar with the ingredients and can work at a good pace.

A few years ago I had the job of designing new studios for a Canadian radio station. Space was a problem and the landlord specified only drywall construction. To further complicate things the floor was reinforced concrete and could not be cut into. The first step was to run tests at the old studios to determine the highest noise level in each of the working areas so that placement of the areas could be worked to advantage in the new floor plan, and so that the degree of sound attenuation for the new studios could be worked out. Figure 1 shows the results of these tests. A noise criteria of 25 was set because the station was not planning to produce dramatic programs and so the greatest difference between the NC-25 curve and peak operating noise was the amount of transmission loss required. This would require walls of a Sound Transmission Class of 45, but I felt that perhaps some of the measurements had been a little conservative and if a few dB could be gained without getting into complicated construction methods, so much the better.

A floor plan was the next step and I decided to separate each control room (high noise area) with a studio, and to place the studios where only voice work would be done at the furthest point from any high-noise level other than loudspeakers that would carry that particular audio. This would give optimum sound-source-to-sound-source placement as well as a buffer between major sound sources. Figure 2 is the final layout. Control room 1 and/or 2 could be used for on air while the other was used in commercial production or routine maintenance. If control room 1 was on air then studio B would be used for news and studio C used for commercial production. Studio A would be common to both control rooms with several microphone outlets—with the idea of phone-in shows with guests. Control room 3 would be used for f.m. The only flaw in this plan was that studio B is next to the news room. The answer here was to put any room acoustic material on the adjoining wall as extra protection. As it finally worked out the news room wanted a large storage cupboard and so it was placed on their side of the wall.

I will not attempt to go through all the calculations here that were required to choose the materials and methods, but I will describe the final design. Any station whose requirements work into the measurements listed above, should be able to follow this design and obtain pleasing results. Before I carry on though I must stress one point: *Details are very important.*

As an example, a total leakage of 1 square inch or a difference of 1 inch of absorbent material can make a difference of 10 dB in the TL of your wall.

Figure 3 shows a cross section of the walls. The materials used are designated in general terms as each manufacturer has his own. A meeting should be arranged between the station engineer and contractor to ascertain that the materials supplied are equivalent to those called for.

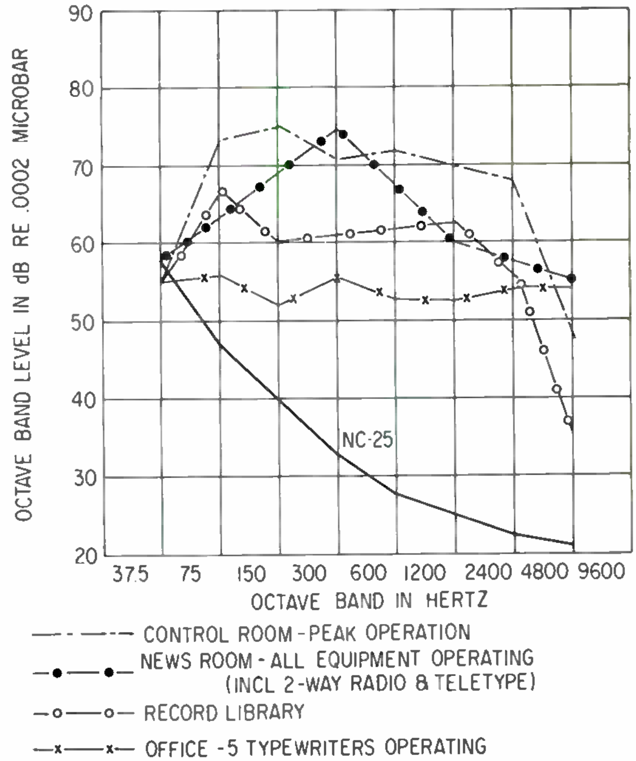

Figure 1. The results of the noise tests described. The noise criteria (NC) of 25 was adopted because of the type of programming.

In the way of general description, each wall was treated as a unit of its own. It is completely surrounded by neoprene; top, bottom, and sides. At a joint with another wall the end is sealed with drywall then the neoprene acts as a buffer and the drywall-to-drywall joint is taped and plastered. Finally the drywall-neoprene-drywall joint is sealed with an elastic caulking that can be painted. All drywall is backed with 5/8-inch softwood fiber board in such a way that the joints do not line up with those of the drywall. Figure 4 should clearly show the type of studs that are used. The wall boards were fastened to the studs with screws made for the purpose. For this type of stud a track is layed on the floor first (in this case on a layer of neoprene)—but since the studs are staggered and protrude into the space of the other wall a 2 inch track was laid for the second wall and the bottom and top of each stud was tailored to match as in Figure 5. The 3-inch glass fiber bats were woven between the studs layer upon

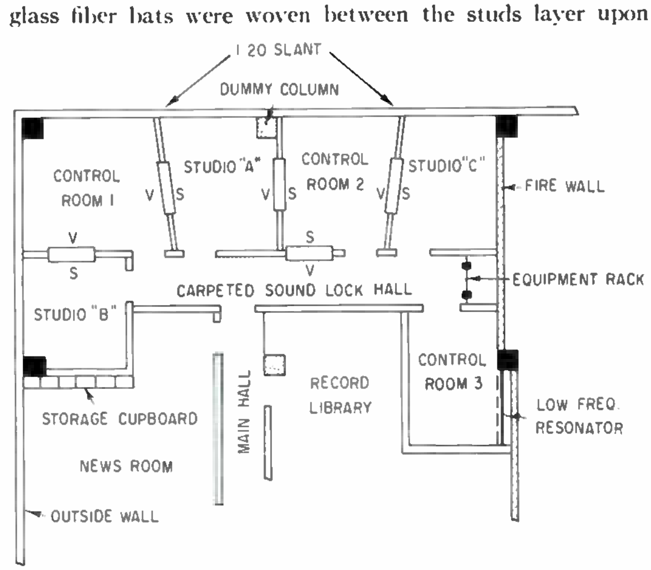

Figure 2. The final studio floor plan decided upon. Note the deliberately angled walls.

layer vertically and with the end joints overlapping. This method offers some protection against the insulation bunching up at the bottom of the wall as the years go by. FIGURE 6 gives a good look at the double-floor track mentioned above. This floor track needed to be fastened down so it would not slip while the studs were being put in place and so it was "tacked" to the concrete floor at each end and the middle until the studs were up. The walls were built from the floor right up to the concrete ceiling slab. To assure a seal at the slab, regular caulking cement was used as this would be above the decorating level. To aid in later acoustics treatment of the internal room, the wall separating control room 1 and studio A, and the wall separating control room 2 and studio C were constructed on a 1:20 slant from left to right.

It is rather pointless to construct walls with a high TL figure and then put in windows that don't match. For this reason details of the window construction must be followed as closely as for the walls. First of all I chose two different thicknesses of glass, one for each side of the double construction. This prevents them from being resonant at the same frequency and passing that frequency as though there was no glass at all. The thicknesses selected are ¼-inch plate and 3/8-inch plate. The thicker of the two is tilted in the frame so that there is a 1½-inch space between the pieces at the top and 7 inches at the bottom. The tilting is clearly indicated in FIGURE 2 by the V and S symbols beside each window. V is for vertical and S is for slanted. This tilting or slanting also helps with the internal room acoustics, as an announcer's voice will be reflected away to the ceiling tile.

Next the size of windows. Make them no larger then necessary to permit comfortable viewing of the next room from the operating position. For an average control setup you will probably find that 3-feet high and 4-feet wide, along with 3½ feet off the floor is ample.

The window frames are to be made in two parts, one for each of the double walls. There are many possible ways to construct them and a discussion on this should be held with the trim carpenter. The important things to remember are:

No leaks can be permitted, so go heavy on the putty at joints.

A means must be found to seal the frame to the wall. Try ¼-inch neoprene.

A socket in each frame for the glass must be provided, and it must be large enough to hold the glass and a rubber or felt seal. In some cases a wood strip is used to hold the glass and seal against the inner stop and it is made adjustable by means of screws. This permits some trimming of glass resonance if a problem arises.

The space between the two frames is best filled with a damping material. I used strips of foam plastic that resembled cork when it was painted.

The next problem that arises are the doors. A good door should be heavy, solid, and if possible have a layer of 1/8-inch lead in the center. It should also make a good seal with its frame and the floor! The frame should also be sealed carefully to the wall. In the particular case I am quoting from, the station management refused to install proper double-slab

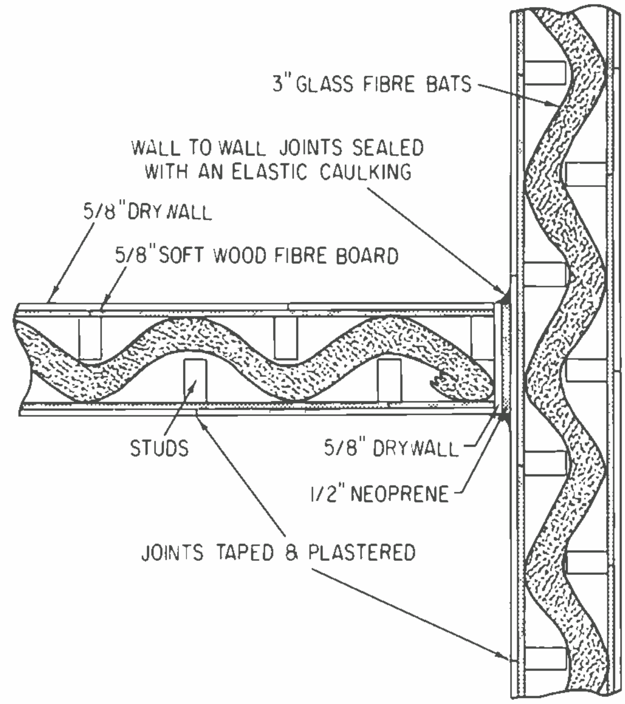

Figure 3. A detailing of the wall juncture necessary to effect the required sealing.

lead-filled doors. Referring to FIGURE 2 again, doorways were placed so that there was no door directly across from another. I think it is this fact and the carpeted sound lock hall that saved the situation. Certainly all the rest of the planning and expense could have been to no avail. As it is, the staff turn the monitors up to the "happy hi-fi fan" level and while it can be heard in the sound lock it does not get back to another room.

I haven't said anything yet about air conditioning—and with good reasoning. I have shown you so far how good studios for today's radio can be built without a great deal of expense, but this in no way means you can air condition on a low budget. The very fact that you have just created sound-proof rooms means that there are no air leaks. The air conditioner must on its own replenish the air constantly.

Figure 4. A view of the studding that should be used.

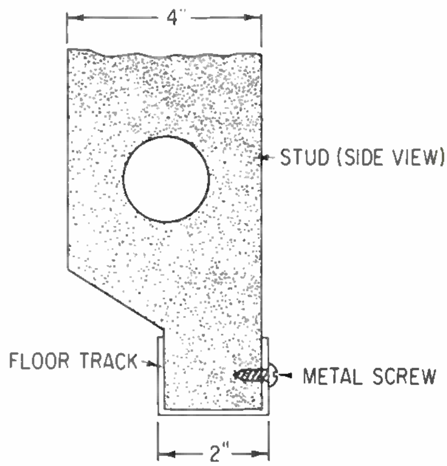

Figure 5. The bottom and top of each stud is tailored to fit the track as shown.

It must do this quietly and at the same time control the temperature and humidity. Yes, I said *humidity*. Most of the acoustic material you will put on the wall will soak up water at a fantastic rate. In one studio I know of, the announcers could do no more than four minutes of news without a change in voice because it was so dry.

Properly insulated or padded ducts will prevent sound from entering at right angles to the duct and cut down air-flow noise, but they will not prevent sound from travelling down the duct. Sound baffle sections must be installed in the ducts (both feed and exhaust) between each room. If the return air from the studios is to empty into a plenum over the general office area then two or more right angle elbows, with the final one pointing up, should be installed at the end of the exhaust duct.

Figure 6. The double floor track into which the studs are placed.

The position where air is to enter the control room or studio should be chosen carefully so that it will not be over microphone positions and yet provide maximum air flows through the room. At the same time the grill for each feed should be chosen for maximum air dispersion without any noise from the air flow coming through it.

Where each duct passes through a wall it was wrapped with neoprene and finally sealed with an elastic caulking cement. While it was not the case here, if the air conditioner is close to the studios it should be mounted on shocks, and flexible canvas ducting should be used to connect it to the main system. Noise-reduction baffles should be inserted in the first lengths of ducting and ducts should be suspended by vibration-isolation hangers. These same hangers should be used on all pipes that run overhead in the studios. In my case, one wall pillar in the studio area was a dummy with high pressure hot water heating pipes running through it. The solution was to fill the column with vermiculite by pouring it in near the roof slab and re-sealing the column with vermiculite by pouring it in near the roof slab and re-sealing the column with plaster.

Everything discussed above should provide good sound isolation—however there are still a few details that should be noted. Do everything you can to maintain the transmission loss just obtained; as an example, do not install back-to-back power outlets. This sort of thing is a perfect way to have leaks from room to room. Avoid cutting into the wall for fixtures of any sort. I used surface-mounted warning lights at the doors for just this reason. If you have a solid concrete floor as I did, bring your cable runs in overhead, above the ceiling. I found that 5-inch plastic tubing installed through the walls, and sealed to it with caulking cement, worked very well. After the cables were all in, this duct qas stuffed with cotton batting. I would recommend a wooden duct be run down the wall from the tubes and across the floor behind the equipment.

It is not the intend of this article to discuss room acoustics, but a few tips before construction starts could well make the job simpler when the time comes. Obtain a good book on the subject. This book should give you some detail on the *golden measurements* for studios and, if at all possible, work these ratios into the room sizes. Once the equipment is installed and the usual number of upholstered chairs put in place, the only wall treatment you should have to use will be well-placed panels of good acoustic tile. If you still have a low frequency boom as I did to control room 3 of FIGURE 2, a wall-mounted low frequency resonator should do the trick. (For further information on this subject, see Michael Rettinger's article elsewhere in this issue. —Ed.)

I have not meant to imply that broadcast studios are easy to construct, or that the highest quality possible should not be strived for. Rather, if the need is for the typical announce / operate control room used so often in today's radio with a booth for news or telephone shows, this method should give you a clean sound without noise from other sources, either by leakage or flanking. It should make the move to new studios much simpler, be faster than older methods, keep construction costs down, and give your listeners a very pleasant sound when you proudly announce you are operating from the new site.

A Modular Console Design

EDWARD J. GATELY, JR.

This article describes a particular approach to the design and manufacture of console components with a view toward user assembly of the finished system.

ABOUT FIVE YEARS AGO complex console design took a giant step forward with the development of the strip. For the first time, logical functions were grouped together in a single component. This development made the building of complex consoles substan-

Edward J. Gately, Jr. is the president of Gately Electronics, 57 West Hillcrest Avenue, Havertown, Pa.

tially easier than it had been previously. However, even with this advance in component design, the building of a complete console was beyond the ability of the average studio engineer or technician if the finished console was to be free of ground loops, hum r.-f. interference and other bugs; because the channel switching, mixing and control functions were not modularized as complete units.

For the past several years we have been working on the development of a new building block concept which would allow the average studio engineer or technician to build his own multi-input console with all the operational advantages of the strip, but eliminating as many as possible of the required interconnections and their potential for hum, noise and other troubles by combining the switching, mixing and control functions in modular building blocks.

The design criteria set down for this program are as follows:

1) A modular building block concept allowing the average technician to build for himself complex broadcast or recording consoles having a variety of functions and uses.

2) A modular concept allowing console design in many different console arrangements to meet similar customer needs, and to allow easy expansion or adaption to changing requirements.

3) Package designs which permit easy field servicing by means of plug-in semiconductors and plug-in circuit card assemblies.

4) All electronic circuitry to represent the existing state of the art by intelligent applications of integrated circuits.

5) The modular packaging concept to be adaptable to either console or standard rack mounting.

It was decided that these criteria could be best realized by design of six modular building blocks. Each modular building block having its particular function repeated eight times. (For example the equalizer module contains eight separate and independent equalizers.) These six modular building blocks are:

1) Input module
2) Equalizer module
3) 8 x 8 matrix switching and active combining module
4) 8 x 4 matrix switching and active combining module
5) 8 x 2 matrix switching and active combining module
6) Control module

Each modular package is completely self contained except for the power supply. All modules operate from a central highly regulated power supply having an output of +18 and −18 volts. One power supply has sufficient capacity to power up to 10 modules. This power supply is short-circuit proof.

Each amplifying function is accomplished using a hybrid operational amplifier utilizing an integrated circuit and discrete semiconductors, and having a minimum of 50 dB of negative feedback. Each integrated circuit or transistor plugs into a socket on a printed-circuit card. Each printed-circuit card contains, in addition to the semiconductors, the compensating circuits, feedback resistors, summing resistors, output capacitor, and other circuit components. The card plugs into a printed-circuit connector. Since most malfunctions of equipment of this type are by semiconductors, this double plug-in arrangement permits rapid service by either replacing the faulty semiconductor or the complete assembly.

For this application the i-c operational amplifier technique is combined with discrete components in order to meet all the requirements of present professional audio standards. The amplifiers must exhibit high open loop gains (90 dB), low noise (−127 dBm), and high output (+24 dBm into 600 ohms), and of course reasonable price. There are several ic's which meet all the requirements except for the output power requirement. By combining one of these ic's with a complementary pair emitter-follower output stage all of the requirements can be met.

All amplifiers are identical except for gain which is determined by feedback resistor selection. All amplifiers are capable of driving an unbalanced 600-ohm line directly. Where transformers are required for line balancing or phase reversal, a 600: 600-ohm transformer may be patched directly into the circuit.

INPUT MODULE

The input module contains eight input circuits each of which accomplishes the following functions:

Channel gain by means of a slide potentiometer. (Echo send by means of a rotary potentiometer. Mic gain of 30, 45 and 60 dB front panel selectable by means of a pushbutton switch. Off-line-mic selection by means of a push-button switch. Echo-send take-off point selectable by means of a pushbutton switch, PRE 1 (before equalizer and pot), PRE 2 (after equalizer and before pot) and POST (after equalizer and pot). Size 8¾ in. high x 19-in. wide x 7½-in. deep.

A functional block diagram of one of the eight sections of the input module is shown in FIGURE 1. Cue position on the slide pot along with an integral cue summing amplifier is available optionally.

EQUALIZER

This module wires into the input module and contains eight independent equalizers having Hi freq peaking type equalization at five frequencies and LO freq equalization at two frequencies. An *in-out* switch is also incorporated. Boost and cut curves are symetrical. The equalizer has unity gain. Detailed specifications for one of the eight equalizer sections follow:

Hi-freq. equalization; Peaking type-boost or cut at the following frequencies: 1.5 khz. 3 khz. 5 khz. 10 khz and 20 khz. Steps available; Boost or cut of 2, 4, 6, 9, and 12 dB. Lo-freq. equalization; 40 or 100 Hz boost or cut in steps of 2, 4, 6, 9 and 12 dB. in-out switch. Size 7-in. high x 19-in. wide x 7½-in. deep.

A functional block diagram of one of the eight sections is shown in FIGURE 2.

8 x 8 MATRIX SWITCHER AND ACTIVE COMBINING MODULE

This unit accepts eight inputs and assigns any input to any of nine outputs in any combination by means of a nine-position push-push switch. The ninth position permits assigning the output to a solo channel. The signals from the assignment switch are sent to the input of the operational amplifiers for mixing. Channel isolation exceeds 70 dB. Two extra summing inputs are provided on each amplifier; the first allows high level paralleling of matrix switch modules, and the second allows simultaneous slating of all channels. Two versions of this module are in turn available; the first as described above and the second includes for each channel an additional double push-push switch and associated amplifiers allowing two-channel echo assignment. This module has unity gain and is phase normalized to allow patching on both sides of the module. A functional diagram is shown in FIGURE 3. Size of this module is 7-in. high x 19-in. wide x 7½-in. deep.

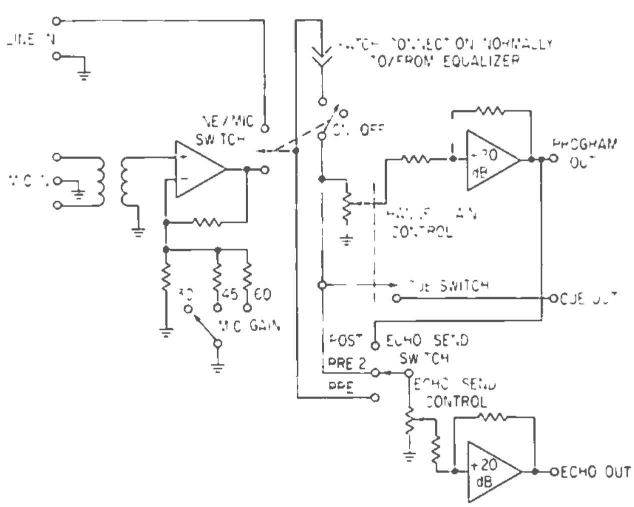

Figure 1. A block diagram of the input module described in the text Only one of the eight sections is shown.

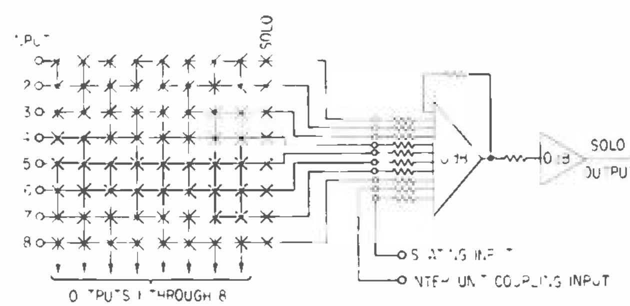

Figure 3. The 8 x 8 matrix switcher and active combining module. Only one channel is shown. For simplicity the grounding of inputs in off position is not shown. All switches are push-push for multiple assignments.

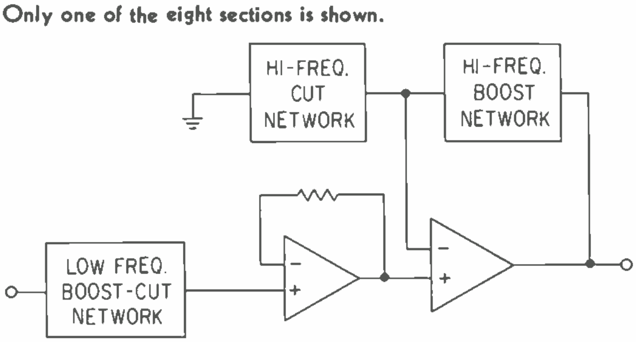

Figure 2. One of the eight sections of the equalizer module.

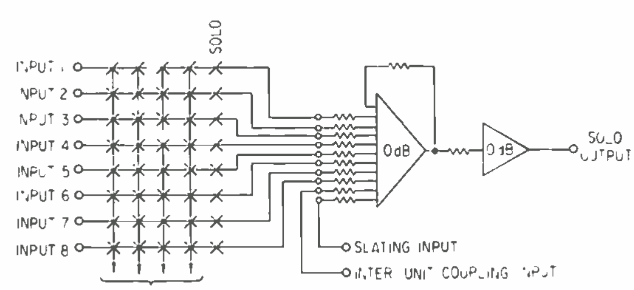

Figure 4. The 8 x 4 switcher. This unit is otherwise similar to that shown in Figure 3.

8 x 4 MATRIX SWITCHER AND ACTIVE COMBINING MODULE

This unit accepts eight inputs and assigns any input to any of five outputs by means of a five-position push-push switch. This unit is intended for use as a 4-channel echo switcher in 8- or 16-track systems or as a channel switcher in 4-track systems. As in the 8 x 8 switcher all mixing is accomplished by i.c. opamps and the fifth position is intended for a solo function. See FIGURE 4

An optional version is available having an additional double push-push switch for two channel echo assignment when used in a four track system. Size of this module is 5¼-in high x 19-in. wide x 7¼-in. deep.

8 x 2 MATRIX SWITCHER

This unit is similar to the matrix modules described above except being limited to 8 x 2 switching and combining. Size is 3½-in. high x 19-in. wide and 7¼-in. deep. This unit is useful for two-channel matrix echo switching in small consoles.

MASTER CONTROL UNIT

This unit is intended for combining the eight-channel information into stereo and mono signals for monitor or re-recording purposes. The eight program input signals can be panned individually between either of the two stereo output channels. Four echo-return signals can be level controlled and panned between the two stereo output signals. Stereo output level setting is accomplished with slide pots. A derived mono signal is available and has its own slide pot for level control. Provision is incorporated to accept signals directly from spring-type echo devices as well as high-level echo devices. FIGURE 4 shows the functional diagram of the complete control module. This module is 7-in. high by 19-in. wide by 7½-in. deep. It also has application in existing consoles where it is desired to add panning and echo facilities. See FIGURE 5.

All modules are normalled to have 0-degree phase shift in order that patching can be installed anywhere in the system without phase-reversal problems.

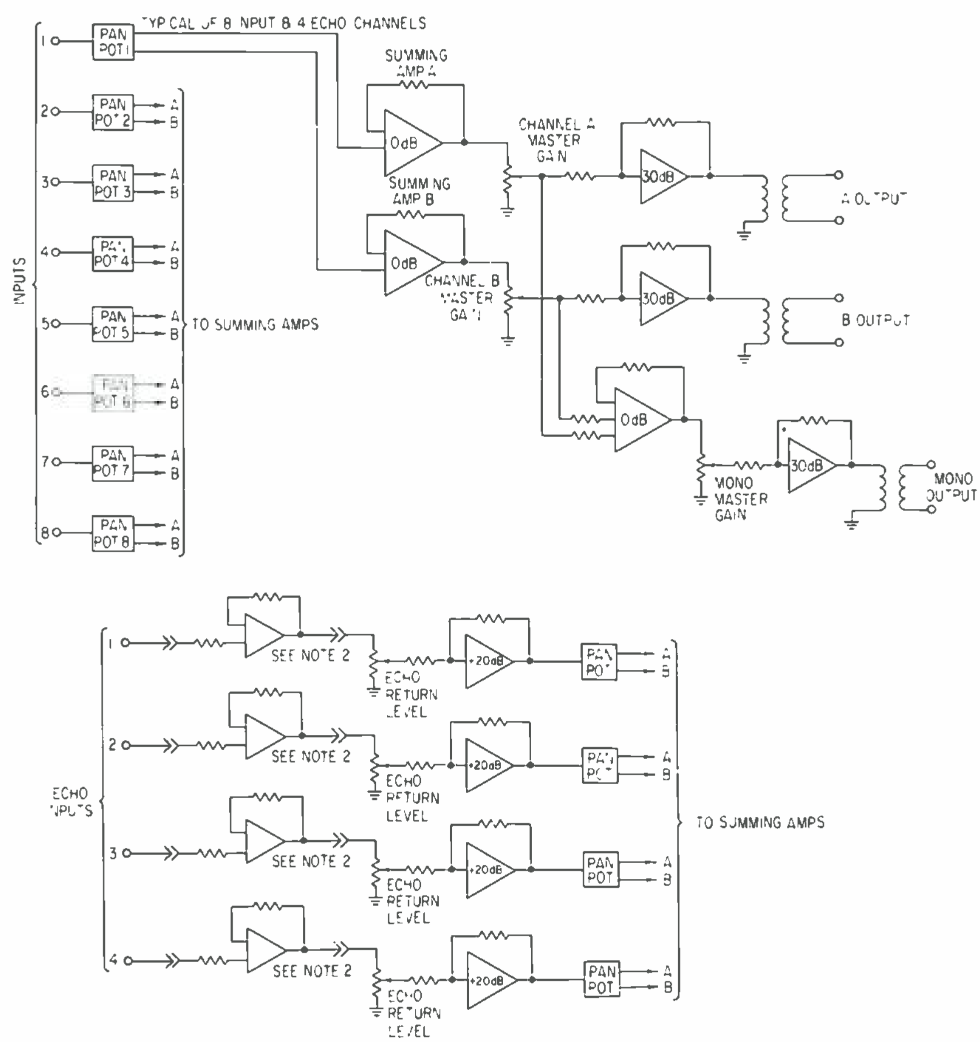

Figure 5. The control module. Only connections from pan-pot 1 to the summing amplifier are shown for simplicity, all pan-pots being identical. Provision is incorporated for echo return booster amplifier for use with low-level echo devices. High-level devices bring the signal directly to the echo-return pot.

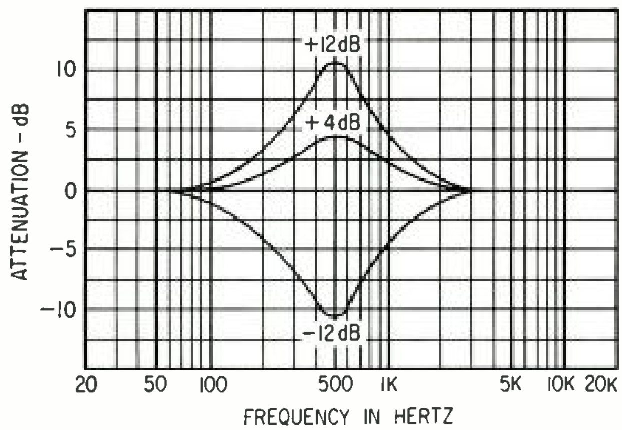

Figure 6. Typical equalizer high-frequency curves. Other frequencies and attenuation or boosts are similar.

Now that we have described the design criteria and how the various modules were packaged to meet these criteria, let us explore some typical console application problems and how they could be solved using these building blocks.

TYPICAL APPLICATIONS

People doing remote recordings, small studios doing stereo and mono recordings, broadcast station production studios and studios doing mix downs of 4- and 8-channel recordings require a small console meeting the following requirements: 1) 8 inputs switchable between line and microphone. 2) echo send on each input assignable to either of two echo buses. 3) equalization on each input channel 4) signal panning 5) reduction of the multichannel input information to stereo or mono signals with echo return addition.

Such a system could be built using the described building-block modules. Input signals would be processed and equalized using the input module and equalizer. 8 x 2 matrix switcher and active-combiner module derives the two channel echo-send signals, and a control module handles signal panning and mixing as well as the audition of echo-return signals. This combination of modules would occupy an area of 25¾-in. high x 19-in. wide not including the meter panel which would add 3½ in. to the height. Maximum depth would be 7½ in. Packaging could be accomplished by mounting a standard 19-in. rack in a portable case for remote work. a table-top console or a free-standing console.

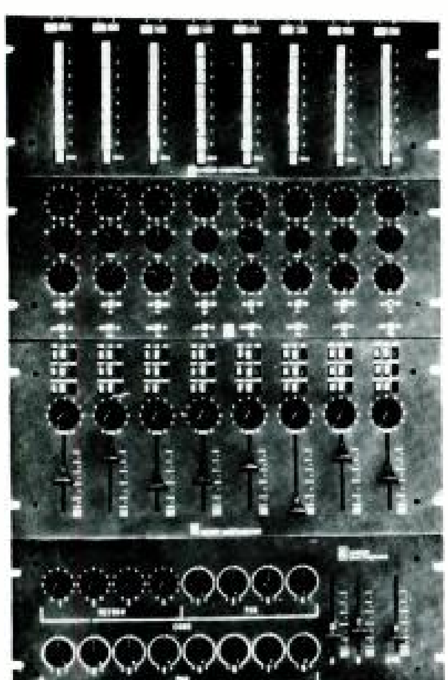

Figure 7. The four basic modules described in the text. From top to bottom they are—the matrix switcher and active combining network, the equalizer module, input module, and control module.

The recording world is constantly changing and as a result it is not uncommon for remote recorders and small studios to work in the 4-track medium in order that mixing and balance decisions can be made at a leisurely mix-down session rather than under the pressure of the original session. A further advantage of the 4-track medium is its ability to work in the sel-sync mode allowing corrections or additions to previously accomplished work.

Such applications can be met by a console or portable mixing system having eight line/mic inputs, equalization, 8 x 4 matrix program switching, 8 x 2 echo matrix switching, and panning of the input signals. Outputs would include four to the recorder, two to the echo device, panned and mixed stereo, and mono output signals.

Such a system could be easily fabricated using a Series Eight input module, a equalizer module, a matrix switcher with the optional 8 x 2 echo switching, and a Control Module.

8-track studios have requirements for eight, sixteen-, or even twenty four-input consoles having eight outputs to the recorder as well as mixed-down mono and stereo outputs. These consoles frequently have more extensive echo requirements so that more than one type of echo device can be driven simultaneously.

The simplest console for such an application would have eight inputs, four echo buses and eight- two- and one-channel outputs.

This relatively simple console can be fabricated using a Series Eight input module and an equalizer module to handle the input signal chores. An 8 x 8 matrix switcher and combiner would derive the signals to the recorder. An 8 x 6 matrix switcher and combiner would be used to derive the echo-send signals. A control module would handle the panning and mixing requirements. This equipment would mount in an area 19¼-in. high by 38-in. wide (alternate 35-in. high by 19-in. wide) and could be rack mounted or built into a table top or free-standing console. Expansion to sixteen or even twenty-four inputs can be handled by adding additional input, equalizer, and 8 x 8 and 8 x 4 matrix switcher modules.

SYSTEM GROWTH

The typical system such as first described above can grow to become the second system or even the most complex version of the third by merely adding additional input, equalizer and matrix combining networks. Thus it can be seen that it is possible to start with a very simple eight input, stereo or mono output console and have it grow to become a twenty-four input, eight-output recording console with stereo and mono mix down facilities with full echo facilities. It should be further noted that the associated wiring problems of adding or rearranging the console to meet new requirements are minimized to simple module interconnections as the eight element building block modular concept eliminates for the console builder much of the inter-component wiring. Since all inter-module signals are low impedance and at line-level, potential problems with hum, noise and r.-f. interference are virtually eliminated. ■

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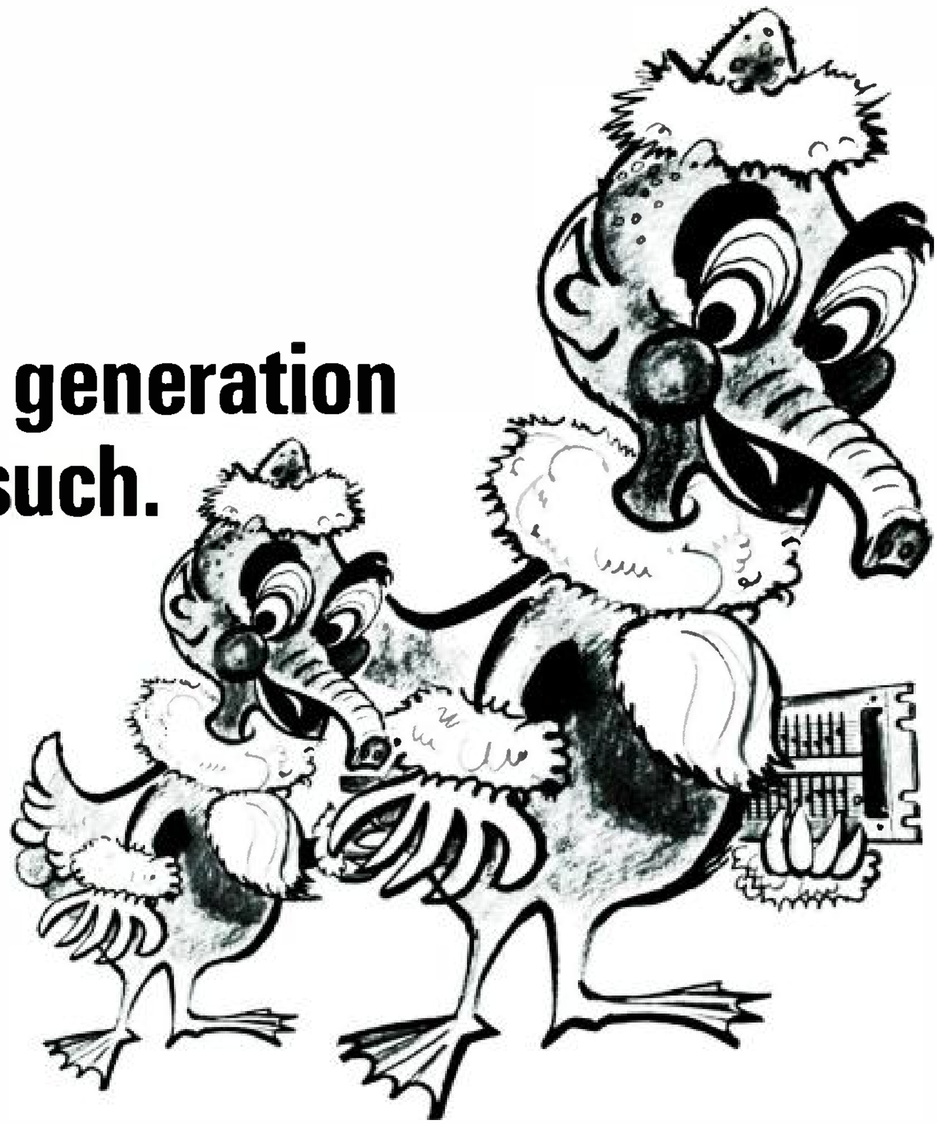

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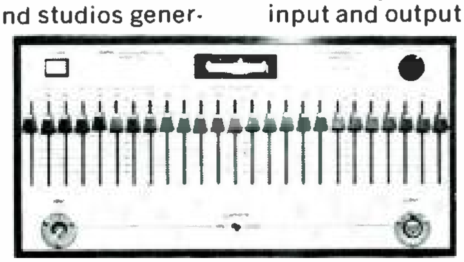

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Low-Frequency Sound Absorbers

MICHAEL RETTINGER

The control of low-frequency reverberation and accentuation in small studios can be a most vexing problem. The author shows a number of methods of combat that may be used.

IT IS WELL KNOWN that small rooms exhibit boominess or an accentuation of low-frequency notes generated therein. This is due to the fact that in small enclosures (say, less than 10,000 cu. ft.) there exist large frequency intervals between the low-order normal modes. Thus, when the bass component of a complex musical note coincides with a mode, the room will resonate to it and the pitch will become intensified.

For sound-recording studios, monitoring rooms, and reverberation chambers it is desired to restrict the low-frequency reverberation to achieve a flat reverberation characteristic, or variation of reverberation time with frequency. Because most commercial acoustic materials exhibit an absorptivity at 1000 hertz materially greater—sometimes several times greater—than at 100 hertz, a special compensatory sound-absorptive measure is required for such rooms. It is the purpose of these pages to discuss several ways and means of constructing sound-absorbers which are more effective for the low than the high registers.

Membrane Absorbers. The fundamental frequency of a thin, pliable, square sheet is given by

$$f_o = \frac{0.705}{a}\sqrt{\frac{t}{m}} = \frac{13.8}{A}\sqrt{\frac{T}{M}}$$

where t and T are the membrane tensions in dynes/cm and lb/inch respectively; m and M are the surface densities of the membrane in g/sq. cm. and lb. sq. ft.; and a and A are the lengths of the sides of the membrane in cm. and ft.

Thus a 0.015 inch-thick sheet of plastic, weighing 0.09 lb. sq. ft. arranged in a one-foot-square frame, with a tension of 2 lb./inch, will have a fundamental frequency of

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Michael Rettinger is a consultant on acoustics. He has written many articles and books on the subject.

$$f_o = \frac{13.8}{1}\sqrt{\frac{2}{0.09}}$$
$$= 65 \text{ hertz}$$

This type of membrane is quite closely approached by oil paintings—tightly and uniformly stretched layers of canvas with a more or less sound-opaque surface. Indeed, art galleries are often characterized by a rather flat reverberation characteristic, while circus tents have practically no low-frequency reverberation.

Spaced Screen. This term is meant to represent a sound-transparent layer, like a hung curtain, placed at a distance from a rigid wall. When the space between sheet and wall is a quarter-wavelength or multiple thereof, maximum absorption results; when the distance is a half-wavelength or multiple thereof, minimum absorption occurs. Thus a curtain spaced two feet from a reflective wall is highly absorbent at $1128/4\times2 = 141$ hertz, but little absorbent at 282 hertz, so that its absorption characteristic is beset by a series of peaks and dips extending to the very high registers.

When a membrane is spaced at a distance from a wall, the stretched sheet may absorb sound not only at its fundamental frequency as explained in (1) above, but also because of the distance effect described here. The effect is due to the fact that here, at a quarter-wavelength from the rigid wall, maximum air-particle velocity occurs for this frequency. As a sound wave strikes the spaced screen, the pressure pulse passes readily through the curtain, because of its porous nature. When it impinges on the massive barrier it is almost totally reflected due to the large difference between the speed of propagation of sound in the hard material compared with the speed of sound in air. As a result of this very efficient reflection, a sound-pressure maximum occurs at the wall surface. Also, since the wall does not move, the air-particle velocity near the wall is a minimum. The nature of compressional wave motion, however, is such that a velocity maximum must occur at a location one-fourth of a wavelength removed from a velocity minimum, just as a pressure minimum must occur at one-quarter wavelength distance from a pressure maximum. When a low-frequency sound wave, say 141 hertz, strikes the wall, the air-particle velocity at the wall is essentially zero, while at a distance two feet from the wall (where the curtain is located), it is maximum. Since for high sound absorption the movement of the air within the volume occupied by the porous material must be great, practically all acoustic products are efficient low-frequency absorbers only when they are spaced at a relatively great distance from a rigid barrier. This phenomenon also explains the comparatively high sound absorption of so-called integrated or suspended acoustic ceilings.

Spaced Panel. When a sound-opaque panel is spaced at a distance from a hard wall, the air between panel and wall acts as a spring to which a mass (the panel) is fastened. Such a mechanical system has a fundamental frequency given by

$$f_o = \frac{170}{\sqrt{MD}}$$

where M = surface density of panel, lb./sq. ft.

D = distance between panel and rigid backing.

Thus a 3/8 inch-thick plywood panel, weighing 0.9 lb. sq. ft., and spaced one inch from a wall, has a fundamental frequency of $179/(0.9)^{1/2}$ = 179 hertz.

FIGURE 1 shows the relationship between resonant frequency and plywood panel thickness for several air spaces between panel and wall.

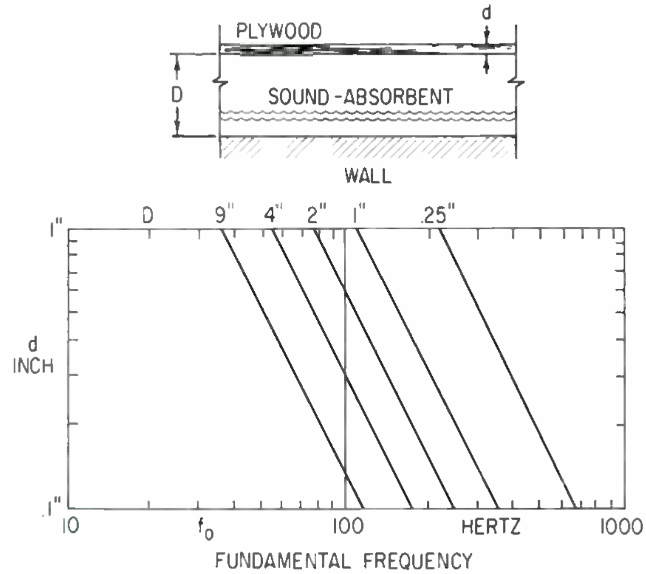

Figure 1. The resonant frequency of a plywood panel as a function of its thickness, for various wall spacings.

As in the case of a stretched membrane, a spaced panel absorbs sound energy through its internal viscous damping, whereby some of the acoustic energy is converted into heat. Because diaphragm excursion is greatest at resonance, energy conversion is maximum at this frequency.

Perforated Panel. FIGURE 2 shows the fundamental frequency of a perforated panel as a function of the product of its thickness and wall spacing, for various amounts of open surface. The apertures in the board, together with the air-space behind the panel, constitute a Helmholtz resonator.

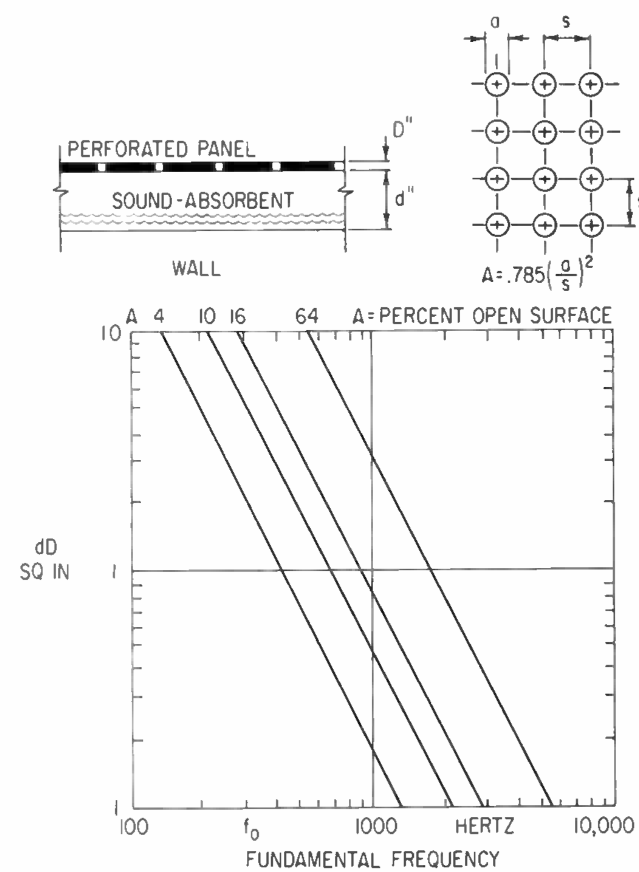

Figure 2. The resonant frequency of a perforated panel as a function of the product of its thickness and wall spacing, for various amounts of open surface (surface porosity).

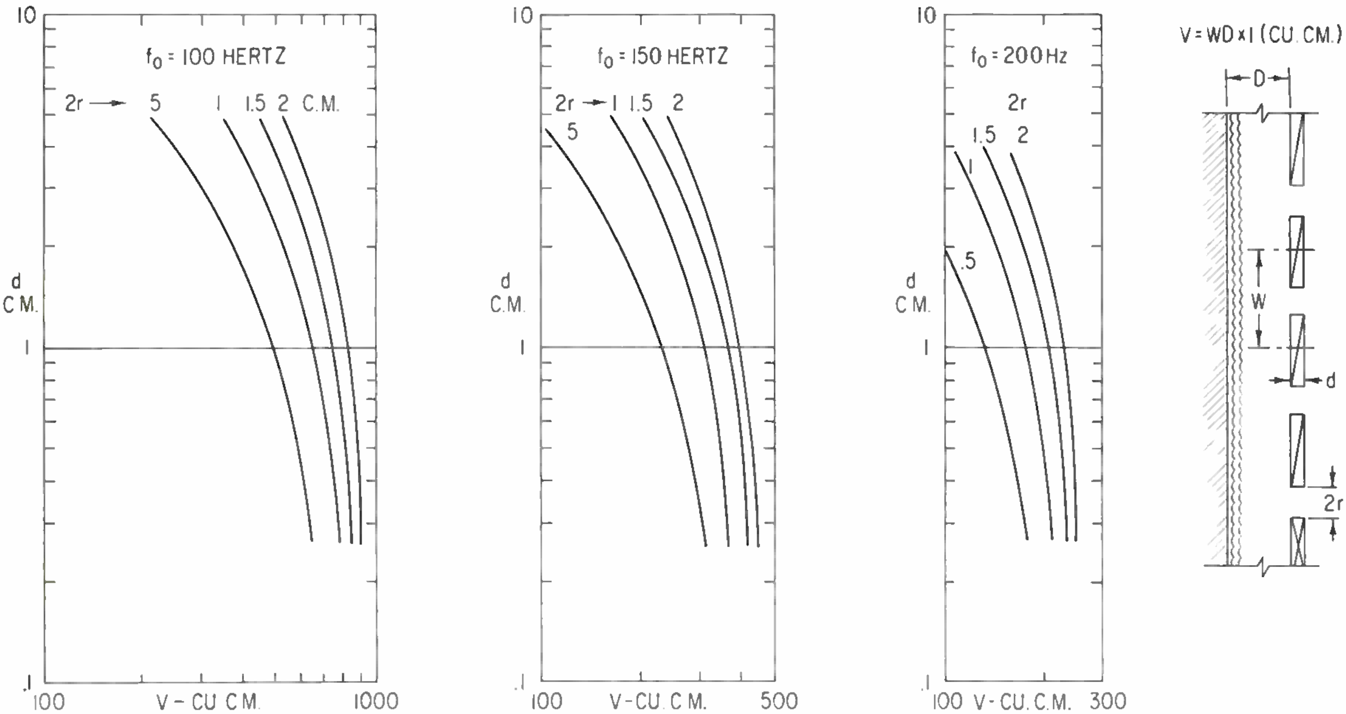

Figure 3. A design chart for slat absorbers.

with a neck length equal to the thickness of the sheet. The equation for the resonant frequency of the perforated acoustic treatment is given by

$$f_o = 2130\sqrt{\frac{A}{dD}}$$

where A = percent open surface

$$= \left(\frac{a}{s}\right)^2 \frac{\pi}{4}$$

$$= 0.785 \left(\frac{a}{s}\right)^2$$

a = hole diameter, inches
s = spacing between hole centers, inches
D = panel thickness, inches
d = depth of air-space behind perforated panel.

Note that for any given hole diameter, the resonant frequency is proportional to the number of holes per unit area, and inversely proportional to both panel thickness and depth of air-space. Thus, to achieve a low resonant frequency—say, 150 hertz—with the usual 1/8-in. thick Masonite panel with its 10 per cent open surface (A = 0.10), the depth of the required air-space behind the panel comes to 162 inches. By making the panel 1-in. thick and employing only 1 per cent open surface (1.44 sq. in./sq. ft.), the air-space can be made as little as 2 inches.

An acoustic disadvantage connected with perforated panels of this sort is the high-frequency echoes produced by large sheets when an impulsive sound source is located in front of them, and the flutter echoes produced when such a source is placed between parallel sheets of this kind. This is due to the anti-resonant frequencies of such a panel, for which frequencies the absorption becomes a minimum. These *holes* are not readily evident in the absorption characteristic of such a construction when it is measured in the usual manner in a reverberation chamber at discrete frequencies. The reason for it is that in such a room the sound strikes the material at all possible angles (random sound incidence), so that the effect of the anti-resonant frequencies, dependent as they are on the angle of sound incidence, becomes little noticeable. The detection of these holes is made more difficult by the fact that none of the test frequencies may coincide with any of the anti-resonant frequencies of the perforated panel. In the case of a handclap, however, originating in front of a panel, the sound strikes the panel at normal incidence, and the return of the pertinent signal component becomes readily audible as an echo or echoette.

Slat Absorbers. This type of absorber may be constructed with slats or battens—generally 1 × 3 inch boards—spaced from 0.1 inch to 0.5 inch or farther apart, with an air-space of 1 to 10 inches behind the slats, with the space often containing a generous amount of sound-absorbent material.

FIGURE 3 contains design information for this type of low-frequency absorber. As an example, consider 1 × 3 inch (2.5 cm × 7.5 cm.) slats spaced 0.2 inch (0.5 cm.) apart, so that d = 2.5 cm. (1 inch), W = 7.7 cm. (3 inches), 2r = 0.5 cm. (3/16 inch). For a resonant frequency of 150 hertz, the volume behind a set of slats, per centimeter length of slat, comes to 150 cu. cm, as shown in the center part of the figure. This means that the air-space, D, behind the slats should be 150/7.7 = 19.5 cm. = 7.7 inches.

This type of low-frequency sound-absorber has proven very effective in disk and motion-picture scoring stages and monitoring rooms, and wherever it has been deemed desirable to have a nearly flat reverberation characteristic. A refinement in the construction consists in applying the slats not as a flat surface but as a cylindrical surface, as when the slats are nailed against space circular wood saddles to form vertically slotted, cylindrical splays.

In FIGURE 3, the volume per unit length of absorber, that is, per centimeter along its length or height, is given by

$$V = WD \times 1 \text{ cu. cm.}$$

$$= \frac{3 \times 10^7}{f_o\left(\frac{d}{2r} + 6.62 - 1.462 \log 2rf_o\right)}$$

where f_o = resonant frequency
d = slat thickness, cm
W = center-to-center distance between slats, cm.
= w ¾ 2r
w = slat width, cm.
D = depth of air space behind slats, cm. ■

THROUGH THIS DOOR

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38th AES Convention and Exhibition

QUICK SUMMARY

LOS ANGELES HILTON HOTEL

SUNDAY, MAY 3, 10:00 A.M. Exhibitors Breakfast
Los Angeles Room

SUNDAY, MAY 3, 5:00-7:00 P.M. No-Host Welcoming Cocktail Party
Sierra Room

REGISTRATION

Monday, May 4—8:00 A.M. to 5:00 P.M.
Tuesday, May 5—9:00 A.M. to 8:00 P.M.
Wednesday, May 6—9:00 A.M. to 5:00 P.M.
Thursday, May 7—9:00 A.M. to 8:00 P.M.

EXHIBIT HOURS

Pacific, Wilshire and Garden Rooms

Monday and Tuesday, May 4 & 5	1:00 P.M. to 9:00 P.M.
Wednesday and Thursday, May 6 & 7	11:00 A.M. to 5:00 P.M.

DEMONSTRATION ROOMS

Mission, Cleveland, Washington, Detroit, Boston, St. Louis, and Foy Rooms

TECHNICAL SESSIONS

Golden State Room: Sessions A, C, E, F, G, J, K, L, M, and O
Los Angeles Room: Sessions B, D, H, N, and P

Monday,	**May 4—9:30 A.M.**	A & B	
	2:00 P.M.	C & D	
	No sessions Monday evening		
Tuesday,	**May 5—9:30 A.M.**	E	
	2:00 P.M.	F	
	7:30 P.M.	G & H	
Wednesday,	**May 6—9:30 A.M.**	J	
	2:00 P.M.	K	
Social Hour	**—7:00 P.M.**	Los Angeles Room	
Awards Banquet	**—8:00 P.M.**	Golden State Room	
Thursday,	**May 7—9:30 A.M.**	L	
	2:00 P.M.	M	
	***2:30 P.M.**	N	
	7:30 P.M.	O, P and Q	

*Note late starting time.

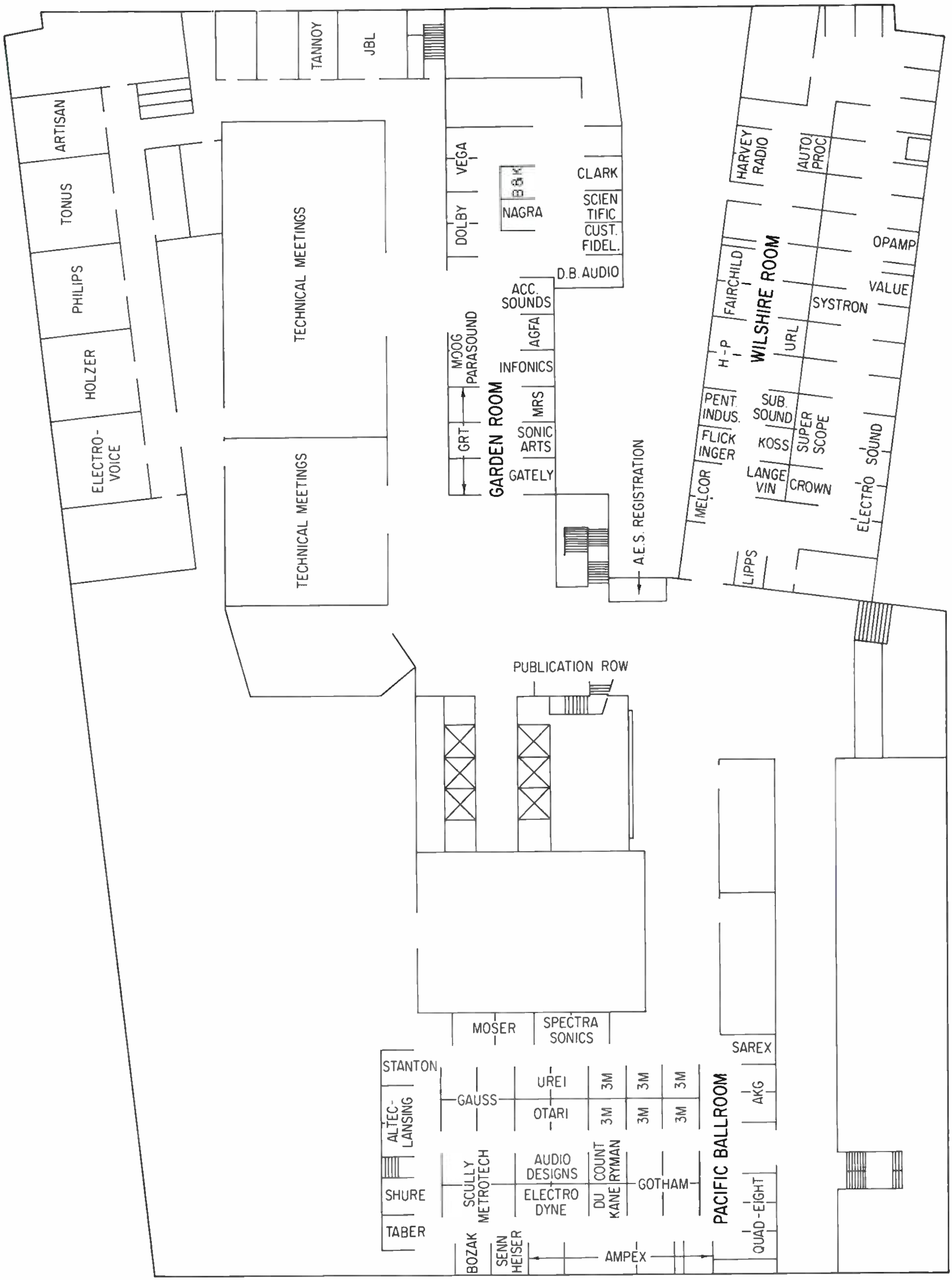
Session Q, Recording Studio Workshop, will be held in several recording studios in the Hollywood area. Registrants who have made reservations for the workshop will board buses at the 7th Street entrance to the hotel on the lower lobby at 6:30 P.M.

LADIES ACTIVITIES

Many and varied activities have been arranged for the ladies. A separate sheet describes these in detail. Ladies may join the hostess and her committee in the New York Room at 9:00 A.M. each day for coffee and sweet rolls before commencing the day's activities.

Ladies Committee

Gail (Mrs. Hugh) Allen, Gotham Audio Corporation
Jeri Boyd, Langevin
Jane Geier, The Jane Geier Co.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ELECTRO DYNE
COUNT RYMAN
DU KANE
GOTHAM
PACIFIC BALLROOM
SHURE
TABER
BOZAK
SENNHEISER
AMPEX
QUAD-EIGHT

TECHNICAL SESSIONS

Monday, May 4, 1970—9:30 A.M.
Golden State Room

MOTION PICTURE SOUND TECHNIQUES

Chairman: JOSEPH D. KELLY
Glen Glenn Sound, Hollywood, California

A New Production Sound Dolly and Automated Transfer Unit—Elliot Bliss, CBS Studio Center, Studio City, California

Re-Recording Process—James G. Stewart, Glen Glenn Sound Co., Hollywood, California

The Sound Recording Console—Barry K. Henley, Glen Glenn Sound Co., Hollywood, California

Film Recording Equipment, as Installed at the American Zoetrope Company—San Francisco, California—K. Kenneth Miura, Dept. of Cinema, University of Southern California, Los Angeles, California

A New Sprocket Driven Audio Recorder/Reproducer—Donald R. Collins, Tele-Cine Inc., New York, New York

An Electronic Looping System—Otto Popelka and Norman Prisament, Magna-Tech Electronic Company, Inc., New York, New York

Monday, May 4, 1970—9:30 A.M.
Los Angeles Room

ACOUSTICAL NOISE AND NOISE CONTROL

Chairman: KENNETH M. ELDRED
Wyle Laboratories, El Segundo, California

Some Problems and Successes in Controlling Noise Exposure in California Industry—William W. Steffan, Division of Industrial Safety, State of California, San Francisco, California

A Systems Approach to Aircraft Noise Control—Daniel W. Emory, Daniel W. Emory & Associates, Newport Beach, California

The Motor Vehicle Noise Problem and What is Being Done About It—Ross A. Little, Engineering Section, California Highway Patrol, Sacramento, California

Measurement of Traffic Noise on Connecticut Highways—Gerald A. Budelman and Edward J. Foster, CBS Laboratories, Stamford, Connecticut

Needs and Specifications for Audio Equipment Used in Psychoacoustic Work—Lawrence E. Langdon, McDonnell Douglas Corporation, Douglas Aircraft Co., Long Beach, California

Monday, May 4, 1970—2:00 P.M.
Golden State Room

DISC RECORDING AND REPRODUCTION

Chairman: STEPHEN F. TEMMER
Gotham Audio Corporation, New York, N.Y.

Development and Application of a New "Tracing Simulator"—Dieter Braschoss, Georg Neumann, Electro-acoustic GmbH, Berlin, Translated and read by Stephen F. Temmer, Gotham Audio Corporation, New York, N. Y.

Interaction Between Tracing and Deformation Errors—Duane H. Cooper, University of Illinois, Urbana, Illinois

An Evaluation of the Forces Required to Move a Tone Arm—John J. Bubbers, Stanton Magnetics Inc., Plainview, New York

Maximum Levels in the Record/Playback System—Arnold Schwartz, Micro-Point, Inc., White Plains, New York

The Compatible Stereo Generator and its Application to All Stereo Media—Howard S. Holzer, Holzer Audio Engineering Corporation, Van Nuys, California

Monday, May 4, 1970—2:00 P.M.
Los Angeles Room

AUDIO IN AM, FM, AND TV BROADCASTING

Chairman: RICHARD W. BURDEN
Richard W. Burden Associates, Mt. Kisco, New York

Transmission of Additional Aural Channels on Television Carrier—John A. Moseley, Moseley Associates, Inc., Goleta, California

Report on Possible Multiplex Methods for the Transmission of Four Channel FM Stereo—W. S. Halstead, Multiplex Development Corp. and RTV International, New York, N. Y.

A Review of Program Level Indicating Systems—John G. McKnight, Ampex Stereo Tapes Division, Redwood City, California

Readout Devices Other Than the Standard VU Meter as a Better Means of Measuring Peak Levels—Leroy C. Granlund, Western Broadcast Services, Sunnyvale, California

Panel Discussion—A Review and Discussion of the Problem Areas of Peak Levels and Loudness Control and Measurement—John G. McKnight; Leroy C. Granlund; Bernard Katz, B & K Instruments; Arno Meyer, Belar Electronics Lab. Moderator: Richard W. Burden

Microphone Recordings for Radio and TV When Loudspeaker Equipment is Simultaneously Used for an Audience—Ernst-Joachim Voelker, Stierstadt/Taunus, West Germany

Tuesday, May 5, 1970—9:30 A.M.
Golden State Room

MICROPHONES AND PLAYBACK CARTRIDGES

Chairman: ROBERT W. CARR
Shure Brothers, Evanston, Illinois

Miniature Electret Microphones—Freeman W. Fraim and Preston V. Murphy, Thermo Electron Corporation, Waltham, Massachusetts

Third Order Gradient Microphone for Speech Reception—B. R. Beavers and R. Brown, LTV Research Center, Anaheim, California

Experimental Wide Bandwidth Tooth Contact Microphone—Austin J. Brouns, LTV Research Center, Anaheim, California

Microphone Accessory Shock Mount for Stand or Boom Use—Gerald W. Plice, Shure Brothers, Evanston, Illinois

Closing the Wireless Versus Wired Microphone Dependability Gap—Barry M. Kaufman, Vega Electronics, Santa Clara, California

Bi-Radial and Spherical Stylus Performance in a Broadcast Disc Reproducer—J. R. Sank, RCA, Camden, New Jersey

New Directions in Microphone Placement—James Cunningham, 8-Track Recording Company, Chicago, Illinois

Tuesday, May 5, 1970—2:00 P.M.
Golden State Room

LOUDSPEAKERS

Chairman: RICHARD C. HEYSER
Jet Propulsion Laboratories, Pasadena, California

Loudspeaker Measurement Techniques—Charles L. McShane, Acoustic Research, Inc., Cambridge, Massachusetts

Some Observations and Speculations on the Role of Speakers in Stereophonic Reproduction—Joel C. Finegan, 3M Co., St. Paul, Minnesota

The Inter-Relationship of Cabinet Volume, Low Frequency Resonance and Efficiency for Acoustic Suspension Systems—Joel C. Finegan, 3M Co., St. Paul, Minnesota

Acoustical Circuits Revisited—Robert Howard, James B. Lansing Co., Los Angeles, California

Time Delay Distortion in Multi-Speaker Loud-Speaker Systems—Martin Gersten, Rectilinear Research Corp., New York, New York

Wisdom and Witchcraft of Old Wives Tales About Woofer Baffles—J. Robert Ashley and Thomas A. Saponas, University of Colorado, Colorado Springs, Colorado

Tuesday, May 5, 1970—7:30 P.M.
Golden State Room

ELECTRONICS APPLIED TO MUSIC

Chairman: JODY C. HALL
Thomas Organ Company, Sepulveda, California

Techniques of Generating and Gating Source Signals in Modern Electronic Organs—Allan E. Winsberg, Thomas Organ Company, Sepulveda, California

The Electronic Piano—Harold Rhodes, CBS Musical Instruments, Fullerton, California

Changing Pitch and Timbre of Woodwind Instruments by Electronic Means—Bradley Plunkett, United Recording Electronics Industries, North Hollywood, California

A Ring Modulator Device for the Performing Musician—Thomas E. Oberheim, Oberheim Electronics, Inc., Santa Monica, California

The Use of Buchla Synthesizer in Musical Composition—Morton Subotnick, Consultant, CBS Musical Instruments, Fullerton, California

Demonstration of the Practical Application of Electronics in Music—Martin Subotnick, Allan Winsberg, Bradley Plunkett, Thomas Oberheim, Harold Rhodes, and Jody C. Hall, Moderator

Tuesday, May 5, 1970—7:30 P.M.
Los Angeles Room

AUDIO MEASUREMENTS AND INSTRUMENTATION

Chairman GERALD G. GROSS
Hewlett-Packard Corporation, Palo Alto, California

An Improved Field Corrector for Free-Field Microphone Calibrations—Edward J. Foster, Louis T. Fiore, and Benjamin B. Bauer, CBS Laboratories, Stamford Connecticut

Simplified Spectral Analysis by Use of a Band Limited Random Noise Test Record—Robert R. Beavers, Altec-Lansing, Anheim, California

Impulse Responsive Adapter for Chart Recorder—Edward J. Foster and Benjamin B. Bauer, CBS Laboratories, Stamford, Connecticut

Acoustic Impedance Calibrator for Mask and Microphone Measurements—A. J. Brouns and C.T. Morrow, LTV Research Center, Anaheim, California

The Measure Flutter in Audio Tape Record/Reproduce Machines—R. A. Christner, Date Measurements Corporation, Palo Alto, California

Precision Sound Level Recording System for Industrial Environments—Gerald G. Gross and Wolfgang Glietsch, Hewlett-Packard, Palo Alto, California

Wednesday, May 6, 1970—9:30 A.M.
Golden State Room

ARCHITECTURAL ACOUSTICS AND ELECTROACOUSTICS

Chairman: MICHAEL RETTINGER
Consultant on Acoustics, Encino, California

Noise: The New Pollutant, Motion Picture Produced by the National Educational Television Network With a Grant From the Acoustical Materials Association—NET Film Service, Indiana University, Bloomington, Indiana

An Historical and Architectural Review of Opera Halls of the World—Wilfred A. Malmlund, Bolt, Beranek and Newman, Inc., Van Nuys, California

Microphone Thermal Agitation Noise—Harry F. Olson, David Sarnoff Research Center, RCA Laboratories, Princeton, New Jersey

Planning of U. S. Air Force Audio Visual Center, Norton Air Force Base, San Bernardino, California—Robert W. Houts, Sound Services Division, AF Audio Visual Center, Norton Air Force Base, California, and Michael Rettinger, Consultant on Acoustics, Encino, California

Wednesday, May 6, 1970—2:00 P.M.
Golden State Room

SIGNAL CONTROL AND PROCESSING

Chairman: WILLIAM P. BRANDT,
Altec Lansing, Anaheim, California

Design Philosophy in the Construction of Multichannel Portable Mixing Consoles—George Alexandrovich, Sr., Fairchild Sound Equipment Corporation, Long Island City, N. Y.

Modules. . .Why?—Oliver Berliner, SounDesign Engineers, Beverly Hills, California

When is Phase Shift Objectionable?—Robert A. Bushnell, Bushnell Electronics Corporation, Van Nuys, California

Electronic Adjustment of Monitoring Acoustics—Daniel N. Flickinger, Elektracoustics Division, Daniel N. Flickinger and Associates, Inc., Hudson, Ohio

The Stereo Synthesizer and Stereo Matrix: New Techniques for Generating Stereo Space—Robert Orban, Kurt Orban Co., Inc., East Palo Alto, California

The Disclosure of Hidden Information in Sound Recording—E. Roerbaek Madsen, Bang & Olufsen A/S, Struer, Denmark

Thursday, May 7, 1970—9:30 A.M.
Golden State Room

MAGNETIC RECORDING AND REPRODUCTION

Chairman JOHN T. MULLIN
Mincom Division, 3M Co., Camarillo, California

A New Rotary Turret Head Mounting System for Multiple Track Configurations—Pat Tobin, Program Dynamics, Inc., Los Angeles, California

A Standard Vocabulary for Audio Tape Duplicators—Haskell M. Metz, Otari of America, Ltd., Inglewood, California

Development of a New Magnetic Tape for Music Mastering—Delos A. Eilers, 3M Company, Magnetic Products Division, St. Paul Minnesota

Measurements of Mechanical Properties of Magnetic Tape—Robert A. Finger, Patrick Murphy, and Edward J. Foster, CBS Laboratories, Stamford, Connecticut

A Drop-Out Perceptibility Counter—Edward J. Foster, CBS Laboratories, Stamford Connecticut

Specifications for Magnetic Recording and Reproducing Heads, and Tapes—John G. McKnight Stereo Tapes Division, Ampex Corp., Redwood City, California

Musicassette Interchangeability: The Facts Behind the Facts—E. R. Hanson, North American Philips Corporation, New York, N. Y.

Thursday, May 7, 1970—2:00 P.M.
Golden State Room

MUSIC, SPEECH AND HEARING

Chairman: LEO L. BERANEK
Bolt, Beranek, and Newman, Inc., Cambridge, Massachusetts

Growth of Vocal Output—Mark B. Gardner, Bell Telephone Laboratories, Inc., Murray Hill, New Jersey

Application of Rating Scales to Quantify subjective Evaluations for a Group of Concert Halls—B. G. Watters, R. Johnson, R. L. Kirkegaard; Bolt, Beranek and Newman, Inc., Cambridge, Massachusetts

The Stimulation of Moving Sound Sources—John Chowning, Dept. of Music, Stanford University, Stanford, California

A Demonstration of Moving Sound Sources—John Chowning Dept. of Music, Stanford University, Stanford California

Determination of an Effective Tone Ring Signal—Richard M. Hunt, Bell Telephone Laboratories, Inc., Indianapolis, Indiana

Simulating Jet Aircraft Flyover Noise for Subjective Judgments—Karl S. Pearsons; Bolt, Beranek and Newman, Inc., Van Nuys, California

Thursday, May 7, 1970—2:30 P.M.
Los Angeles Room

AMPLIFIERS AND AUDIO CIRCUITRY

Chairman: JOHN P. JARVIS
Consultant, Northridge, California

Audio Engineering and the Publications Group—Charles R. Horton, Altec Lansing, Anaheim, California

Eliminating RF From Audio Systems—Paul E. Gregg, Bauer Broadcast Products, Granger Associates, Palo Alto, California

Operational Amplifier Implementation of Ideal Crossover Networks—J. Robert Ashley and Lawrence E. Henne, University of Colorado, Colorado Springs Center, Colorado Springs, Colorado

A Low Noise Approach to the Mixer Stage Amplifier—P. B. Spranger and J. Pritchett, Altec Lansing, Anaheim, California

A Gain-Reduction Amplifier that Employs a Junction Field-Effect Transistor as an Active Element of a Resistive Divider—John P. Jarvis, Northridge, California

Thursday, May 7, 1970—7:30 P.M.
Golden State Room

SOUND REINFORCEMENT

Chairman: HERBERT M. JAFFE
Atlas Sound Division, American Trading and Production Corp., Parsippany, N. J.

Design of a High Quality Public Address System for Aircraft Use—Allan J. Rosenheck; Bolt, Beranek and Newman, Inc., Van Nuys, California and James D. Kronman, Lockheed California Company, Burbank, California

Acoustical Treatment and Sound Reinforcing Systems for the Washington State Legislature—Herbert T. Chaudiere, Robin M. Towne and Associates, Seattle, Washington

The Design and Testing of Various Sound Reinforcement systems for the International Hotel, Las Vegas, Nevada—Robert E. Reim, Hannon Engineering, Inc., Los Angeles, California

Providing Foldback with out-of-Phase Loudspeakers—Edward S. Jones, Brigham Young University, Provo, Utah

Multichannel Sound Systems for Multipurpose Halls—Lewis S. Goodfriend, Goodfriend-Ostergaard Associates, Subsidiary of Zurn Industries, Inc., Cedar Knolls, New Jersey

The Big Sound is on the Move with Disney on Parade—Albert A. Huff, Hannon Engineering, Inc., Los Angeles, California and William E. Blanton, Disney on Parade, Anaheim California

Sound Systems in Reverberant Rooms for Worship—David L. Klepper; Bolt, Beranek and Newman, Downer's Grove, Illinois

Thursday, May 7, 1970—7:30 P.M.
Los Angeles Room

AUDIO APPLIED TO EDUCATION, SCIENCE, AND INDUSTRY

Chairman: NORMAN L. CHALFIN
Jet Propulsion Laboratory, Pasadena, California

Acoustical Holography and its Potential as a Tool For Studying Sound Fields—Alexander F. Metherell, Douglas Advanced Research Laboratories, McDonnell Douglas Corp., Huntington Beach, California

Audio Communications for the Scientist—Claron L. Oakley, Audio-Digest Foundation, Glendale, California

Digital-Audio Industrial Control Devices—Clarence Hemphill, Aeronautical Dept., California Institute of Technology, Pasadena, California

Multimedia Audio Visual Techniques and Related Sound Signal Actuation Techniques—Martin R. Klitter, The Klitter Company, Inc., Pacific Palisades California

Transient Response of Earphones for Auditory Research—J. E. Jenkins-Lee, Department of Surgery, Stanford University School of Medicine, Stanford, California

Thursday, May 7, 1970—6:30 P.M.

Bus Pick-Up for AES Registrants with Reservations Only at 7th Street Entrance of Hotel on Lower Lobby

7:00 P.M.—Various Recording Studios in the Hollywood/Los Angeles Area

A RECORDING STUDIO WORKSHOP

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

Co-Chairman: ANDREW BERLINER
Crystal Industries, Inc., Hollywood, California

Co-Chairman: J. JERROLD FERREE
United Recording Corp., Hollywood, California

A Recording Studio Workshop, William L. Robinson, Andrew Berliner, J. Jerrold Ferree, and others to be announced

A recording studio workshop will be held in several recording studios in the Los Angeles/Hollywood area. Actual live sessions will be conducted, and microphone and multi-track recording techniques will be demonstrated and discussed. Dub-down, or reduction of the multi-track tape to finished two-track tape recording, will be demonstrated, using specialized and auxiliary equipment. Further, the use of artificial reverberation devices will be demonstrated and discussed.

NOTE: In order to attend the studio workshop, it is necessary to be registered for the convention and to return the reservation form no later than April 20, 1970 to:

Mr. Wm. L. Robinson, General Manager
Sunset Sound Recorders
6650 Sunset Boulevard
Hollywood, California 90028
(213) 469-1186

Primer on Methods and Scales of Noise Measurement, Part 4

WAYNE RUDMOSE

This is the concluding installment of this work. Earlier sections dealt with various fields of measurement, the relationship of media to sound propagation, and equipment for sound measurement and its use. The discussion on psychophysics now wraps it up.

PSYCHOPHYSICS

Turning now to a new but allied subject, the field of psychophysics relates the psychological response of humans to physical stimuli. The word "psycho-acoustics" is sometimes used to describe the field if the stimulus is sound. In general, man serves as part of the experimental apparatus and his response—oral, written, or some other—is based upon his psychological judgment of the attribute under study. Some measure of his psychological response must be developed in order to quantify the relations between the physical stimulus and the psychological response. For our purposes there are three such psychological terms which need definition: *loudness level*, *loudness*, and *noisiness*.

Wayne Rudmose is a group vice president of Tracor, Inc., located in Austin, Texas. The material of this series originally appeared in the American Speech and Hearing Association Reports 4 (February 1969).

Loudness level is a hybrid term and does not represent a true psychological unit of measure. It involves a psychological judgment but it is also characterized by a physical measurement of sound pressure level. By definition, *loudness level* is the sound pressure level of a 1000 Hz pure-tone stimulus that has been judged to be equally loud as the stimulus to be defined .Thus if the loudness level of a noise is to be measured directly, subjects must be asked to equate the loudness of a 1000 Hz stimulus with the noise. Once this equality has been established the sound pressure level of the 1000 Hz signal is measured and this level, expressed in phons (equal, for the 1000 Hz tone, to the sound pressure level in dB), becomes the loudness level of the noise.

Of more interest to psychophysicists is the matter of the true loudness of the noise. *Loudness* is a psychological measure, and it has been scaled by a number of experimenters. The experiments of S. S. Stevens and E. Zwicker dominate the literature dealing with this subject, and the methods developed by these experimenters are the only ones in general use by people in the field. The unit of loudness is the *sone*. The sone is arbitrarily tied to the physical scale by setting the value of the loudness of a 1000 Hz signal of 40 dB SPL (also 40 phons) as having a loudness of one sone. All other loudnesses are obtained by psychophysical judgments following the techniques of Stevens and Zwicker. Their techniques are similar in many respects but their methods of calculation differ principally in their interpretation of the effects of inhibition which takes place when humans listen to sound and judge its loudness. Each procedure has its camp of followers, and you will hear experimenters refer to "Stevens sones" or "Zwicker sones." For the purposes of this particular article it is my feeling that you can forget the subtle differences between these two procedures and accept the fact that from an engineering point of view we are in a position to calculate the loudness of a sound if we know its octave or one-third octave spectrum.

The advent of commerical jet airplanes spurred investigations to establish the relative acceptability of the noise of jet aircraft compared with the noise of propeller aircraft. Research in this field led to the development of the concept of ***noisiness***. The unit of noisiness is the ***noy***. The development of the noisiness scale paralleled the procedures used by Stevens in developing the loudness scale. The noisiness in noys, or the perceived noise level in PNdB, is principally associated with the name of K. Kryter. The perceived noise level, expressed in a decibel form as PNdB, represents the noisiness on a logarithmic scale rather than on the linear scale of noys. The details of how one calculates the PNdB value of a noise are not important at this point; but it is important to understand that if the octave or one-third octave spectrum of a noise is known, one can calculate, by similar procedures, the loudness in sones, the loudness level in phons, the noisiness in noys, or the perceived noise level in the PNdB. These values may differ somewhat for the same noise spectrum; but as long as one restricts noisiness to airplane-type noise, Kryter feels there is a significant difference between noisiness and loudness and that these two concepts are not the same. I think it only fair to point out that other people in the field share a feeling that the concept of noisiness may not differ significantly from loudness over a wide range of noise characteristics. Research in this area has not been extensive enough to decide the matter one way or another. At present it would seem best, in my opinion, to limit the application of the noisiness concept to jet aircraft and propeller aircraft noises.

MISCELLANEOUS TERMINOLOGY

Several other terms will probably appear during the course of investigations and should be defined. One of these is *speech interference level* (SIL). Many problems of noise control are related to establishing an environment conducive to carrying on conversation by individuals separated by conventional distances and speaking in normal levels of speech. It has been shown that the masking noise which lies within the frequency range of approximately 500 to 5000 Hz is important in determining the masking effect of the ambient noise in terms of speech. The original psychophysical work related masking to the average sound pressure levels measured in the three octave bands: 600-1200, 1200-2400, 2400-4800 Hz. The numerical average of the dB values per octave in these three octave bands is defined as the speech interference level. In terms of the new octave designation by center frequencies, the corresponding SIL value is the average sound pressure level measured in the 1000, 2000, and 4000 Hz bands. The term SIL can also mean the average SPL in the 500, 1000, and 2000 Hz bands.

Criteria for various types of environments such as offices, libraries, and theaters have been developed in several countries. The result is a series of sound spectra contours extending over a wide range of sound pressure level values. These curves have been defined as Noise Contours (NC) and are extremely useful in many aspects of noise-control work. An acceptable NC contour for a given space will permit a spectrum which decreases in sound pressure level as the frequency increases; that is, more low frequencies are permitted than high frequencies. Spectral ranges have been established for a wide variety of everyday activities, and these criteria are published in the standard textbooks and in the scientific literature.

A few closing comments must be made concerning the terminology which you will hear used in connection with various criteria where the stimulus, as related to the noise criteria, will inevitably be given in terms of ***noise exposure***. It must be recognized that noise by itself is not the total stimulus which is used to establish criteria, but it is the combination of noise and exposure time that is important. The manner in which noise levels and time are combined results in a noise exposure variable. The methods of combination differ depending upon the subject.

In much the same way, the measurement of the human response depends upon the subject. If the problem is damage to hearing. The measurement of hearing is, at present, performed by pure-tone audiometry. The pure-tone audiometer determines the sound pressure level at various frequencies at which the subject just hears the pure tone. The difference between the threshold level for the individual being tested and the accepted normal threshold level for young individuals is the measure of the hearing threshold shift at that particular frequency. If this shift is a temporary one, the difference is referred to as temporary threshold shift (TTS); however, if this threshold shift is stable with time, it is referred to as a permanent threshold shift (PTS). For airport noise problems, the measure is the "annoyance" or the "complaint" level of the community. In other cases speech interference level is the measure—and so it goes.

It is inevitable that within the small space of this presentation certain terminology which should have been defined and discussed will have been omitted. An attempt has been made to cover the major topics in terms of their terminology, their concepts, and their definitions. ■

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Classified

(Letters—from page 2)

With regard to the use of the Ortofon 2-15k transformer; a performance comparison of the low impedance system versus the transformer and a standard preamp system will readily demonstrate the superiority of the low impedance system in signal-to-noise and distortion improvement.

My statement regarding the wide bandwidth of the preamplifier was intended to indicate the performance potential of the preamplifier. Certainly, one cannot hear 300 KHz. but the transient performance of the preamplifier offered by wide bandwidth is indisputable. There may not be program material down at 10 Hz. but the subharmonic *feel* of low bass passages is often desirable for optimum sound reproduction.

The RIAA equalization curve does recommend rolloff below 30 Hertz primarily to cover up rumble and noise performance in both turntables and preamplifiers. When a top quality turntable with low rumble performance is used with the preamplifier .noise and rumble below 30 Hertz are not evident in listening tests on the best wide-range loudspeaker systems available.

The measurement of signal-to-noise performance of a preamplifier with equalization is not an easy one and most standards are inaccurate in this regard. Since the equalization network provides low-frequency boost and high-frequency attenuation, low-frequency noise components are exaggerated beyond their true value. The proper way to measure the preamplifier signal-to-noise performance of this system is to construct a preamplifier with the opposite equalization curve of the phono preamplifier and insert it and a band-pass filter which is ±3 dB down at 20 Hertz and 20k Hertz between the phono preamplifier and a vtvm. Playing a test record recorded at a velocity of 5 cm./sec. through the system will produce a reference output level which will be +5 dBm. at the output of the phono preamplifier. Then, remove the pickup arm from the record and measure the residual noise with the cartridge connected. This measurement will provide a test that is typical of actual performance in a reproduction system.

I would be most interested to evaluate your company's new preamplifier. How will it compare under the measurement test I have described above?

Allan P. Smith

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People, Places, Happenings

Allen

●The explosive growth of the tape duplication industry has prompted an announcement of the major reorganization of **American Tape Duplicators.** ADT president **Richard Allen** said that the company is forming a management committee to help guide the future growth. The committee is formed of Mr. Allen, **Warren Gray,** executive v.p. and secretary, **Jay Leanse,** v.p. and director of sales, **Donald Anderson,** v.p. and director of operations, and **Stanley Moss,** treasurer and administration. Mr. Allen and Mr. Gray have long experience in the duplication field. Both go back to the mid fifties when they worked for **Bel Canto,** a division of **TRW.** In 1962 they formed the present company with an investment of $1000. The remaining men on the committee have joined the company more recently.

Plunkett

●Brad Plunkett has been appointed director of engineering at **United Recording Electronic Industries.** of North Hollywood, California. For the past three years he has been an engineering consultant, specializing in the fields of professional audio and electronic music. Prior to joining UREI he was with the **Thomas Organ Company** where he was senior engineer. He holds eight U.S. patents and a number of foreign ones in the music and audio fields, with six others pending.

●**Switchcraft** has announced the sale of its audio-visual division according to **Wilfred L. Larson,** president of the company. **Robert D. Hall,** former sales manager of the division will head a new company. **Avedex, Inc.** to continue the line and handle its distribution. According to Mr. Larson, the sale was motivated by a need to place greater concentration on other Switchcraft products such as phone jacks and plugs, connectors, switches, and molded cable assemblies.

●Flanked by **Representative Charlotte T. Reid** and **Senator Charles H. Percy, Jack Stone** (on the left center) president of **DuKane Corporation** accepts the President's E Award for outstanding contribution to the nation's export expansion program. Making the award is **Rocco C. Siciliano,** Under Secretary of Commerce (with glasses). Export sales of electronic sound communications equipment now account for more than 10 per cent of the St. Charles, Illinois firm.

●A new company has been formed to provide consultant services and to represent the **Westrex Division** of **Litton Industries** in the Northeast portion of the U. S. The company is **Alpha Sound Equipment Corp.** and is based in the Bronx, New York. **William J. LaIliff** is the company president.

●At this moment, a group of adventurous audio men are somewhere in the Soviet Union visiting various audio installations. **John Woram** helped organize the trip, officially sponsored by the **Citizen Exchange Corp.,** and he and **Christina Woram** are there. Included in the group are **Alfred Bruck** and **Jack Clink,** respectively president and director of sound devices at **Capital Film Labs** of Washington D. C. and **Randall Kling** a recording engineer based in Chicago with the **RCA** record division. When the trip returns we expect interesting editorial material from John on Soviet, Czech, and East and West Berlin audio.

●**Craig H. Stevenson** is now manager of **James B. Lansing Sound, Inc.'s** new professional applications division. The division is a result of a merger of the professional products and OEM branches of the JBL corporate structure. In the announcement by **Barney Rigney,** director of marketing for JBL it was noted that Mr. Stevenson has an extensive background in commercial sound and professional audio marketing. Prior to coming to JBL, he was sales manager for the Procast division of **Harman-Kardon, Inc.** and earlier was consumer products manager for **Jerrold Electronics Corporation.** For the past six months, he has been manager of the Procast division at JBL.

In another announcement from JBL it was noted that **Bart N. Locanthi** has been promoted from v.-p. engineering at JBL to v.-p research for all Jervis divisions (JBL is a Jervis Corporation company)

●The **Rupert Neve Company** continues to install consoles all over the world. Recent installations have been made at the **Arne Bendilsen** in Oslo, Norway (this was an 8-track console) and smaller desks have gone into Yugoslavian and Swiss studios. Several desks are headed for Spain and one will soon be installed in **Whitney Recording** on the West Coast.

Hofman

●A news item from **Sennheiser** states that **Cornelis Hofman** has been appointed v.-p. marketing and sales for the New York based company. Mr. Hofman comes from Amsterdam, Holland where he can look back on more than twenty years of experience with Sennheiser and other products with the **N. V. Kinotechniek.**

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