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THE SOUND ENGINEERING MAGAZINE

June 1970 75c

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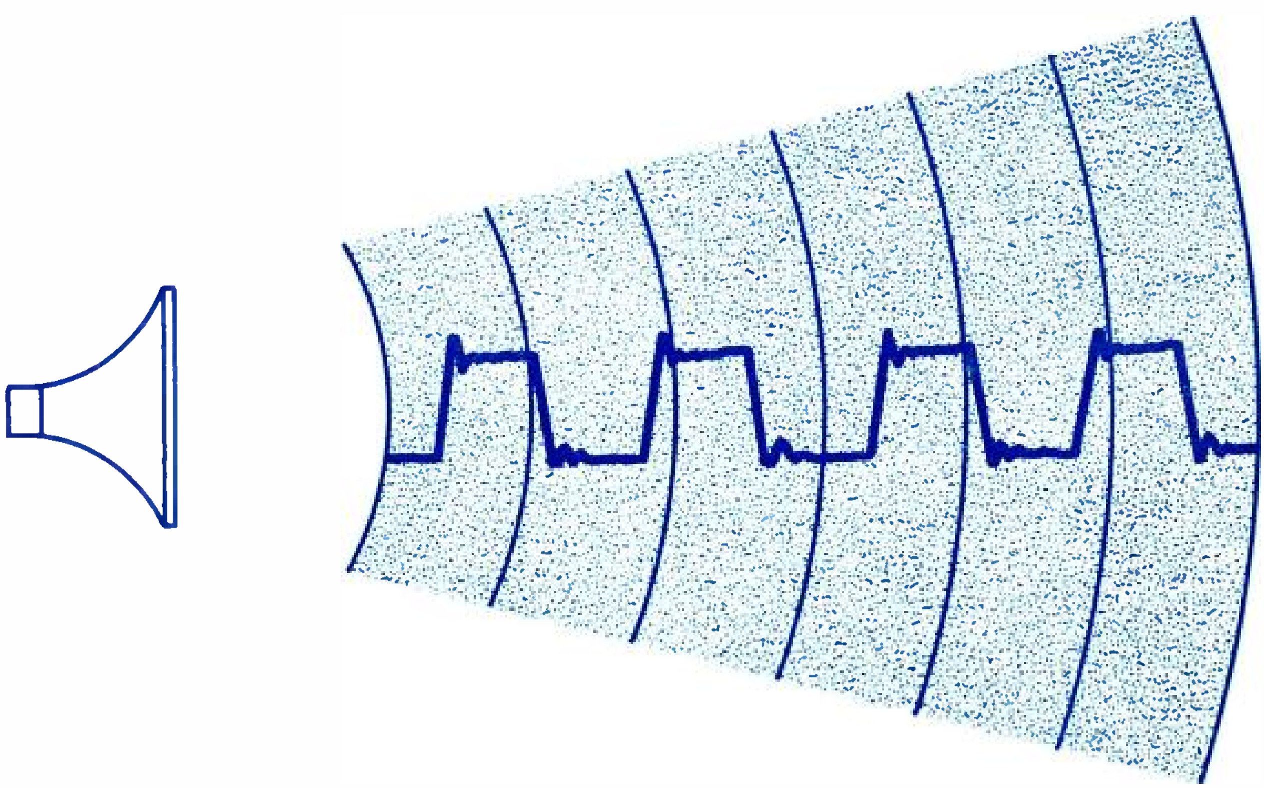
Powering Condenser Microphones

Hearing Loss and the Audio Engineer



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Coming Next Month

● The world of motion picture sound is explored in two articles.

Sidney Silver is represented with SYNCHRONOUS RECORDING TECHNIQUES in which he discusses the various methods of synchronizing sound to film. You will learn how to achieve the best results from current equipment.

OPTICAL SOUND TRACK PROCESSING by J. W. Dorner is based on a paper originally presented to the SMPTE. It looks to the preservation of sound quality on the sound track as the film passes from generation to generation.

ELECTRONIC VIDEO RECORDING — IN COLOR is an updating of E. T. Canby's original **db** article on the CBS EVR system. With both hardware and software promised for this year, EVR is sure to be a major audio-visual medium.

A picture gallery taken at the AES Convention just concluded in Hollywood will show latest equipment and scenes snapped at the show.

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

● The NAB Convention, represented this month with a picture gallery, captured the imagination of our technical artist Irwin Handel. He has represented radio propagation, phono pickup and audio in his blue-on-white.

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THE SOUND ENGINEERING MAGAZINE

JUNE 1970 • Volume 4, Number 6

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Letters

The Editor:

As a manufacturer of professional audio equipment who has attended many AES conventions I have been most impressed with the size and attendance of the most recent two held in New York and Los Angeles. However, the costs involved in preparing and exhibiting at two shows a year is becoming quite considerable as hotel and other costs go up.

The thought has occurred to me that the entire industry could be better served if the show was held one year on the west coast and the following year on the east coast. By alternating sites the convention could then attract more manufacturers who can ill afford the time and expenses involved in two very similar shows in one year.

It would be interesting to hear from others who feel that this is a real problem.

NAME WITHHELD

The Editor:

Being constant readers of your magazine we were pleased to note that in your January issue, the article Decca's Vienna Venue by John Borwick appeared.

However, we did wish to make one amendment. While all the details and subcontractors such as the echo plates by EMT, the monitor speakers by Tannoy, the power amplifiers by Quad, the microphones by Neumann and AKG, and the tape machines by Studer, were mentioned, no mention was made that the control system and the control desk, respectively (which are the heart of the system), were designed, built, and installed by our firm. This is the more regrettable to us since it was our firm, in close liason work with the representatives of Decca, in particular Mr. Parry and Mr. Brown, established this system. At that time the name of our firm was Wiener Schwachstromwerke GMBH (WSW). Effective October 1st, 1969 the name has been changed to Nachrichtentechnische Werke AG (now contracted to NTW).

Nachrichtentechnische Werke AG
Apostelgasse 12
A-1031 Vienna, Austria

Our faces are red. In the Letters column of the April issue, Roger K. Odom wrote in response to an earlier article by Allen P. Smith. Mr. Odom's employment was incorrectly attributed by us. He is in fact the engineering manager of Sparta Electronic Corporation of Sacramento, California.

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

● This time I shall make good on my promise to discuss active filters as applied to modern solid-state circuits.

Classical circuits and their derivation can and should be viewed as a science all by themselves. If you wish to study filter design consult books by authors such as Bessel, Butterworth or Chebychev. My goal is to familiarize you with ready-designed circuits that are easy to apply to almost any existing system. You will notice that circuits mentioned will not use any inductors; this is the major expense when constructing filters or equalizers. The circuits to be described will be of mixed variety representing active circuits with bandstop or bandpass properties.

The first circuit is designed around an operational amplifier or any other amplifier approaching the ideal amplifier characteristics. This means that the amplifier should have high input impedance, low output impedance, flat frequency response from d.c. to r.f., minimal phase shift, high open loop gain, and very low noise.

The circuit in the FIGURE 1 is the low-pass circuit, while reciprocal function exists in the circuit shown in FIGURE 2. In both cases the same amplifier is used. In order to figure the values of the components for FIGURE 1, the following formula should be used.

$$C_1 = \frac{2}{R_x \Omega_c} \quad C_2 = \frac{0.5}{R_x \Omega_c} \quad \text{where } \Omega_c = 2\pi f$$

f being cutoff frequency
R can be selected

For Desired Input Impedance

The circuit in Figure 2 is a high-pass filter and its values can be determined using following formulae:

$$R_1 = \frac{1}{2 \times C \times \Omega_c}\ ;\ R_2 = \frac{1}{0.5 \times C \times \Omega_c}$$

where $\Omega_c = 2\pi f$
and f is cutoff frequency

Both circuits can be used one after another and depending on the cutoff

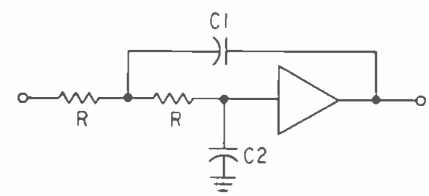


Figure 1. A low-pass filter circuit.

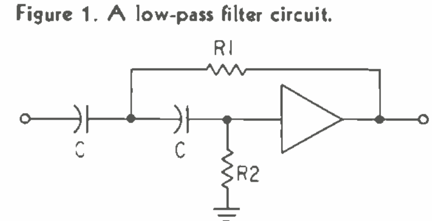


Figure 2. The reciprocal function of the circuit shown in Figure 1.

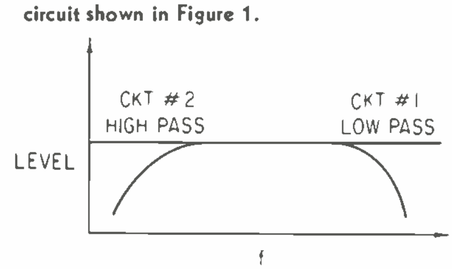


Figure 3. The circuit action of Figure 1.

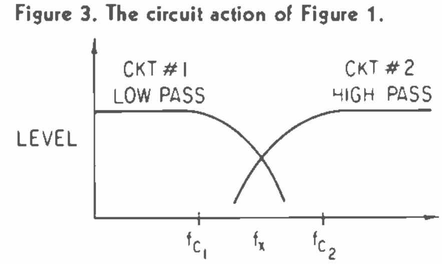


Figure 4. The circuit action of Figure 2.

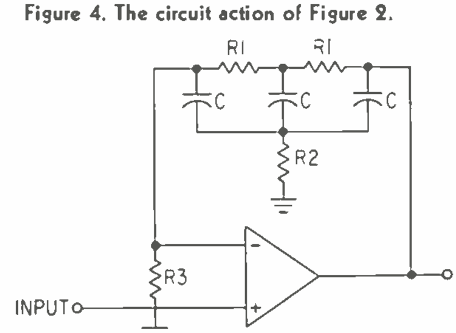
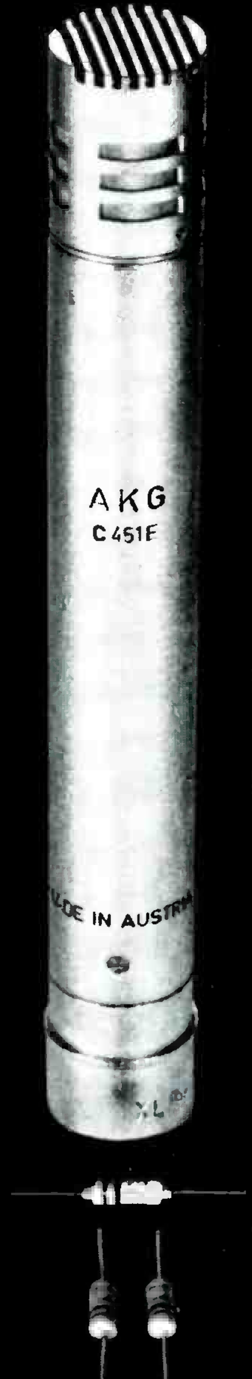


Figure 5. A notch-filter circuit using an operational amplifier.

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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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.

Last, but not least, a practical and economic modular design permits you to choose from a variety of interchangeable capsules with different response characteristics and polar pattern (cardioid, omni-directional, figure-eight and shot gun tube).

AKG OFFERS A CHOICE— NOT A STANDARD

Write for complete technical data.

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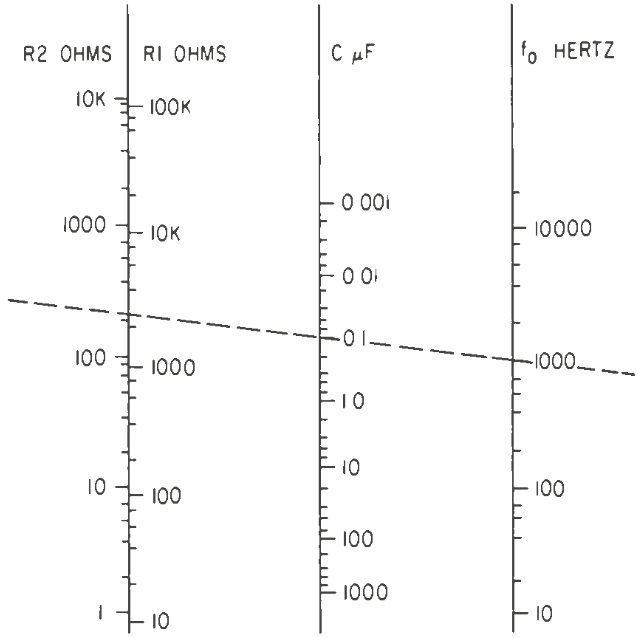


Figure 6. A nomograph for the selection of component values for frequency using the circuit of Figure 5.

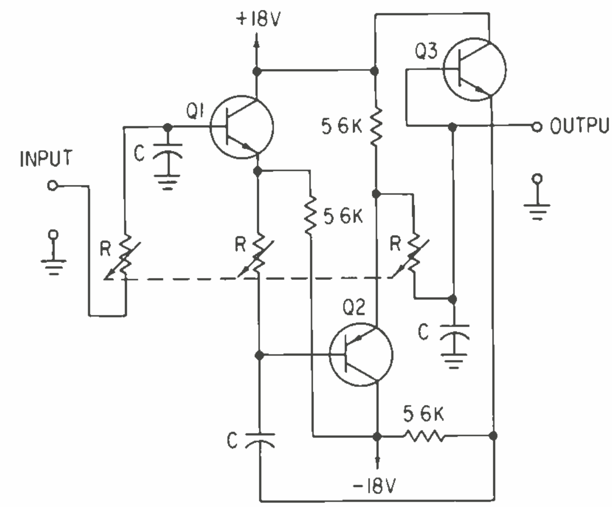


Figure 7. This circuit will produce fairly sharp low-frequency cutoff points.

frequencies can be used in either narrowing the frequency band being active on both sides of the spectrum or as a band-stop filter having overlaping functions.

By changing the cutoff frequency points of either of the filter circuits in FIGURE 3 or 4, level at f_x can be varied. It can be seen that the circuit in FIGURE 1 lends itself to adjustment more readily than the circuit in FIGURE 2. Using a ganged rheostat or potentiometer permits both resistors to be varied, changing the cutoff frequency.

There is one more interesting and fairly simple notch filter circuit (see FIGURE 5) using an operational amplifier. It can be very useful in designing of environmental equalizers, taking out sharp response peaks. Selection of the component values for the particular frequency is accomplished by using the precalculated nomograph of FIGURE 6. R_3 resistor is selected to produce zero offset voltage at the input of the amplifier. R_2 is selected to produce zero phase shift at the center frequency and is equal to 1/12th of R_1. The center frequency is $f = \sqrt{3}/(2 . R_1C)$

An example on the nomograph shows the resonant or center frequency of 1000 Hz. From the graph, R_2 has to be 220 ohms, R_1 should be 2400 ohms and the capacitors should be 0.1μF each. The advantage of this circuit is that it has very high Q, especially at the low frequencies where it is desirable to eliminate large and costly inductors.

One more circuit producing fairly sharp low-frequency cutoff using three transistors is useful in systems where low-frequency rumble noise or resonances would produce unwanted coloration of the signal. This circuit can be used in sound reinforcement amplifiers, turntable preamps, and other communication circuits. The circuit of FIGURE 7 uses a three gang variable potentiometer or rheostat. This enables shifting of the cutoff point over the considerable frequency range. If this is not desired, then fixed resistors can be substituted. This circuit rolls off the response at approximately 16-18dB per octave. Frequency of the cutoff point is found from the equation; $f = \frac{1}{2} . RC$

The only disadvantage of this circuit is that it requires a split power supply. However, if one considers the advantage of not needing all these inductors and the ability to choose the rolloff frequency, advantages outweigh the inconvenience of a split power source.

The above mentioned circuits are only a sample of what can be done using active filters. Fast development of microcircuit technology will enable us to pretty soon start using some of the packaging techniques used in computers. Serious efforts to develop circuits packageable onto a single chip are beginning to bear fruits. Although not all electronic elements can be achieved in monolithic circuits, but almost all electronic functions can be reproduced. With a minimal number of external components LSI (large-scale integration) devices can work as complete systems. You can have radio circuits, computers, amplifiers, clocks, calculators and thousands of other circuits packaged on a single chip. The day will come when one can purchase the complete environmental equalizer, perhaps with self-seeking peak compensator circuits and automatic gain controls to take advantage of every bit of available dynamic range and adjust the reproduce levels for optimum performance.

I can't resist the temptation to mention one more simple tunable twin

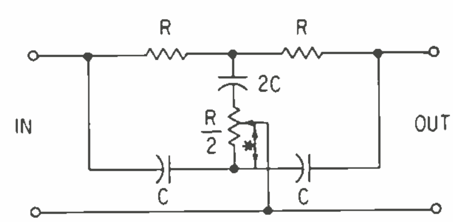
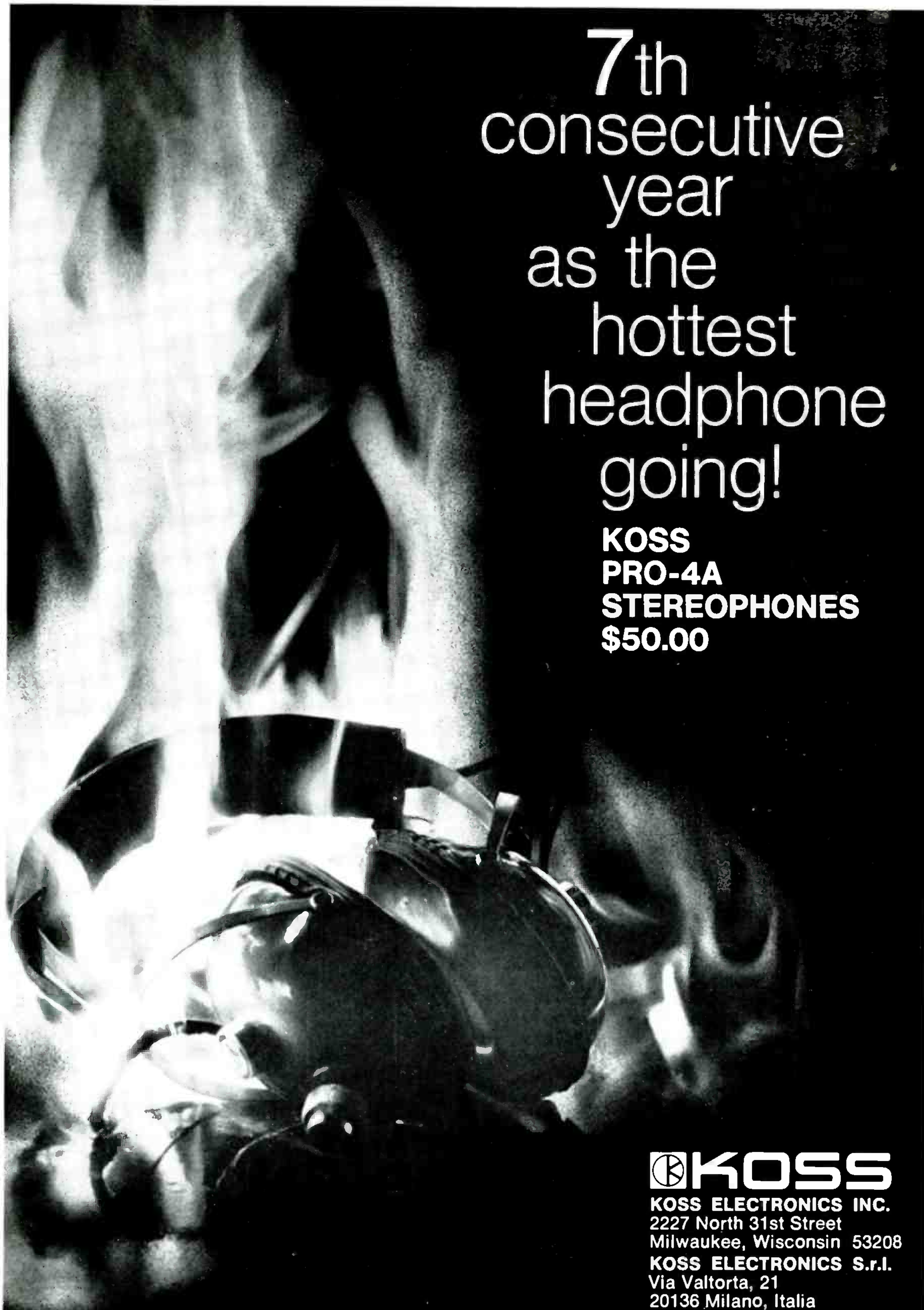


Figure 8. A twin T filter configuration. $* = \frac{AR}{2}$ **where**

$$A = \text{factor and } \Omega_0 = \frac{1}{RC\sqrt{1-A^2}}$$

T filter, I have recently come across. It can be seen in FIGURE 8. If the potentiometer is set so that the slider is down A becomes 0 in the formula of the figure. As a consequence, $\Omega_0 = \frac{1}{RC}$ where R is in ohms and C is in farads. ■



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The Feedback Loop

ARNOLD SCHWARTZ

●What is the upper frequency limit of the disc recording-playback system? Can the bandwidth be extended beyond present limitations? These questions may have added significance if four-channel stereo graduates from the experimental state. In that event, disc recording will have to be developed to accommodate four channels instead of the present two. One possible way of adding the two additional channels involves increasing the record/playback system bandwidth. I would like to discuss the basic principles underlying the present bandwidth limitations, and try to predict how it may be extended.

In any discussion of bandwidth we can say that virtually all purely electronic devices either have been, or can be designed with a higher upper frequency limit than today's nominal 20 kHz. Disc-recording bandwidth is limited, however, by the electro-mechanical and mechanical devices found in the record/playback chain. Broadly speaking, there are two types of bandwidth limitations. One category, to be discussed this month, relates to the frequency response of the individual devices. A second category, to be discussed next month, relates to basic geometry and wavelength considerations. Bandwidth limitations due to the limitations of frequency response are more familiar than those due to system geometry. Very often both types of limitations are confused and talked about interchangeably when, in fact, important distinctions must be kept in mind.

Let us take a look at the electro-mechanical and mechanical devices used in the recording and playback of a disc, and see what the frequency response of each one is, and if this response can be improved.

RECORDING HEAD

The best present-day stereophonic recording heads can respond to about 20 kHz. This is adequate for present-day records. An immediate alternative, although with some drawbacks, for extending the frequency response would be to record at half speed (or two-thirds speed depending upon the bandwidth extension). All current electronic equipment could be used but some modifications would have to be made both to the tape recorder playback equalization, and to the RIAA recording characteristic. In the long run it may be possible to design recording heads with increased bandwidth, designed to operate at normal speed.

RECORDING STYLUS

At my company (Micro-Point) we have designed our recording stylus so that there is no restriction of the cutter-head frequency response. We have also designed, and, if the need arises, will be able to manufacture recording styli capable of cutting records with at least twice the present bandwidth.

PHONOGRAPH CARTRIDGE

The typical high-quality cartridge has an upper frequency limit of about 15 to 20 kHz. In the March FEEDBACK LOOP there was a detailed discussion of the factors that affect the upper frequency limit. Regrettably, Miller's formula for the frequency response (H) of the cartridge, essential to that discussion, was omitted because of a typographical error. Since the subject comes up again, I'll take the opportunity of putting down the formula again.

$$H = \left[\frac{1}{\left|1 - \left(\frac{f}{f_o}\right)^2\right|^2 + \epsilon^2\left(\frac{f}{f_o}\right)^2} \right]^{\frac{1}{2}}$$

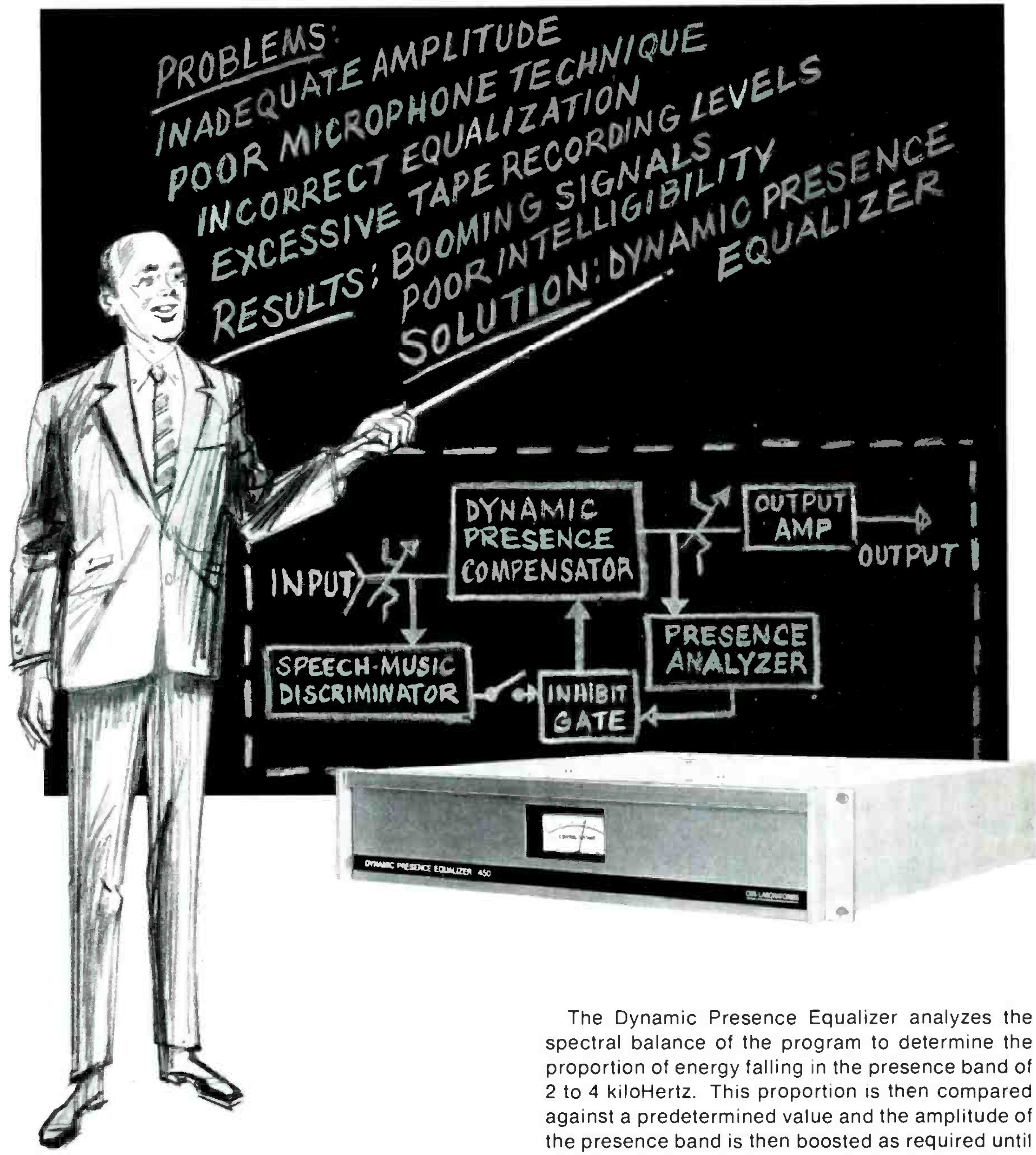
$$\text{where } f_o = \frac{k}{\sqrt{m}} \left[R_1 F_s E^2 \right]^{\frac{1}{6}}$$

$$\epsilon = \frac{r}{m(2\pi f_o)}$$

f_o = stylus-groove resonance frequency
m = mass at stylus tip
r = mechanical resistance at stylus tip
ϵ = damping factor
R_1 = Tip radius
E = modulus of elasticity of record material
F_s = tracking force
K = constant

By analyzing the equation and plotting responses of typical cartridges, we do know that the upper frequency limit is about one-half octave above the stylus-groove resonance, f_o. How can we increase the cartridge upper frequency limit? As we stated in March, increasing the tip radius or the tracking force to increase f_o would be counterproductive to other performance characteristics. This leaves us with the modulus of elasticity (that is stiffness) of the record material, and the mass at the tip. Increasing the stiffness of the record material to raise the stylus-groove resonance frequency means finding a material that has all the desirable characteristics of vinyl but with a significantly higher modulus of elasticity. This is a tall order when we consider such things as low noise, ease in molding, durability, stability—all at a price competitive with vinyl. If such a material were introduced, all cartridges would experience a proportional increase in frequency response. Thus, changing the record material is an attractive, although very difficult, solution if it were necessary to increase the bandwidth.

Decreasing stylus mass at the tip is the other alternative to increasing f_o. At least one current cartridge has a response to beyond 30 kHz, which means that the effective mass at the tip for this particular cartridge is about one-fourth of that found in many current high quality cartridges. If there was a need for extending the frequency response, other cartridge manufacturers would probably follow suit. ■



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Theory and Practice

NORMAN H. CROWHURST

● The postal strike and April 15 both came and went before you learned, in the delayed March issue, about my problems with the IRS. Reading my Declaration evoked quite varied responses, reflecting even further on theory and practice in America today.

One man told me that when the imigration authorities fingerprinted us, that identified us as criminals, a fact we should have accepted when I was denied security clearance. Now, he said, if I would just quit my dictatorial ideas and learn about democracy, everything would be all right. A tongue-in-cheek letter, maybe? No, he also "informed" the County DA, urging him to take action against us!

However, the majority of responses indicated a feeling of shame, that our Country's agents should so far forget its Constitution as to treat us this way, and admired us for our courage in taking such a stand. A few spoke of the impossibility of fighting the system, suggested we should give up the struggle, go on welfare and live it up. Ever tried to live it up on welfare checks?

With the exception of the man who thinks only criminals are fingerprinted, and thus all foreigners are criminals and/or dictators, everyone who recommended compliance also rejected completely the practicality of principle. What happened to the Declaration of Independence—A document of principle, if ever there was!

Whenever someone says we have courage, my retrospective view tells me we really had no choice: we were forced into it. But were we? We did have an alternative: to do something or to do nothing.

As we could not pay, and we refused to take up crime, we could have done nothing but wait for them to "come and get us." What defense would that be? We must do something, in self-defense, to prove we are neither delinquent nor evasive. It was not easy. That document was compiled, revised, let sit, revised again, let sit again, before we finally filed it on February 18.

Many ask if I recommend everyone to refuse paying taxes. My answer is the same as that I give to those who ask if I think everyone should take the same religious position I do: this position is what my conscience, in my circumstances, demands. My conscience does not demand that I refuse to pay taxes because of the Country's involvement in Vietnam, although I respect those whose conscience does thus object. However my conscience does not allow me to participate willingly in fraud or any other criminal activity. When it was made impossible for me, due to circumstances peculiar to a situation the government has imposed to me, to pay honestly, I took a long hard look at what I had gotten into.

"But," my pragmatist friends will say—and this bothered me, too—"What would happen if nobody paid taxes?" Does asking that question justify IRS pressuring me (or anyone else) into some form of criminal activity? The Constitution provides adequate means for maintaining a just government. Did you imagine that Congress and government employees worked for nothing before 1913?

But back to technical matters. The circuit shown last month, to provide protection against over-dissipation, as well as over-current (repeated here at Figure 1) is not quite ideal. Q8 either conducts, charging capacitor C to shut off the audio, or it does not, according to the zener voltage of the diode. The over-heating caused by dissipation, results from voltage, current, and time.

A momentary excess of dissipation—say for a millisecond—will not damage the transistor, because heat does not have time to raise temperature to the danger point. On the other hand, the same dissipation continued for an appreciable fraction of a second could destroy the transistor.

This means that finding the right zener voltage for the diode poses a problem. If it is low enough to protect against the continued dissipation, it will trip the circuit out of action on momentary peaks that really pose no danger. On the other hand, if it is high enough to allow safe momentary peaks, one of longer duration could blow a transistor.

This leads to the more satisfactory circuit of Figure 2. Here R1 and C1,

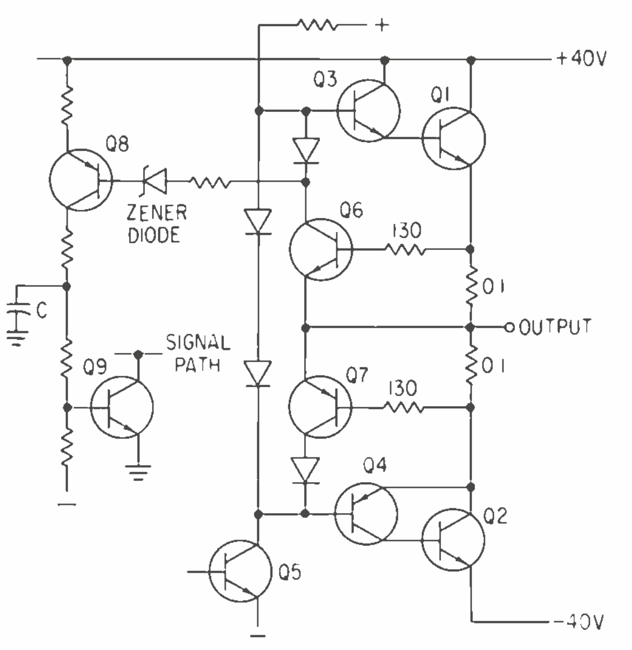

Figure 1. The circuit developed in the last discussion, reproduced as a beginning point for this one.

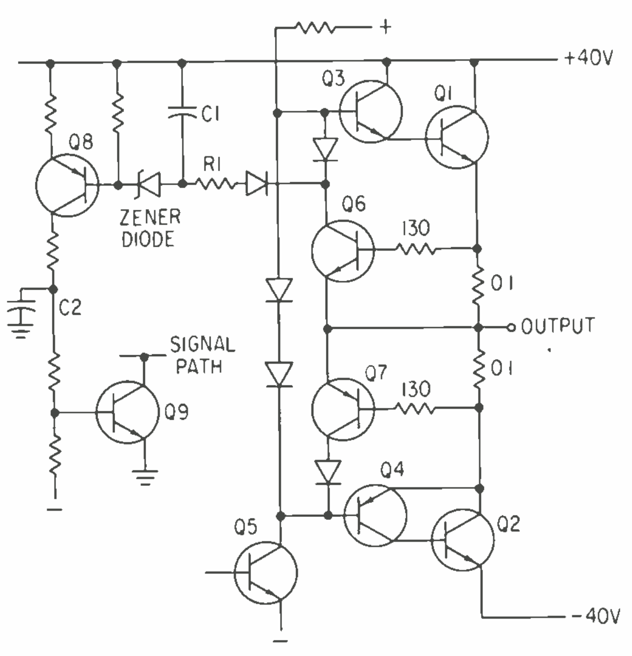

Figure 2. The changes shown here (principally the addition of C1) enable the circuit to be flexible in its tripping point, so it can accommodate short or long over-dissipation threats equally well.

STEREO REVIEW, JAN. 1969

EQUIPMENT TEST REPORTS

By Hirsch-Houck Laboratories

REVOX A77 TAPE RECORDER

● It is a pleasure to report that the widely acclaimed, but no longer available, Revox G-36 Mk III tape recorder has actually been surpassed in performance by Revox's new Model A77. The A77 has fully solid-state electronics, a bias-oscillator frequency of 120 kHz (as opposed to 70 kHz for the G-36), and a new electronic motor-speed control. The A77 model we tested is a three-motor, four-track, two-speed recorder; however, it is substantially lighter and smaller than its predecessor.

The Revox A77 has its operating controls grouped into separate recording and playback areas. On the playback side are two rotary switches with concentric knobs. One switch establishes the playback mode—stereo, either channel through both outputs, or both channels combined for mono. Playback level is controlled by the concentric knob. The other switch connects either the signal input or the output of the playback amplifiers to the output jacks in the rear. Two playback-equalization characteristics are provided; NAB or IEC (for European tape recordings). The recording equalization is to the NAB standards. The knob concentric with this switch is a playback channel-balance control.

On the right side of the recorder panel are two VU meters with real VU-meter characteristics. Adjacent to each is a red button of the push-on, push-off type. Depressing either channel's button alone records both inputs on that channel. If both buttons are depressed, a stereo recording is made. These supplement a record-interlock button, providing a double safety against accidental tape erasure. Recording levels may be set up before the tape is put into motion. When the recorder is in operation in the recording mode, the selected channel's VU meter (or meters) is illuminated.

Under each meter is a recording input-selector switch, with a concentric recording-level control. There are inputs for high- and low-impedance microphones (with front-panel jacks in parallel with rear phono connectors), radio (via a rear DIN connector), and auxiliary inputs with connectors in the rear. In addition, each switch has a position for recording the output of that channel combined with any additional source onto the other channel.

The transport mechanism is operated by a row of five pushbuttons, activating solenoids to control fast speeds, stop, play, and recording. A connector in the rear permits the use of an accessory remote-control unit for these functions. The tape speeds (7½ and 3¾ ips) are selected by a switch that also controls a.c. power to the recorder. Each speed setting has two switch positions that set the tape tension to optimum values for 10½-inch or smaller reels.

The servo-controlled drive system of the Revox A77 is unique and effective. The tape-drive capstan is powered by an eddy-current motor that delivers a high torque, free of the pulsations that are inevitable with any motor having a pole structure. The speed of this motor can be adjusted by varying a d.c. control voltage, with relatively little torque variation. The motor has a built-in tone generator that produces an a.c. signal whose frequency is proportional to motor speed. This signal is amplified, limited, and applied to a discriminator, whose d.c. output is proportional to speed. This is further amplified and used to correct the motor speed. The change between 7½ and 3¾ ips is accomplished electronically by shifting the resonant frequency of the discriminator circuit. The chief advantages of this technique are independence from power-line voltage and frequency variations, as well as reduced flutter. Flutter of the A77 motor is inherently so low that the capstan can be driven directly from the motor shaft instead of through a separate belt-driven flywheel. According to the manufacturer, line voltage fluctuations of ±20 per cent cause a speed change of only ±0.05 per cent, and a change in the a.c.-line frequency of 50 to 60 Hz causes a speed change of less than 0.05 per cent. Thus, the Revox A77 is a truly universal machine, capable of operating from 110 volts to 240 volts, 50 to 60 Hz, by adjustment of a switch in the rear of the recorder.

When the full-width head cover is swung down, two more pushbuttons are revealed. One cuts off the signal to external speakers, and the other switches off the power to the reel motors. This is for convenience in editing. When the reel motors are turned off, and the recorder placed in a fast-speed mode, the reels may be turned by hand with the tape in contact with the playback head. At the desired point, the tape may be lifted from the heads and placed in the tape splicing guide which is molded into the fixed portion of the head cover. The only problem with this arrangement is the possibility that one may spill tape by forgetting to turn on the reel motors before placing the machine back into normal operation.

We stated that the A77 surpassed the older G-36 in performance. This is best illustrated by its phenomenally flat record/playback frequency response, measured with Scotch 203 tape, for which the machine's bias was adjusted. At 7½ ips, the response was within +0.5, −2.0 dB from 20 to 20,000 Hz. This has never been equalled by any other recorder we have tested. Perhaps even more impressive is the response at 3¾ ips, which was +2.5, −5.5 dB from 20 to 20,000 Hz. The high end falls off smoothly and is perfectly usable all the way to 20,000 Hz. The NAB playback response, with the Ampex 31321-04 test tape, was +1.5, −0.5 dB from 50 to 15,000 Hz.

The signal-to-noise ratio was very good, 51 dB at 7½ ips and 48.5 dB at 3¾ ips, referred to a 0-VU recording level. Noting that the distortion at 0 VU was a mere 0.65 per cent, we increased the recording level until the distortion reached approximately 3 per cent, which occurred at +10 VU for the higher tape speed and +9 VU for the lower speed. At these levels, the signal-to-noise ratio was 59 dB at 7½ ips and 54.5 dB at 3¾ ips, figures that closely approach true professional performance.

The transport worked smoothly and with complete silence. Except for the turning of the reels, one could not tell the machine was operating from a distance greater than about 12 inches. Wow was 0.01 per cent (actually the residual inherent in our instruments) and flutter was 0.09 per cent at 3¾ ips and 0.07 per cent at 7½ ips. In fast speeds, 1,800 feet of tape was handled in about 90 seconds, and the machine could be brought to a stop in about 2 seconds.

The Revox A77 is housed in a teak cabinet with a fold-away carrying handle. It is one of the handsomest, as well as best-performing, tape recorders we have seen. We have never seen a recorder that could match the performance of the Revox A77 in all respects, and very few that even come close. It sounds as good as it tests, which speaks for itself.

The Revox A77 is offered in a variety of configurations. It is available with either half- or quarter-track heads, in either the teak cabinet or a portable carrying case. The price of the deck in a wood base is $569; the deck with built-in power amplifiers is $599.

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feeding the zener diode, duplicate the heat-up time constant of the transistor, in the dissipation range involved. The zener voltage allows the circuit to trip when the dissipation, sustained for more than a moment, would be dangerous. But the same dissipation for a very short burst will not charge C1 to the zener voltage. However, if the shorter burst carries enough dissipation to be dangerous, it will.

C2, with its associated resistors, produces a time constant designed to hold the audio off for long enough to allow the already heated transistors to cool to the point where they will again be safe to risk applying the over-dissipation that caused tripping. If the danger is still there, the circuit will trip out again.

One weakness with this circuit is that it will only trip if the voltage/current combination on Q1, Q3 represents dangerously high dissipation. Q2, Q4 are protected against over-current, but not against associated over-voltage. FIGURE 3 shows a circuit that provides an additional sensing arrangement so both are equally protected.

Transistors Q1 through Q8 serve the same purposes designated earlier. Q10 and Q11 combine input with feedback, to drive Q5. Q12 duplicates the function of Q8, for the other half of the output, also triggering the voltage on C2. Diodes D6 and D7 prevent C1 and C3 (which duplicates C1 function for the other half) from being discharged unnecessarily on repetitive peaks that could prove dangerous. D8 and D9 are the zeners.

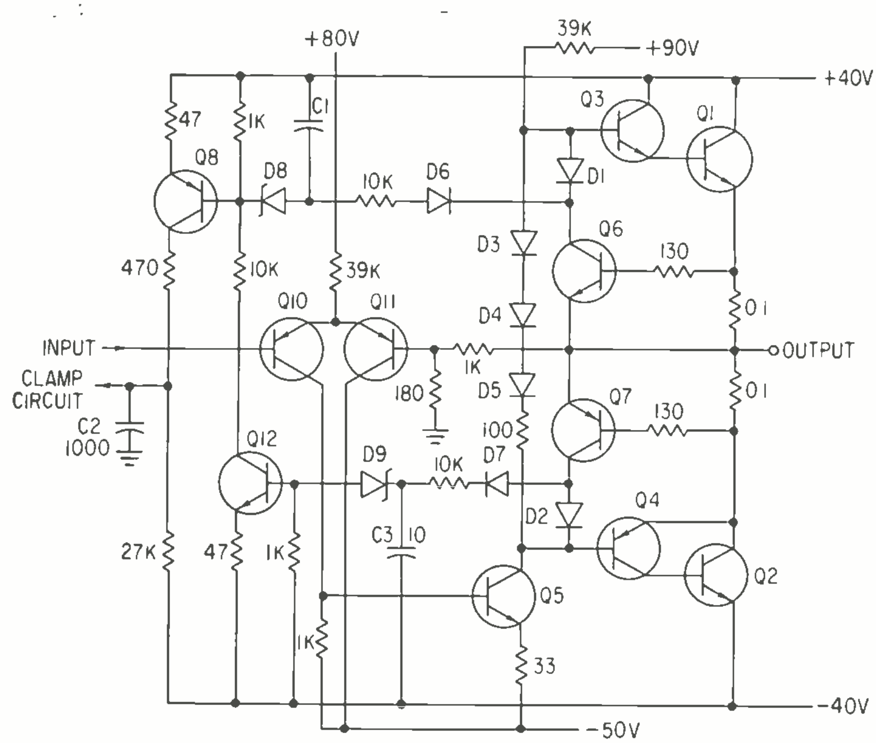

Figure 3. A completed circuit, with typical values inserted, to cater for full protection of both sets of output transistors.

Now there remains a problem in providing the clamp to kill the audio. In FIGURE 2, signal is shown as clamped to ground (by Q9), from a positive signal direction, because a positive-going signal is what will cause trouble in Q1, Q3. The same clamp will not work for negative-going signal, to protect Q2, Q4. In fact, unless the emitter-base zener voltage of Q9 is adequate to accommodate negative-going voltages, they will be clipped by it, dangerous or not.

The circuit needs a two-way clamp, that brings signal voltage down to ground, from whichever direction it momentarily happens to be. Such a circuit is shown in FIGURE 4. Each clamp is prevented from clipping or conducting in the opposite direction by a diode in series with its collector.

The positive-going clamp, Q9, is direct-acting, as before. The negative-going clamp, Q14, uses a phase-inversion with a transistor, Q13, connected as emitter-follower to inject clamp current into Q4's emitter, instead of the base of the clamp transistor, as at Q9.

One more possibility may need safeguarding. The circuits so far presented provide protection against too much current or voltage, in the normal directions. Normally, each transistor controls current flow and corresponding voltage drop, in its formal operative direction. But if the load is reactive, current in the usual direction can produce voltage kicks that exceed the

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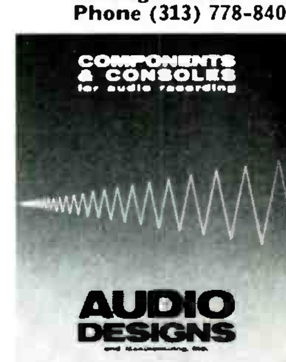

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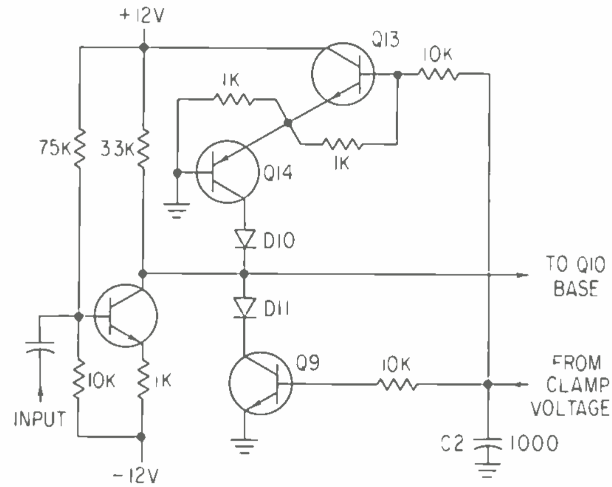

Figure 4. The double clamp circuit needed, to go with Figure 3.

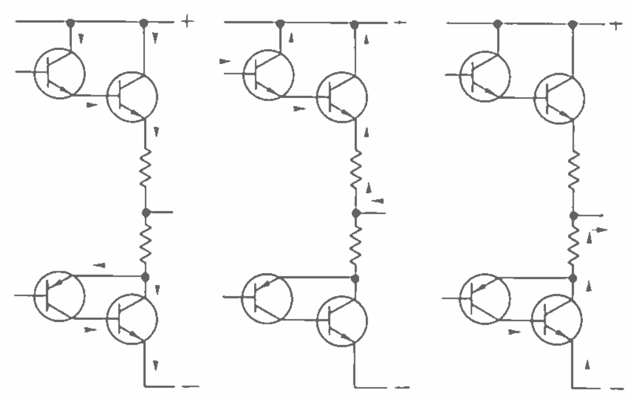

Figure 5. What can happen when voltage kicks exceed either supply voltage: (A), current paths when operating normally; B, current paths when signal voltage at output kicks above supply positive; C, when signal goes beyond supply negative. Double-headed arrows represent danger currents.

supply voltage that got the current started.

This means that the voltage across the load can, for a moment swing beyond the supply voltage on the other side of the transistor (FIGURE 5). This is a danger point. Now the transistor conducts freely in the opposite direction, and base input no longer controls it. The functions of emitter and collector are suddenly transposed, and base can go haywire.

Recirculating the output current in this way can blow the drive transistor and next the output transistor goes, as a secondary effect. Some means must be adopted to prevent the excessive voltage swing that starts this chain reaction. One way is to use diodes that conduct before the voltage reaches this point (FIGURE 6).

The voltage at a transformer tapping further from ground, or supply center, than the output tapping, will swing beyond voltage positive or negative before the output tapping voltage itself does. By connecting the other side of diodes from this higher (audio-wise) tapping to supply positive and negative, they conduct and prevent voltage from going any further before the output point of the amplifier reaches the danger point.

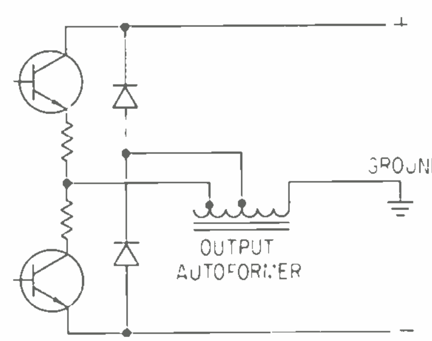

Figure 6. A relatively simple circuit to protect against the voltage kicks and reverse currents illustrated in Figure 5. Because the output impedance is close to required load values, an output autoformer can be designed more economically (and at much less cost) than the traditional output transformer of tube days.

These diodes are built to withstand the reverse voltage necessary under normal operation (the double supply voltage, with a safety margin) and the full output current necessary under protective operation.

That is probably enough about transistor output circuits for the time being. Next time we will turn to one of the other things that readers have been writing in about. ■

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Sound with Images

MARTIN DICKSTEIN

IMAGE BRIGHTNESS

● The effectiveness of any projection system, whether front or rear projection, is judged generally by the ability of the audience to read printed material shown on the screen, or, in other words, by the *intelligibility* of the projected material.

There are suggested standards for the size of lettering to be used, recommended thickness for lines in graphs and diagrams, and templates available to indicate the best area to use for the printed message, but these factors still depend on the performance of the projection system to provide the viewers with satisfactory images.

The term *luminescence* is used for image brightness and is determined by the amount of light either reflected from the screen (front projection) or transmitted by the screen (rear projection), divided by the area of the image. (The foot-Lambert is the unit used to define the brightness, and it is assumed that the slide being projected is the standard 35mm or double-frame size.)

Although each viewer may have his own ideas for how bright a slide image should be, depending on the material being shown, there should be some value assigned to the acceptable brightness of a slide image. There is no actual prescribed value, but a minimum of 10fL for slides has been suggested by Kodak. (The SMPTE recommended values for 35mm and 16mm film are 16fL plus or minus 4 and plus or minus 2 respectively.)

It has been found that if the image of films is too bright, the viewer is disturbed by what seems to be a flutter or flicker. Depending, of course, on the darkness of the room and the speed of the shutter, a brightness greater than about 20fL becomes distressing. For slides, however, the image brightness has no top level as far as the viewer is concerned. The brighter the better, it seems. However, it must be remembered that there will be times, in the course of presenting many slides, that some will seem to wash out when the image is too bright—with a possible loss of readability. In most instances, though, the brightness will be desirable, especially with printing on different colors of background.

The image brightness levels given are for blackout conditions in the theater, where the screen is illuminated by only 0.1fL of stray light. The stray light falling on a screen is usually higher than this (from the projector lamp where the projector is in the same room, from windows not completely blacked out, from lights at exit doors, ceiling lights to permit taking of notes, etc.). Thus, in order to maintain a proper balance between the brightness of the image and the ambient or stray light hitting the screen, consideration must be given to the type of material to be projected, as well as establishing a satisfactory ratio between the image brightness and the non-image brightness of the reflected (or transmitted) light. (Some of the causes of stray light on a rear-screen projection system are a work light turned on by someone behind the screen without realizing the problem this can cause, or the opening of a corridor door leading into the rear-projection booth where no precautions have been taken to prevent the outside light from reaching into the booth. Painting the walls of the rear-projection booth black helps to eliminate bouncing around of stray light—even from the projector itself.)

In order to determine the levels of light, both desired and undesired, coming from the screen, it is suggested that a foot-Lambert meter be used to measure the reflected and transmitted light. (Standard light meters usually are made to read footcandles, or incident, not reflected light.) A reading should be taken of the light coming from (or through) the screen on or close to the axis of the projected center of the image. This is done by turning on the slide or film projector with no slide or film in the unit. (The room should, of course, be darkened to normal projection illumination.) This is the image, or desired, brightness level.

A second reading is then taken with the foot-Lambert meter of the light being reflected (or transmitted) by the screen with the projector still on but with the lens capped to prevent source light from hitting the screen. (The projector is left on so that any stray light coming from the projector will be taken into account.)

Dividing the first of the two readings by the second will now provide a screen-brightness ratio. (Note that in taking reflected light readings, the entire system has been accounted for including the screen and the material of which it is made, the light source, the optic system, etc.). This ratio will act as a guide to determining if the system is providing sufficient brightness for the material to be presented.

Different values have to be given to provide a guide for determining the ratio required for different material. Where the image will be required to show good details in colors or in various shades of gray, the minimum brightness ratio might be considered to be 100. For less detail, whether in color or gray areas, a value of 25 might be used. For graphs or charts or typewritten material, the ratio might be 5.

By multiplying the stray light brightness by the proper factor based on the material to be shown, a minimum screen brightness level is determined. If, for example, the stray light is found to be 1fL and the material to be presented requires a screen brightness ratio of 100, the screen brightness level should be 100fL. If the material to be projected falls into the category where the screen brightness ratio should be 5, projecting in a room with 1fL stray light on the screen would result in a screen brightness of only 5fL which is well below the suggested minimum of 10fL for slides. Increasing the source brightness, or any other factor, to raise the ratio to the recommended level would only result in an extremely contrasty image with resulting loss of legibility. The ratio can only be raised properly by increasing the stray light around the screen (but not allowing the additional lighting to fall directly on the screen) and then increasing the system brightness to the proper level to establish the proper ratio required above the minimum recommended level.

There are means to estimate the stray light and to calculate the output brightness of the projection system so that the required ratio can be determined without the use of the foot-Lambert meter. This will be the subject for another discussion in the near future. ■

The Sync Track

JOHN M. WORAM

MONITOR SPEAKER STANDARDIZATION

● No matter what standards our industry may now or later adopt, it's a pretty safe bet that there will never be an industry-wide standard speaker system.

You won't get into much trouble insisting that from this day on a dBm shall be 1mW through 600 ohms, and everyone will nod agreement when you proclaim that it would be a good idea for all tape machines to run at 15 in./sec. But, sometime when you're bored try calling all your friends and revealing that your speakers should henceforth be considered the industry standard. You may not get very far. Naturally, your speakers are the finest, and the rest of the industry is tone deaf anyway, but you'll never convince the others that only you have the correct system.

Part of the problem is that speaker standards are unlike say, tape speed—where one speed might be as good as another. There is no particularly scientific reason for choosing 15 rather than 14 or 16 in./sec. Were it not for the obvious necessity for a standard, any old speed would do nicely. That's about the way it is now with speakers. Everyone has their favorite design and type, and no one feels inclined to change, since speakers don't have to conform to a standard in order to be compatible with other speakers. In fact, they are best judged subjectively, the way music itself is. Once certain minimum requirements have been met, the choice of one speaker over another depends on personal preference, which may be difficult to define, let alone measure.

It's reasonably easy to follow a signal from microphone to speaker, where it is converted into acoustic energy and launched on its way to your ear, there to be converted into who knows what. Depending on you, the information received will warm the heart, or turn the stomach. Before the ear, its easy to make measurements. Behind the ear, it's a little more difficult, and even measurements that can be made are often overshadowed by those that can't. How then, can we expect to prescribe the standard speaker, which is where the subjective begins?

There are a few points that can be made. For example, we're all against distortion (unintentional distortion, that is). This has nothing to do with our musical tastes. A distorted rock group is just as bad as a distorted string quartet, and if the rock listener doesn't complain as loudly, it's more a commentary on his personal tolerance level than on the acceptability of the distortion.

As speakers, and all that other stuff, have gotten better, distortion figures presumably have been minimized. Now, along with other data, some manufacturers include a graph to show you how well their speaker passes the frequency run test in an anechoic chamber.

But, suppose you are one of those poor few who don't live in an anechoic chamber? What then? Then, as the sound meanders from the speaker to your ear, it is *equalized* by your listening room. Every window, drape, wall, and piece of furniture or equipment throws in a few dB (+/−) until by the time the sound reaches your ear, any resemblance to a flat response curve is coincidental. Quite often, the same speaker in different rooms will produce greater differences than that between different speakers in the same room.

All of which brings us (finally) towards the point of this month's

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Scott Muni
WNEW-FM, New York

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Tracking Force	3-7 grams
Cartridge Weight	5 grams
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STANTON

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column. If we can't all agree on the ideal speaker system, we can at least agree that each listening room imposes a different set of conditions on the loudspeaker/listener combination. This means that even if we could define the standard speaker system, we would be compelled to include the listening room as part of the system. The plot thickens —even if you could get everyone to use the same speaker, try getting everyone to build identical listening rooms! One studio—The Record Plant—has done exactly that. Their control and mixing rooms (studios too) on the east and west coast have been built to the same blueprints. Since today's studio personnel are quite mobile, if each studio imposes a different sort of listening environment, it can make work difficult at best. Presumably, Record Plant people don't have this problem. For others, the solution is often to make final decisions after listening at home on the living room hi-fi. It may not be great sound, but at least it's constant, and once you're used to its little idiosyncrosies, you have a, sort-of, reliable frame of reference.

To help get around this problem, many studios not prepared to follow the Record Plant's example are now running response curves in their control rooms to find out exactly what influence the room has on the speakers' frequency response. Once measurements have been made, corrective equalization is inserted—usually before the power amplifier—until the combination of equalizer, amplifier, speaker and room yields as flat a frequency run as possible. Depending on whose system you use, the process is known as *Acousta-Voicing* or *Environmental Equalizing*.

To find out just what a little equalization would or wouldn't do, I recently tricked Frazier representative Jim Loder into analyzing and correcting my living room listening conditions. It seems to me I may have given him the impression that if I liked the system, I might buy units for all my friends. He found out too late that I didn't have any.

Surprisingly, a 2/3-octave analysis showed that the room wasn't really so bad. Actually, I had no complaints with the sound I was getting. However, after Mr. Loder made the required adjustments on the Frazier model SEE-24 Equalizer, no one had any trouble detecting the equalizer's effect during an a-b test. The increase in clarity was remarkable, far beyond what I should have imagined. This is not so much a testimony to any remarkable qualities within the Frazier unit as it is to the soundness of environmental equalization. Of course, the Frazier is an excellent unit, and the complete set of 2/3-octave filters on one chassis means that you can re-equalize everytime you change speakers or other listening conditions. However, any device that will help flatten out your peaks and valleys is worth investigating.

Depending on the condition of your listening room (and budget) there are units (Frazier or others) available that allow 2/3-, 1/3- or even 1/6-octave compensation. For most professional applications, perhaps a 1/3-octave analysis and compensation would be adequate.

Now then, this month's column began in speaker standardization, and here we are talking about room equalization. Since I'm supposed to be making some sort of point, let's remember that the reason for any standard is to minimize compatibility problems and eliminate ambiguities between two or more systems, speakers or whatever.

Since all the listening rooms in the industry will never be identical in speaker complement and/or physical layout, the way to avoid the vagaries of each location's personality is to get out the spectrum analyser and the noise generator, see what's happening, and make the necessary corrections. Then, as one moves from location to location, the monitoring conditions will probably be as close to standard as one can realistically hope for.

If you still don't like the high end, maybe it's time for a behind-the-ear checkup. ■

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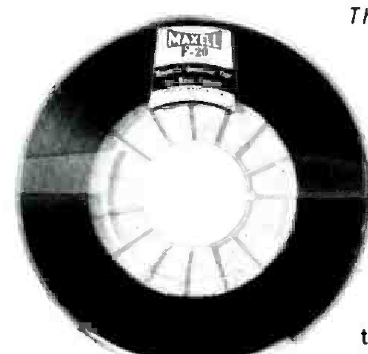

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Powering Condenser Microphones

GERHARD BORE

The user of transistor-equipped condenser microphones must decide among three possible systems, something which certainly runs contrary to the purpose of standards. This article investigates the reasons which have led to this situation as well as the pros and cons of the three systems.

According to DIN 45 595 *Connection of Transistor Equipped Microphones with Modulation Lead Powering*, solid-state equipped condenser microphones are powered by sending the necessary d.c. along the same conductors which serve the audio output of the microphone. The two wires are traversed by the current in opposite directions; that is one lead provides the current supply and the other the return leg.

In the DIN 45 596 standard *Connection of Transistor Equipped Condenser Microphones using Multiplex Powering*, another method is suggested in which the d.c. is supplied through the electrical center of the audio output leads; that is, half the current flows in each wire and is returned through the cable shield. Two voltages are suggested: 12V and 48V.

Each of these powering systems have one thing in common: they require but a single shielded pair of wires. Up to now,

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Dr. Gerhard Bore is the chief physicist of Georg Neumann GmbH, Berlin. The paper which originally appeared in German in Elektronorm, *Vol. 23 (1969), No. 7 was translated by Stephen F. Temmer, president of Gotham Audio Corporation.*

condenser microphones have needed five (or even six) conductors, while dynamic microphones have always used but a shielded pair. It is this very desirable simplification of microphone connections and cables which has led to the criticism of the modulation lead powering recommended in DIN 45 595. For such powering methods wire No. 1 carries the plus pole (+), and wire No. 2 the minus pole (−) of the d.c. supply voltage as well as the audio modulation. Outlets so equipped therefore cannot be used directly for the connection of other microphone types, since dynamic microphones and all microphones equipped with an output transformer produce severe distortion if the supply voltage is not disconnected. Ribbon microphones and certain moving coil units will furthermore suffer permanent damage.

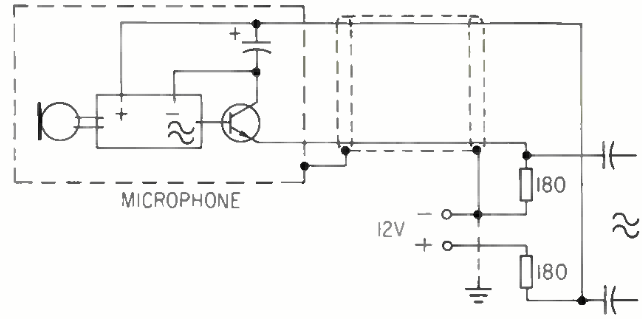

Figure 1. The modulation-lead powering standard according to DIN 45 595.

MODULATION LEAD POWERING ACCORDING TO DIN 45 595

The modulation-lead powering standard was decided upon at a time when transistor-equipped condenser microphones were only possible using the rf principle. By standardizing the power-feed resistors and their electrical arrangement, the output circuitry of such microphones was also determined in advance as FIGURE 1 shows.

The cable shield is connected to the microphone housing, but not to the circuitry within the microphone itself. The output symmetry with respect to ground prohibits the connection of either modulation lead to the cable shield. Only the "electrical center" may be so connected. This electrical center on the other hand, is only accessible in the remotely located power supply branch-off, and is shown here as the two resistors.

It is an *advantage* of the modulation-lead powering system that no current is fed through the cable shield; a *disadvantage*, on the other hand, that the audio output and d.c. supply voltages are mixed within the cable. As a result the supply voltage must be filtered most carefully, since it is in parallel with the audio modulation. If more than one microphone is to be supplied from a common power supply, a decoupling network must be used to prevent audio crosstalk between microphones.

With the advent of the field-effect transistor (fet) it has become possible to construct transistor-equipped condenser microphones using circuitry of conventional audio-amplifier design known and used for decades in tube technology. The input circuits of such fet equipped condenser microphones are high impedance and are protected against noise interference by the microphone housing. To produce really effective shielding, it is absolutely necessary that the housing (chassis) and 0-volt potential be identical. It is for this reason that all tube-equipped condenser microphones have a 0-volt/chassis connection.

Should one wish to construct fet-equipped condenser microphones for modulation lead powering, problems arise which were not predictable when the DIN 45 595 standard was promulgated:

Due to the missing 0-volt/chassis connection in the microphone, noise which appears along the cable shield will reach the gate of the fet as a result of the stray capacitance between microphone case and amplifier input, and will be amplified along with the modulation. Beyond that, such stray capacitances adversely influence the balance of the microphone output, even if their values are but fractions of a picoFarad. This is due to the fact that the relatively high impedance of this stray capacitance between gate and microphone housing is stepped down by the transistor stages along with the capsule capacitance, so that output balance will be affected adversely by *it*, and not, as it is indicated in DIN 45 595: "principally by the balance of the 180-ohm power feed resistors."

For microphones of simple construction, these disadvantages can be negated through the use of double shielding of the case, in which the inner shield is connected to the zero potential of the first amplifier stage. More complex microphone types such as switchable characteristic or stereo units would be structurally difficult to produce, since such a second shield might even have to be transparent to sound.

MULTIPLEX POWERING ACCORDING TO DIN 45 596

In multiplex powering both modulation leads are at the same potential (see FIGURE 2). The principal advantage lies in the fact that dynamic microphones as well as all microphones equipped with an output transformer such as tube-type condenser or ribbon units may be connected, wthout the necessity of disconnecting the d.c. supply voltage.

Since the powering current flows equally through both modulation leads to the microphone, it is completely decoupled from the audio modulation which at any given instance flows through the audio pair in opposite directions. The supply voltage may therefore contain significantly more ripple and noise without influencing the audio signal flowing through the cable.

The advantage of using 12 volts as the supply is that this voltage is more usually available in existing powering systems. The d.c. power supplies found in studio consoles, portable tape recorders and wireless microphone transmitters may be used directly for powering. On the other hand such condenser microphones using straight forward audio amplifiers must be equipped with d.c./d.c. converters or batteries for producing the capsule polarizing voltage of about 48 volts, and as a result consume some 80 mW of power.

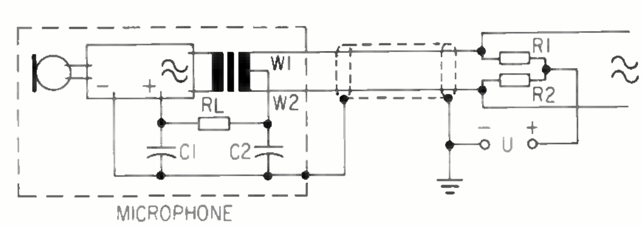

Figure 2. Multiplex powering has both modulation leads at the same potential. This permits the use of dynamic and other output-transformer equipped microphones without the need for supply voltage disconnection.

The other multiplex powering voltage, 48 volts, will not be readily available, and will have to be produced in a.c. supplies, battery boxes or d.c. converters. It would therefore have been obvious to recommend the use of the 12-volt system, *if* the use of 48 volts did not have the following significant advantages:

1. Condenser microphones using normal audio frequency circuitry (with fet) can obtain their polarizing voltage directly from the 48 Volt supply, thereby saving numerous components, consuming only 19 mW of power, and permitting considerably smaller size.

2. The higher voltage and lower current permit very small but effective filtering components for the supply voltage to be incorporated within the microphones themselves. This obviates the necessity for special filtering and decoupling networks for each microphone, and prevents any crosstalk via the supply.

3. When compared to 12-volt systems, power supplies may be constructed much more simply for the same reason. Due to the very high values of the power feed resistors R_1 and R_2, central power supplies need to be only slightly over-dimensioned to make sure that an inadvertent short in one of the microphone cables will not affect the operation of the other microphones on the same supply.

4. Only 48-volt operated multiplex powered microphones offer the possibility of operating such transistor units from power supplies intended for the powering of tube-type microphones. The outlets for such tube units may be altered by installation of simple voltage dividers to provide the minimal current (0.4 mA) needed by the 48-volt fet units. Installations using standard tube-type microphone outlets may therefore operate a mixture of tube and fet types without regard to changing power supplies.

A voltage higher than 48 volts was not recommended due to the need for battery powering. Close to 48 volts may be obtained from the series connection of two 22½-volt batteries available everywhere in the world, while higher voltages could only be obtained from special batteries or d.c. converters.

PROBLEMS OF POWERING VIA THE CABLE SHIELD

The advantages listed under 2. above, of building the filtering components for the supply voltage into the microphone itself, is especially of value when the current flows through the cable shield. Such a supply system is new to studio technology, and it was therefore necessary to investigate whether and under what conditions such a powering method could lead to problems in operation.

The d.c. through the cable shield produces a small voltage drop which might fluctuate slightly during operation if, for example, two connectors along extended cable runs of two adjacent microphones touch for a moment, shorting out this voltage drop for an undefinable period of time. The result is fluctuation of the supply voltage by the amount of this voltage drop.

Besides this, undesirable currents might be caused to flow through the shield should a connector or the microphone housing come in contact with a metal object which is at a different potential from operating ground. This might occur if the console and its shields are at ground potential and a connector from a microphone cable comes in contact with a pipe of the central heating system or an electrical device grounded to the protective a.c. ground. This would cause a.c. with power-line frequency to flow through the cable shield.

An alternating current can also be induced in the cable shield by magnetic fields if the shields of two microphone extensions come in contact with each other outside the console. The ground loop thus formed has a very low resistance, allowing induced currents which may lead to noticeable interference.

The currents described influence the balanced and floating modulation leads within the cable only very little. However, as soon as a part of the a.c. also flows through the 0-volt lead of a microphone powered by d.c., noise is coupled into the supply voltage. This produces the same effect as if the microphone were powered from an a.-c. supply with insufficient filtering. This noise can be completely removed by including filtering components for the supply voltage within the microphone themselves. Estimates as well as lab experiments have shown that components which have a filtering factor of 10,000 at power-line frequency, will keep out interference of the strongest kind introduced via the cable shield.

At first, ideally-balanced application of the poorly filtered supply current was assumed. Should one wish to maintain the commonly obtained circuit balance while still permitting strong random a.c. noise currents in the supply, it is sufficient to introduce a series impedance R_L as in FIGURE 2.Its a.c. resistance must be so large as to cause a voltage drop across *it*, rather than the multiplex feeding resistors W_1R_1 and W_2R_2. In FIGURE 2 this means that C_2 must be dropped and R_L made adequately large.

Summing this up: the use of multiplex powering in microphones, using the cable shield as the supply return, may always be used without concern if the d.c. supply current to the microphone is passed through an effective filtering circuit, and is connected through an adequately large unbypassed series impedance.

Both of these requirements are easily met in microphones for 48-volt powering. The relatively high voltage and low power drain allows a series impedance R_L to be a pure resistance, while the filtering within the microphone may consist of a two stage r.c. filter using dry tantalytic capacitors of smallest dimensions. Should R_L be replaced by a solid-state circuit which maintains constant current through it, then the supply voltage may even have poor regulation and hum filtering.

In microphones using 12-volt multiplex powering, only a very small voltage drop is available for filtering! One must bear in mind that depending on microphone current consumption, as much as 3.4 volts are dropped across the supply resistors R_1 and R_2 of FIGURE 2. Therefore both the filtering and series impedance R_L must be accomplished by means of electronic circuits with numerous components, which (at best) will still produce a voltage drop of at least 1 volt.

The voltage remaining, while still adequate for the operation of the microphone, produces problems as far as optimizing the circuit with regard to overload and self-noise level. In practice, therefore, one will be hard pressed to allow disturbances caused by induced currents in the cable shield, rather than to adversely affect the principal operating parameters of the microphone's transmission chracteristics.

The agreeable advantage of being able to utilize existing supply voltages for the 12-volt multiplex powering system is therefore accompanied by the need for complicated and bulky circuits as well as greater susceptibility to noise interference. Only after a considerable amount of practical experience will it be ascertained which advantages and disadvantages will turn out to play a key part and which will take on subordinate roles and thus be negligible. ■

A Multi-Speed Capstan Drive System

H. VAN DER WAL

The author describes the design of a capstan drive system for an audio tape recorder that permits variable speed drive. It will be available later this year in both stereo and mono configurations.

THE CHOICE of a driving system for an audio-recorder depends on the demands made on it. The following factors should be considered: wow and flutter, reaction to abruptly changing loads, ability to be switched for various speeds, supply as regards an external reference, and synchronisation during playback by means of a signal recorded on tape.

Wow and flutter require no detailed explanation.

Reaction to abrupt changes in load is one of the things to be considered in coupling the tape to the capstan by means of the rubber pressure roller. The switching phenomena must then be short and damped. This sort of small shock-loads can also occur on tape joints etc.

The ability to switch to various speeds naturally depends on the demands which are made on the recorder, but a tendency towards three or more speeds is noticeable, all the more so as, thanks to the new tapes, such as chromium dioxide (CrO_2) tape, professional quality can be achieved at a speed of 3¾ in./sec. as well. For copying purposes speeds above 15 in./sec. are desirable.

The drive system of the tape using an external reference is all the more important where one cannot be sure of the a.c. frequency or where it is desired to vary the tape speed arbitrarily.

Synchronisation by means of a pilot tone recorded on a tape is of interest if synchronism between picture and sound is the aim.

For the purposes mentioned a choice can be made from a number of systems. The most commonly used is still the synchronous motor, supplied directly from the mains. The external reference frequency is a mains frequency and cannot be varied. The range of speed and, above all, also the switching phenomena (oscillations) leaves much to be desired. If the synchronous motor is fed through an amplifier, the speed can at least be varied, and if special steps have been taken as regards the motor design, a large range of speeds can also be obtained. It is better to use a regulated system in which the capstan generates a *signal* which can be compared in frequency and preferably also in phase with a *reference signal*. The control signal for the driving system is derived from the difference between both signals. For instrumentation recorders a direct-current motor is often used. In this case, however, use is made of an asynchronous motor which drives the capstan. A perforated disc with a photodiode and lamp for generating the comparison signal and a disc, acted upon by a magnet to give the effect of a Foucault brake, are

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mounted on the capstan. This magnet is used to control the system.

In this case the capstan motor can be identical with the winding motors and compared with the d.-c. motor the power driven is relatively small. The elements are shown in the block diagram of FIGURE 1. These are the reference frequency; the frequency generated by the motor—the respective speeds are decided by the interposition of dividers; The comparison circuit, which is discussed below in greater detail; and the drive of the brake-magnet, with a number of corrections which is also to be dealt with in greater detail.

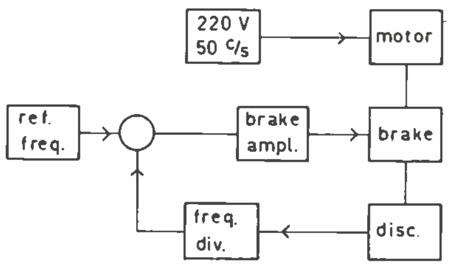


Figure 1. The elements of the capstan drive motor.

The circuit which determines the difference in frequency and phase between the two signals is shown in FIGURE 2.

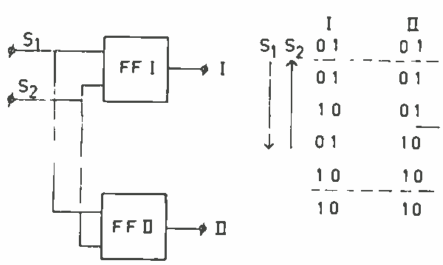


Figure 2. The circuitry that determines the difference in frequency and phase between the two signals generated from the motor and the reference.

This consists of a 4-position counter which, on receiving one signal, counts in the one direction and on receiving the other signal, counts in the other direction. The two end positions are fixed.

The signal at the output of the comparison circuit will therefore be as shown in FIGURE 3.

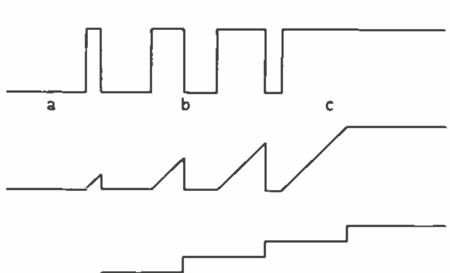


Figure 3. This is the signal that will be seen at the output of the comparison circuit. At (a) it is too slow; at (b) the same speed; and at (c) too fast.

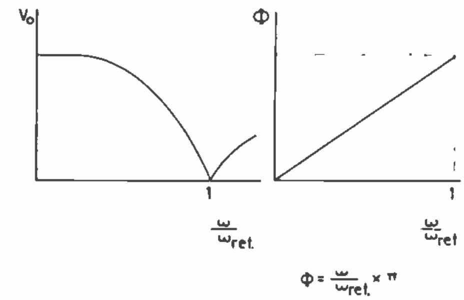
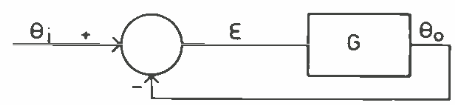


Figure 4. The derived drive signal.



$$G = \frac{\theta_o}{\varepsilon} = \frac{\theta_o}{\theta_i - \theta_o}$$

$$A = \frac{\theta_o}{\theta_i} = \frac{G}{1+G} \qquad A + E = 1$$

$$E = \frac{\varepsilon}{\theta_i} = \frac{1}{1+G}$$

Figure 5. When a control system is indicated in this way in a block diagram, the included formulae apply.

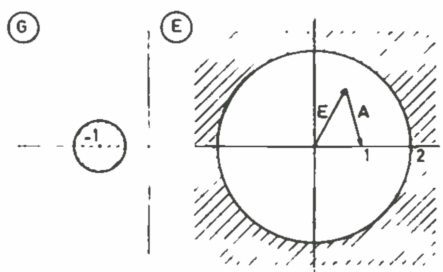


Figure 6. G is normally plotted in a Nyquist diagram.

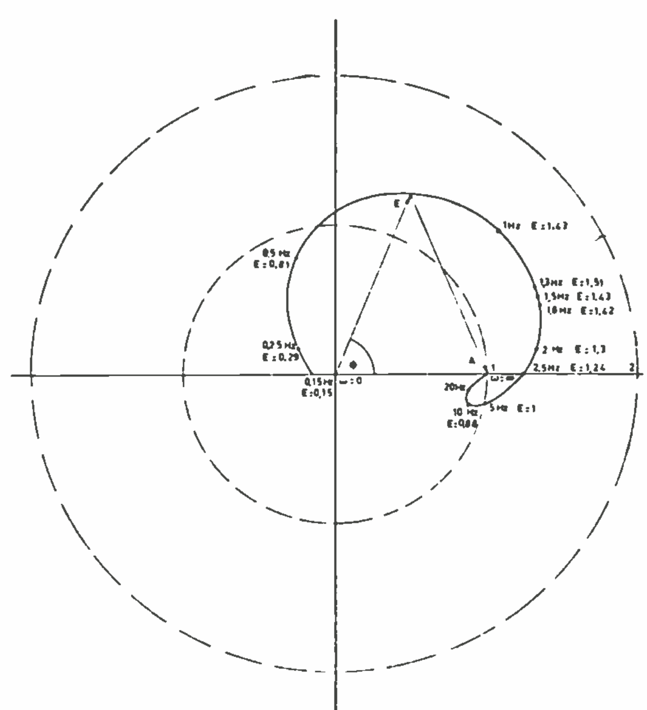


Figure 7. The AE diagram of Figure 6.

The width of the pulse is a measure of the difference in phase. By converting this signal into a sawtooth that is sampled at the peaks, a drive signal is obtained for the motor with a maximum frequency range and minimum phase shift of only $\frac{\omega}{\omega ref} \times \pi$ (See FIGURE 4).

Some phase corrections are necessary in the drive amplifiers, to assure stability of the system. The results are most simply illustrated in FIGURE 5.

Normally G (in FIGURE 5) is plotted in a Nyquist diagram in which the normal design criterion can apply that a circle with a radius of 0.5 must not be intersected around the point. This means that in a diagram of A and E (FIGURE 6), this must not lie outside a circle of radius 2 around 0. The advantage of the AE method is that measuring can be done with a closed loop.

The AE diagram (FIGURE 7) of the control system discussed here satisfies the requirements as to stability. For comparison, FIGURE 8 shows an AE-diagram of an instrumentation recorder in which the higher frequencies (which are required here) stand out.

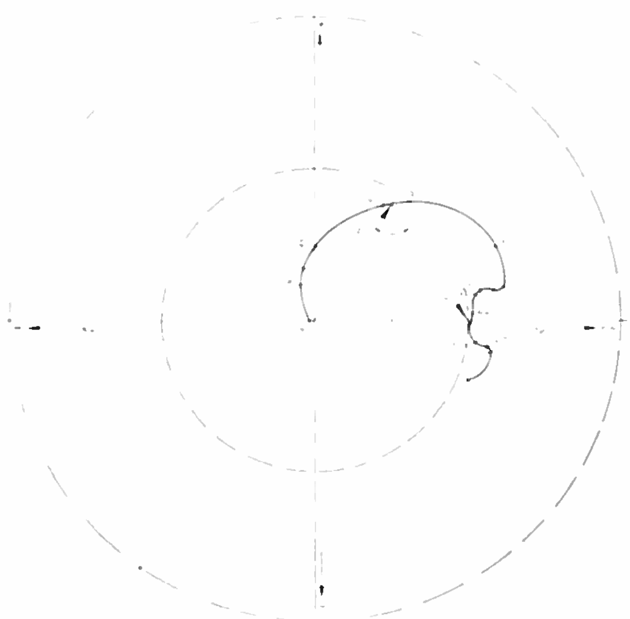


Figure 8. An AE diagram of an instrumentation recorder.

Finally, data tested for a tape recorder using this control system are shown. FIGURE 9 shows the wow and flutter measured at various speeds, and to finish off with, the starting phenomena compared with those of a synchronous motor are illustrated in FIGURE 10.

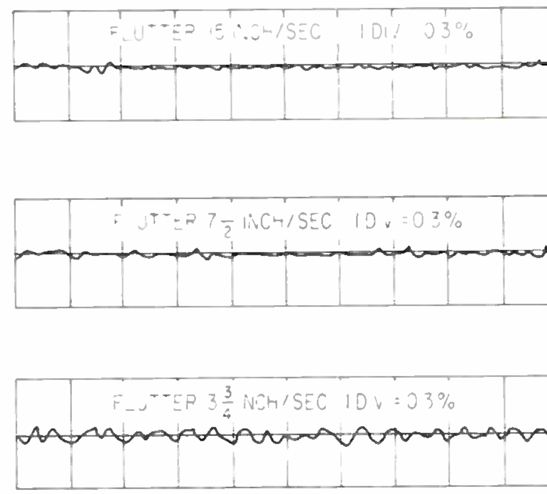


Figure 9. Wow and flutter measured on the discussed machine at various speeds.

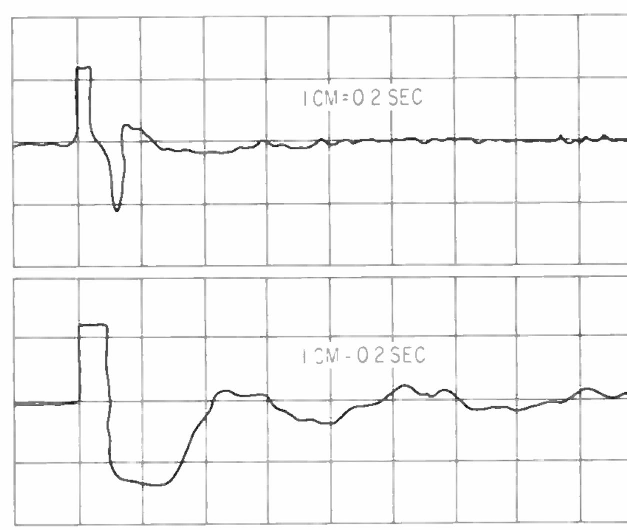


Figure 10. The starting phenomena compared with those of a synchronous motor.

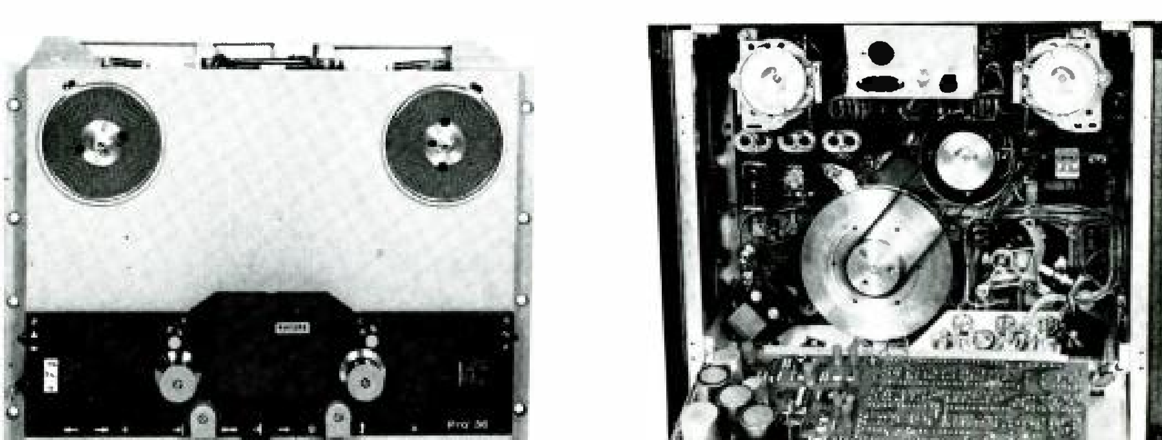


Figure 11. Front and rear views of the Philips PRO 36 transport utilizing the capstan drive described in this article.

Hearing Loss and Audio Engineering

W. DIXON WARD

This is the second (and concluding) part of this article dealing with the state of knowledge existing with this controversial subject.

DETERMINE YOUR OWN TTS

The present thinking of several of us who are actively pursuing research in this field is that a noise exposure is dangerous if either 40 dB or more of TTS_2 is produced, or the ear has not fully recovered from TTS by the next morning, but instead is re-exposed while still fatigued. Generally, though not always, these two symptoms go together. That is, if the TTS for a given ear exceeds 40 dB, then full recovery usually requires at least three or four days. However, under some conditions, especially for spaced intermittent exposures, the TTS_2 at the end of the day may be as little as 20 dB and yet require more than overnight for full recovery.

The best method, then, for determining if a noise is dangerous to *you* is to measure the TTS it produces. The first problem is to obtain a resting threshold measurement at 1, 2, 3, 4, and 6 kHz as free from temporary losses as possible. The time to do this is early on Monday morning, preferably after a 2- or 3-week vacation—assuming, of course, that you have not been cutting your year's supply of firewood with a chain saw, driving a racing auto, shooting a lot of skeet, or the like, during your vacation. Then measure the thresholds at these same frequencies at the end of the day to establish the initial TTS, and on the following morning as well. If more than 5 dB of TTS remains after the 16 hours of rest, then such exposures, if repeated day after day, are hazardous and a reduction in exposure is called for.

A commercial audiometer, while desirable, is not essential, since only relative, not absolute, values of your hearing sensitivity are called for. The main requirements are an oscillator, a voltmeter, a 100-dB attenuator, and an earphone that is not going to be used for any other purpose—*and* a quiet place.

Usually, since the ear is most sensitive to frequencies from 2 to 4 kHz, both temporary and permanent hearing losses first develop at 3, 4, or 6 kHz, and any reasonable studio will offer an ambient noise level well below what is necessary before tones at these frequencies begin to be masked.

The oscillator is set first at 1000 Hz, and the attenuator set at 10 or 20 dB. Then the signal voltage delivered to the attenuator is adjusted, by means of the gain control on the oscillator, to give some value (which you record) that produces a moderately loud tone (*moderately loud* will probably turn out to be about 80 dB SPL, although the absolute value is not important here). Next, with the tone in, say, the right ear, adjust the attenuator to where the tone is just barely audible. Record the attenuation value, and then repeat the process for 2, 3, 4, and 6 kHz; a reversing the headset will allow you to test the left ear in a similar fashion.

Then in the afternoon and the following morning, again set the oscillator to 1000 Hz and the voltage to the same value as before, and repeat the measurements. If a difference of 25 dB at any frequency is found in the afternoon or if a difference of 10 dB persists after an overnight rest, the noise exposure is probably dangerous, and should be materially reduced.

Don't worry about a change of only 5 dB. You can't get the earphones on your head exactly the same every time, so either an increase or decrease of this small a value probably does not indicate any change in your sensitivity. The reliability of your measurements can be increased somewhat by making three or four adjustments, removing and replacing the earphones between trials.

PRE-EXISTING HEARING LOSSES

If you find a large (30 dB or greater) difference between your Monday-morning attenuator settings for 1000 Hz and 4000 Hz, or between 1000 Hz and 6000 Hz, this indicates that you already *have* some hearing loss at the higher frequency. This shouldn't worry you much, because such high-frequency tonal gaps are rather common even among persons with no history of exposure to industrial noise, and could have happened years and years ago (gunfire or firecrackers are likely suspects as the cause). You may be surprised to discover this loss, because you never had any trouble hearing anything before (except, occasionally, your wife, and this affliction appears unrelated to auditory ability anyway), but fortunately 4000 Hz is not necessary for the perception of speech, and that explains why the deficiency failed to be

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noticed. I recall how astonished I was to find a 60-dB loss at 6000 Hz in my right ear when I first got hold of an audiometer.

If you do have such a pre-existing loss, however, I would suggest that you be even more careful than usual about high-level exposure in the future. Although there is no evidence that the ear with a loss from unknown causes is more susceptible to further injury than any other ear, you can't afford to take a chance, because any further increase in loss, to the point at which 3000 and 2000 Hz are affected, will begin to reduce your ability to understand ordinary conversation.

IS MUSIC DANGEROUS?

I wish I could estimate the probability that you will find your noise exposure to be hazardous. Unfortunately, however, I cannot. To my knowledge, no one has ever systematically studied either the noise exposures or the hearing sensitivity of audio engineers. It is certain that rock-and-roll music *in situ* can reach peaks of 110 dBA, and that an average level of 105 dBA is sometimes sustained for a minute or two.[5] If it were not for the intermittency of most exposures to this kind of noise, considerable hearing loss would be expected among those exposed to this music. However, audiometric studies of rock musicians have so far shown an incidence of hearing loss not much greater than in the general population, and then ordinarily only in those who admit doing a lot of shooting. So the intermittency is indeed important, although as we have seen there is no agreement among the experts as to *how* important.

Classical music—in an unamplified form at any rate—is known *not* to be a problem. An audiometric survey of all the musicians in three symphony orchestras[6] showed no greater incidence of loss than in the average population, except that nine violinists had unilateral losses (in the left ear: the ear held next to their own violins). So if you play classical music no louder than in real life, it is not likely to be at all dangerous.

One final comment should be made. Recording engineers do have one problem that is relatively unique: that of preserving hearing for the highest frequencies—i.e. those above 8000 Hz, which is the highest frequence ordinarily tested with a commercial audiometer. While it is true that only frequencies up to 3000 Hz are necessary for speech perception, the entire auditory range up to 20 kHz is deemed important in the field of high fidelity. The detection of high-frequency distortion, squeals or hiss in tapes and discs demand good sensitivity at high frequencies. It might be worth the effort, therefore, to keep track of your hearing at 8, 10, 13 and 16 kHz in addition to the frequencies recommended earlier. At these frequencies, however, the test-retest variability due to headphone placement will be quite large, so even apparent shifts of 10 dB may not be real; repeated testing (removing and replacing the earphones) is even more essential here. At present, little is known about the susceptibility of the average ear to hearing loss at these high frequencies, and any data that accumulates will be welcome.

5. Rintelmann, W. F. and Borus, J. F. *Noise-Induced Hearing Loss and Rock and Roll Music*, Arch. Otolaryngol., vol. 88, pp. 377-385, Oct. 1968.

6. Flach, M. and Aschoff, E. *Zur Frage berufsbedingter Schwerhorigkeit beim Musiker*, Zschr. Laryngol. Rhinol. Otol., vol. 45, pp. 595-605, 1966. ■

Picture Gallery—NAB 48th Convention

When there are over a hundred and fifty exhibitors spread around the exhibition halls of Chicago's massive Hilton Hotel, and only four days to see them, it is a busy time indeed. This year's convention and exhibition was held from April 5-8. Just as in years past, the time was hardly enough for us to see all we wanted to, much less push our cameras everywhere they ought to have gone.

What follows then is only a sampling of the equipment on display that might be of interest to db readers. Each of the photographs has a Reader Service Number which can be circled on the post card bound into this issue to secure detailed information directly from the appropriate manufacturer.

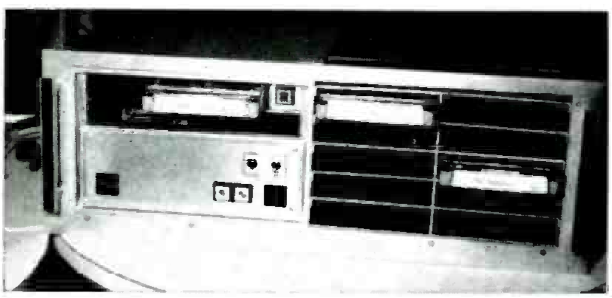
Just as was the case last year, the trend to automated broadcast continues to expand. A variety of complete station or sub-station setups can be had from several manufacturers. In addition, we observed continuing sophistication of existing equipment, particularly broadcast consoles—many equipped for stereo, many others equipped for multi-channel output and portability. There even was console equipment for four-channel origination (when that comes).

It should be understood that our pictures only show a sampling of what the represented manufacturers offer. It can be assumed that if one console or microphone is shown, there are others to cover a wide variety of applications.

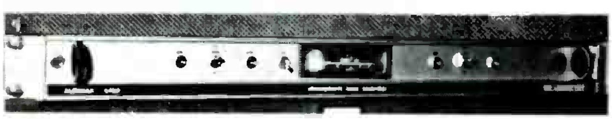
On this page — candid views around the exhibition halls. The following two pages present some of the equipment. If you desire specific information on the equipment shown, simply circle the reader service number that is appropriate on the card bound into this issue. You will receive the information directly from the manufacturer.

AEC-Veritas Documentor 210 records and plays back up to 24 hours on a single 9-in. disc. Circle 50 on Reader Service Card.

Philips Broadcast Equipment brought this portable 4-channel in mixer. Circle 63 on Reader Service Card.

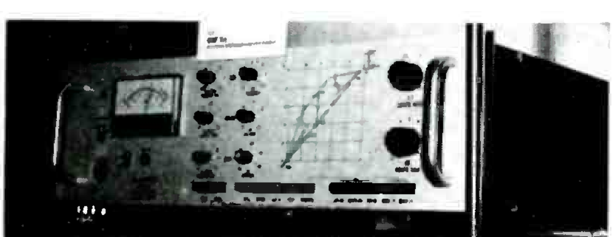


Broadcast Electronics Spotmaster Systems tape cartridge machines. Circle 51 on Reader Service Card.



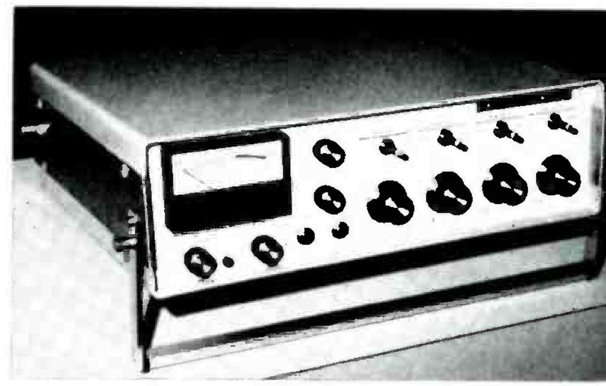
CBS Laboratories have a new thin-line version of their Volumax loudness controller. Circle 53 on Reader Service Card.

CCA Electronics brought new product lines, among them the Ultimate III console. Circle 54 on Reader Service Card.



Gotham Audio has the new EMT 156PDM that is at once a limiter, compressor, and expander. Circle 58 on Reader Service Card.

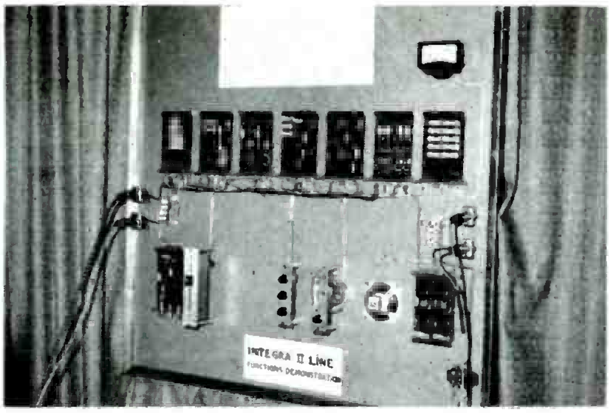
AKG, division of North American Philips showed its microphones, including the D202 dynamic. Circle 62 on Reader Service Card.



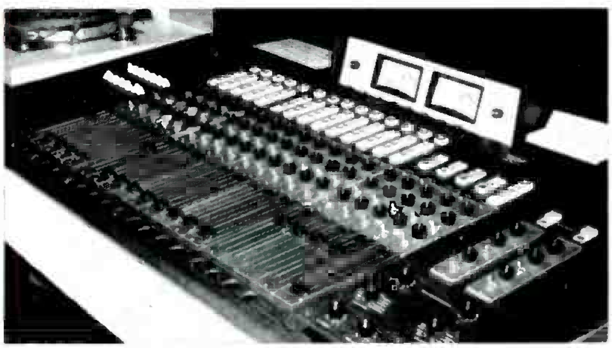
Collins Radio has a new control console, model 212J. Circle 55 on Reader Service Card.

Gray Research displayed turntable model 1012, in their line of disc playback gear. Circle 59 on Reader Service Card.

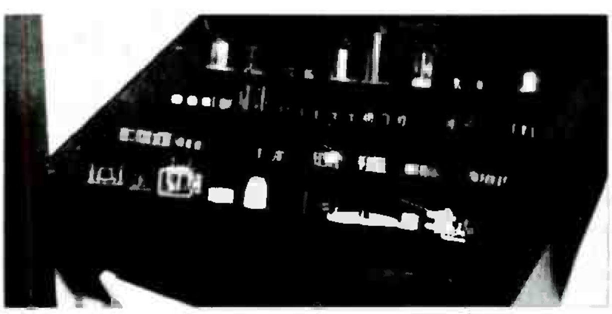
Gates Radio showed much, including the Criterion 80 line of cartridge machines. Circle 57 on Reader Service Card.



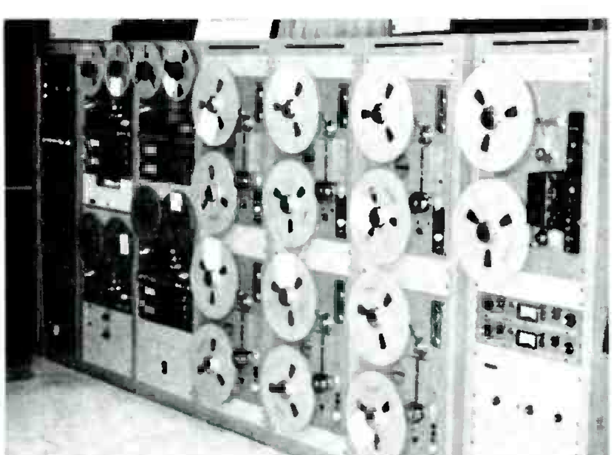
Fairchild Sound Equipment brought a functioning display board of the Integra II console components. Circle 56 on Reader Service Card.



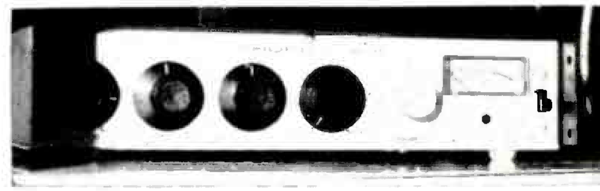
McCurdy Radio Industries featured a multi-channel new console, designated as model SS5200. Circle 60 on Reader Service Card.



Nortronics displayed this tape head group; also a head cleaner liquid. Circle 64 on Reader Service Card.

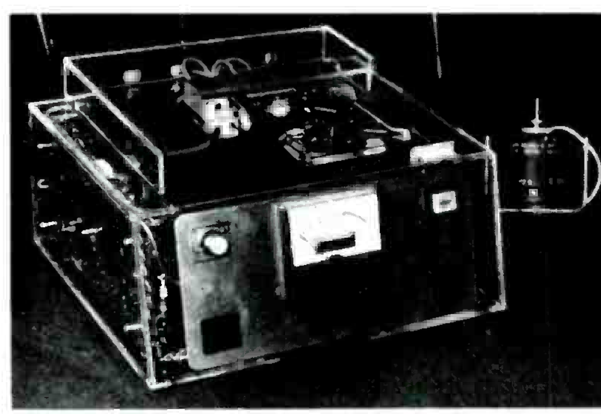


Schafer Electronics had a complete working broadcast automation system. Circle 65 on Reader Service Card.

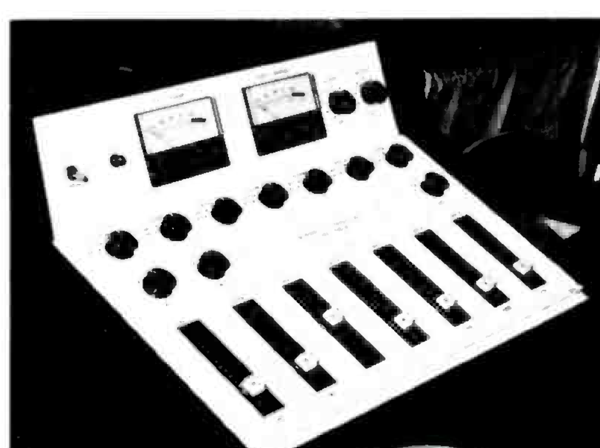


Shure Brothers has the M63 Audio Master with separate brass and treble control and turnover. Circle 66 on Reader Service Card.

Tape-Athon showed a new ¼-inch tape two-track recorder, as well as their logging machines. Circle 61 on Reader Service Card.



Tapecaster TCM have the 700RP record/play tape cartridge machine. Circle 67 on Reader Service Card.



Wilkinson Electronics set up this compact TSC-4 console with desk. Circle 68 on Reader Service Card.

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People, Places, Happenings

●**Calvin College** and **Bolt Beranek and Newman** have jointly sponsored a visual and performing arts conference to be held August 3-7. The conference will be held at the Calvin College Fine Arts Center in Grand Rapids, Michigan. Speakers include **Donald Irving,** director of the School of the Chicago Arts Institute; **Helge Westermann,** architect, Pietro Belluschi and Catalano and Westermann; **George Hutchinson,** partner, Perkins and Will Partnership; and **Peter Smith,** director, Hopkins Center, Dartmouth College. The Festival of the Arts will include art displays, a program of film and video tapes, and an exhibition of models and photographs of significant facilities for the arts. In addition, there will be performances of the Chicago and Detroit Symphony Orchestras. For further information contact **R. Lawrence Kirkegaard, 1740 Ogden Ave., Downers Grove, Illinois 60515, (312) 969-6150.**

●**General Radio,** in cooperation with **Bolt Beranek and Newman** will conduct a three-day seminar on practical techniques of product-noise reduction. The seminar will be held in Chicago on August 3-5 and will be lead by **William Ihde** of **S V Engineering,** acoustical consultant to General Radio, and **George Kamperman,** manager BBN-Chicago. Tuition for the entire three days is $200; registrants will be responsible for meals, lodging, and travel arrangements. General Radio will make motel reservations for those who desire this. For additional information contact **Tom Fricke, General Radio Company, 9440 West Foster Ave., Chicago, Illinois, (312)992-0800.**

●**CCA Electronics,** manufacturers of products for broadcast and recording studio systems has announced the acquisition of radio station **WABY,** in Albany, New York. The amount of cash was not disclosed and the acquisition is subject to FCC approval. In the announcement by **Bernard Wise,** CCA president, it was stated that this station and others to be sought in the future in AM, FM, and CATV, will assist CCA in the development and testing of its technical products in a way that can only be derived from "intimate ownership responsibility," according to Mr. Wise.

●**Edward M. Tink,** on the right, vice-president, engineering of the **Blackhawk** stations looks at a new **Fairchild** custom console designed for **KWWL** of Waterloo, Iowa. With him is **George Alexandrovich** v.-p. and chief engineer for Fairchild Sound Equipment Corp., (and db columnist). The new console will be primarily utilized as an operating console for KWWL-AM and in addition will provide certain control and monitoring capabilities for KWWL-FM's automated facilities. The photo was taken at the NAB convention, where the console was displayed. It was delivered immediately after the convention.

Turner

●An announcement from **Telex Communications Division** indicates that **Richard C. Turner** has been named to the broadcast and industrial product sales group of the division. He will report to the division's sales manager **Paul Bunker** who said that Turner's extensive technical background will make a valuable addition to the sales staff. He has been with Telex since 1967 in the product development engineering department. Prior to joining the company he was with **Honeywell** working on military and commercial guidance systems; before that he was a customer service engineer for **IBM Computor Systems** in Chicago.

Stone

●**Paul A. Stone** has joined **Elpa Marketing Industries, Inc.** as their high-fidelity division sales manager. He will be their primary contact with their representatives and dealers. Before Elpa, he was sales manager for the **R. T. Bozak Mfg. Co.** and earlier had been a product manager for **Utah-American** and a project engineer for the **Heath Company.**

●**Spectra-Sonics** has made an important expansion move. Offices are now established in Hollywood, California to provide complete system design and sales plus direct continuing engineering service to their clients. The manufacturing facility continues in Ogden, Utah. This move into their primary market area will offer consulting, design, and system engineering, component sales, and system sales/engineering. Complete system demonstration facilities are included. In the photo, **Albert V. Siniscal** on the left, who heads the Hollywood office shows the new facility to **Michael Lloyd,** vice-president and director of a & r for **MGM Records.** In addition to the console (a model 1020-8) and other Spectra-Sonics equipment, other manufacturer's support gear (recorders, turntables, etc.) are connected so that a feel for the equipment may be had. Address of the facility is Suite 1117, 6430 Sunset Blvd. in Hollywood. Demonstrations can be arranged by appointment.

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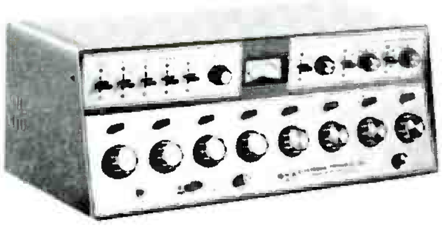

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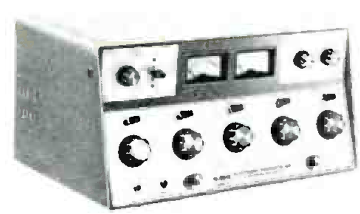

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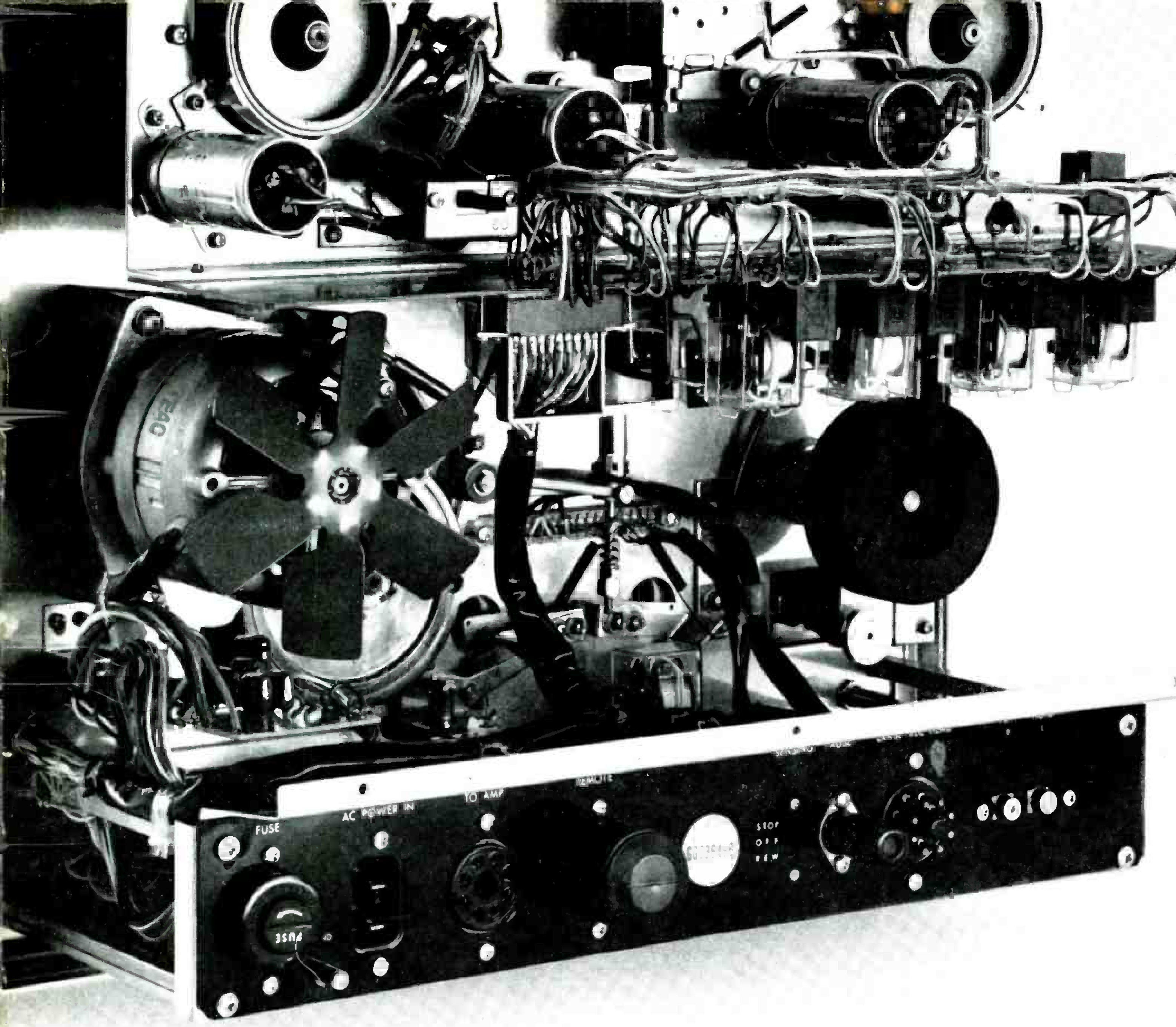

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