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NOVEMBER 1968 75c

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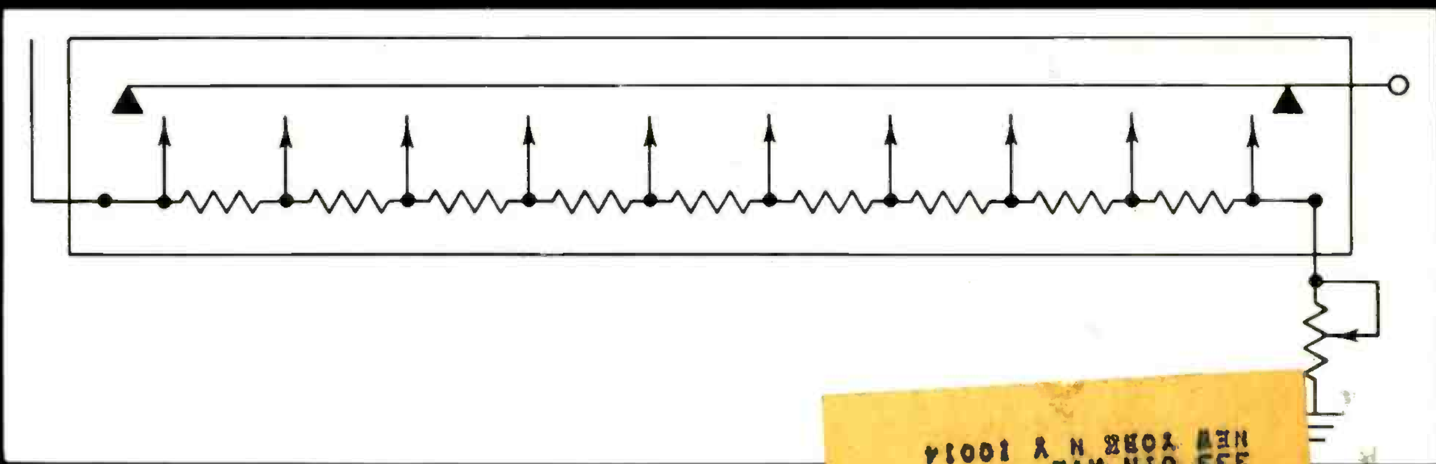

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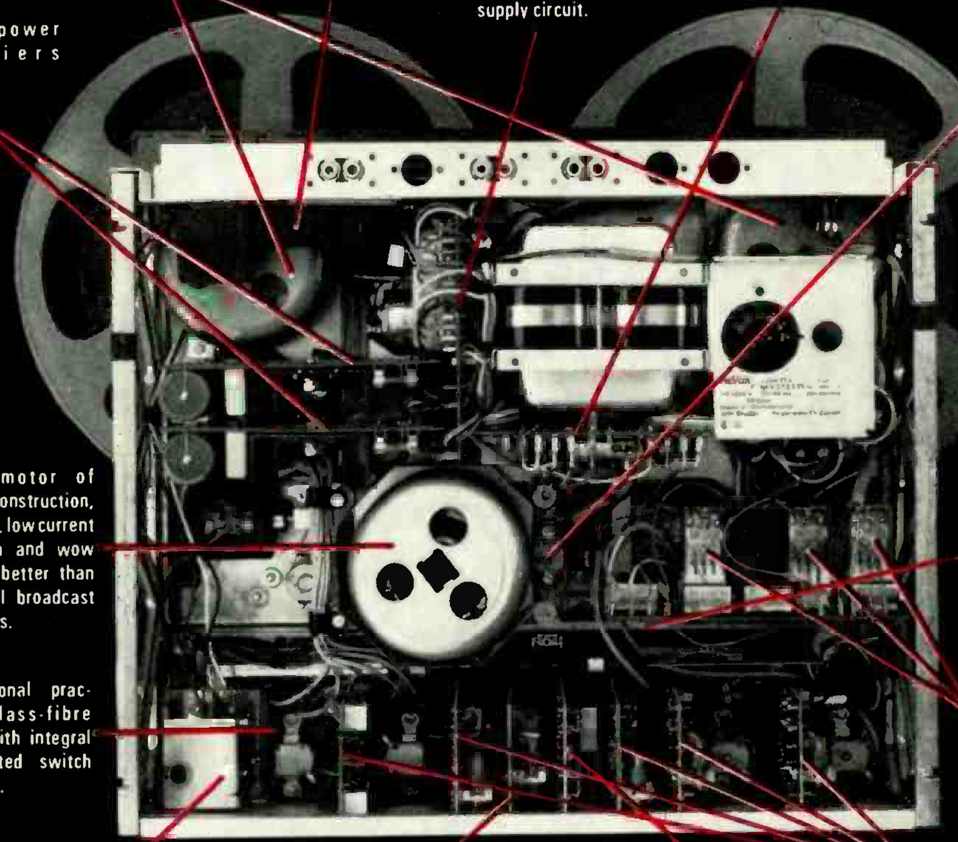

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Coming Next Month

Acousta-Voiced Monitoring Systems part one, by Don Davis of Altec will describe the applications of this special form of acoustic equalization to several auditioning and monitoring rooms of a recording studio. The purpose is to achieve both flat and identical sound in each of the several auditioning studios. According to the author the result is a calibrated listening system that provides the engineer with a true audio picture of what is happening in his studio.

Our roving camera poked its lens into most of the exhibitry at the Audio Engineering Society's New York Convention. The result will be a comprehensive picture gallery of exciting new equipment, many items shown for the first time.

And there will be our regular columnists, George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, and John A. McCulloch.

Coming next month in **db**, The Sound Engineering Magazine.

About the Cover

The artwork represents a linear controller for an electronic music synthesizer. It can be constructed and used much the same as a fret board of a guitar. Robert C. Ehle's article begins on page 22.

db

THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1968 • Volume 2, Number 10

Table of Contents

FEATURE ARTICLES

MONTHLY DEPARTMENTS

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Letters

The Editor:

One or more of your more informed readers may have called attention to the flux density figures in my article in your September issue (A New Ribbon Mirpophone). The actual figure is five thousand Gauss minimum, not three thousand as stated, and it is *not* the highest obtainable or obtained, since it is possible to get higher figures at the expense of extended frequency response.

David B. Hancock
New York, N.Y.

The Editor:

Although **High Fidelity** and **Stereo** were among the first consumer publications to use hertz to replace cycles-per-second, I can see that one may have mixed feelings about this change.

From the standpoint of consistency with other terms, such as decibels, watts, amperes, volts, ohms, and so on—which all were named after people—the term hertz makes sense.

But it can be questioned, from the standpoint of who did what and when. Perhaps Hertz is not the name to use. Has anyone thought of Helmholtz? Or was this name rejected on aesthetic grounds (can you imagine saying to someone, "This speaker goes down to 25 helmholtz," and having him answer, "Yes but mine goes up to 19,000 hockfleisch.")?

Or consider referring to a loudness contour curve as a *Fletcher-Munsonism* or saying that the preamp was "*RIAA-ed* fairly accurately," or "I'm going to my doctor to be *roentgenized*," and so on.

When you get to musical pitch, you might as well justify using *bachs* as hertz in place of cycles-per-second. Old Johann S. tempered the keyboard long before Heinrich tampered with wave propogation.

The real telling point in the anti-hertz argument is that we had a perfectly good, accurate description of the phenomenon in cycles-per-second. Precise, to the point, and no quibbling about exactly what was meant. We had no such expressions for what has become amperes, volts, etc. and therefore had to improvise. As I said, we in HF and Stereo adopted hertz — our of respect for the prevailing usage — but it wouldn't make us or our readers unhappy if the profession went back to cycles-per-second.

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IF THE CROWN CX822 IS JUST ANOTHER TAPE RECORDER

– the Stradivarius is just another violin

The Crown CX822 has been opening eyes in testing labs all over America. In early 1968, Audio magazine put it to the test, and published its findings in the April Equipment Report. Following are a few excerpts from that report:

"The Crown CX822 . . . is probably the finest tape recorder that has been reviewed in these pages. In addition to delivering phenomenal performance, it incorporates numerous features and refinements that place this machine in a class by itself."

"Tape threading is delightfully simple."
"Editing facilities are great."

"We found the tape motion command system to be as foolproof as Crown says it is, and could not beat the computer by design or by accident."

"The CX822 is a serviceman's dream, considering the unit's inherent complexity."

"Playing back first-generation transfers from original masters, the sound produced through the Crown CX822 was peerless. When recording and playing back from records and FM broadcasts, there was absolutely no aural difference between the original and the copy at 15 ips. The same held true at 7½ ips, though theory says, there should have been."

"Construction appears to be rugged enough to withstand parachute drops."

"The new Crown CX822 is capable of providing the most faithful reproduction of sound through the magnetic recording medium that we have observed to date. And it does it in as foolproof and as easy a way as we've seen."

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

PATCH BAYS

● A patch bay is seldom thought of as a switch, along with its patch cords. But the only difference between a regular switch and a patch bay is that the patch cord represents a flexible wiper while conventional switches may have a stationary type. A patch bay offers a degree of flexibility not available in any other type of switch that uses standard proven hardware. Whether you want to insert a piece of equipment into the chain or change the direction of signal flow by patching the signal through different paths — a patch bay is the answer.

Introducing a new element into an audio chain requires that we break the chain and insert the new equipment's inputs and outputs at the proper points. Special contacts in the patch bay accomplish this. The insertion of the patch cord into the connectors switches the circuit from feed-*through* to a feed *into* the patch cord. This, in turn, can be connected to any circuit designed to match the circuit broken by the patch cord.

Patch bays, just as other switch types, are available with contacts that vary from simple form A to sophisticated single or double form D systems. Older type of patch cords used double prongs; more recent patch cords for audio applications use a single prong consisting of a tip-ring and sleeve contact surface. The advantage of this tip-ring-sleeve patch cord is that polarity and phasing is maintained constant — so important to multi-channel installations.

Although much of the engineering fraternity still likes patching facilities, more and more consoles are being made without patch bays. This may, in part, be explained by the fact that the audio industry accepted patch bays as a means of providing access to any part of the circuitry for the ease of troubleshooting. A defective component could easily have the signal patched around it.

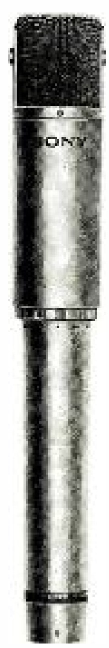
The state of audio engineering advances in equipment reliability have made patching for this purpose unnecessary — leaving the value of patch bays only for circuit selection. The introduction of transistorized, direct-coupled equipment (without transformers), has made the task of applying the patch-bay concept to the new circuitry difficult and expensive. In a transformerless amplifier, ground is the same wire as the low side of the power supply. So it is possible to produce lovely ground loops using patch bays without transformers. The alternative would be to have a transformer at every circuit point to be disconnected. In a multichannel console where a high degree of flexibility is required, patch bays with transformers result in a system so loaded down with expensive and heavy iron that it becomes more economical to install additional inputs and circuits — eliminating the need for patching.

Contemporary practice uses transformers only as a decoupling device for mic inputs or for line feeds for remote devices with different ground potential.

Nevertheless, a patching facility can be used in a modern system for reasons of economy. Not every radio station or recording studio can afford all the equipment they would like to have available for all occasions. Expensive equalizers, limiters, or compressors can not always be installed in every line or input. The obvious answer is a patching facility. This can further save money since it is often possible to *rent* highly specialized equipment that is only rarely used.

The design of a patching facility must take into account both levels and impedances. Maximum compatibility requires each input and output of the broken circuit to be able to accept any piece of equipment with unity gain and

Send us your name, and we'll add an impressive performance chart to this list of reasons for you to specify the Sony C55-FET microphone.



1. We have replaced the conventional, fragile vacuum tube with a field-effect transistor for rugged dependability and elimination of bulky external cables and power supply.

2. Completely self-contained, the C55-FET has an internal 9-volt battery. We think it's a big improvement over the whole power station you used to have to hook up with all the cords and wires!

3. Permits use of standard Cannon XLR-3-12C output connector for battery operation. Now a condenser mike can have the same connector as any dynamic or ribbon mike, providing complete compatibility among all your studio equipment.

4. For ease of operation in a permanent studio location, the C55-FET can be externally powered by an optional AC power supply (AC109).

5. By the use of a field-effect transistor, Sony eliminates overload problems commonly associated with tube-type mikes because of grid blocking.

6. The C55-FET's low current draw permits at least 800 hours of battery life. A pilot light indicates battery condition at the flick of a switch.

7. Movable capsule can be positioned vertically, horizontally, or anywhere between, making the C55-FET unusually convenient for hand-held operation.

8. Ring switch for battery check, flat or two low-cut modes.

These are just a few of the reasons that Sony professional microphones are so popular with studio engineers and sound experts. For more reasons and more information, please write to Harold Watson, Sony/Superscope, 8150 Vineland Avenue, Sun Valley, California 91352.

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a terminal impedance of 600 or 150 ohms. Since there still is much tube-type equipment around using transformer input and output decoupling, patch bays must provide a terminating impedance.

This task is complicated by the fact that most of the transistorized equipment is built for bridging rather than matching termination. As an example, the output of a push-pull transistor amplifier with complementary- or quasi-complementary-symmetry output stages. The output impedance of this amplifier is on the order of several ohms, but the design load is above 100 ohms. Equipment following the amplifier can not be sensitive to a specific source impedance. This classification includes passive equalizers, special filters, and constant-impedance faders, as well as other similar circuits. Placing a transformer at the output of an amplifier doesn't mean that the output impedance of the amplifier is 600 ohms. The recommended load may be 600 ohms but the source impedance may be as low as 10 ohms.

There have been many systems designed with patch bays that are transformerless. However, this puts an extra demand on the operating engineer to know the system thoroughly if he is to patch properly. Nevertheless, it offers many advantages that transformer-isolated patch bays lack. Frequency response is left unaltered and the full use of the low output impedance of transistor equipment is fully utilized allowing longer wires and multiple loads to the same source, without a change in level.

ELECTRONIC SWITCHING

Any discussion about switches inevitably leads to the question of modern electronic devices and elements as switches. Transistors, cadmium sulfide light-sensitive cells, silicon-controlled rectifiers, photodiodes, and the like, all fall into this category.

For purposes of this column we will limit ourselves to transistors and *ldr's* (light-dependent resistors — cadmium sulfide, and selenide cells). The purpose of this review is to spur you to an interest in these new methods of switching circuits without moving contacts. They are operable from a remote location, noise-free, and reliable.

As stated in previous discussions, any switch design must begin with a clear picture of the circuit for which the switch is intended. This includes associated voltages, impedances, and the desired distortion and noise characteristics. In addition, you must consider the speed of switching and the needed reliability factors. The final choice centers around an element or circuit with a minimum number of adverse characteristics consistent with a superior economic situation.

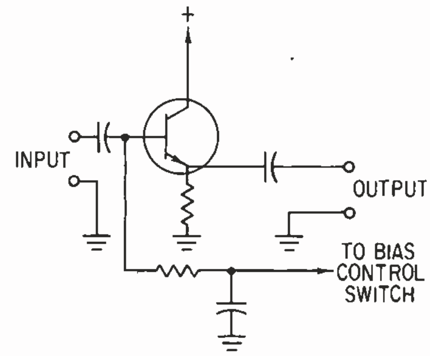

Figure 1. A transistor switcher.

Transistors make wonderful switches, but have you ever stopped to consider how much signal a single transistor can handle, and with what distortion, noise, and impedance. How much leakage is there through the transistor in an *off* state? What frequency-response limitation can we expect from the circuit?

Or have you ever tried to use a cadmium-sulfide cell as a switch only to find out that the release time is too long, you can't get enough attenuation and, as a remote gain control, you get distortion.

These are the parameters of the devices that must be coped with in designing circuits. It stands to reason that you must have a complete understanding of the phenomena taking place if you wish to find a proper circuit to circumvent the limitations of the device.

Transistors as switches in audio circuits are most commonly used as emitter followers. This gives a fairly high input impedance, low output impedance, fast switching response, and capacity to handle levels up to 15 dBm with supply voltages of 24V d.c. Distortion is low, noise is low, and response is as good as the capacitive coupling. Switching is accomplished by switching the base bias from proper bias for normal operation to cutoff. Multichannel switching can be operated from one bias supply to the transistors.

The disadvantages of transistor switches may outweigh the advantages. They are costly, the input and output of the switcher has to be d.c. decoupled, several resistors are required and you must have a means of mounting parts. A well-filtered power supply is necessary to prevent signal pollution with power-supply ripple. The supply voltage must be high enough so that line-level feeds through the transistor stage will not cause distortion of audio peaks. Also, as the bias of the base changes, the emitter potential changes from half the supply voltage to ground potential. See Figure 1.

If flat response with low impedance is desired, large coupling capacitors should be used. They are the ones that will take time to charge and discharge, slowing down the switching cycle and producing distortion during the transition from one state to *off* or reverse.

If high-impedance circuits (several thousand ohms or more) are used the problem is lessened. It is also possible to use other circuits for high-impedance switching (common emitter); this produces additional gain but with increased distortion and greater leakage as limiting factors. Phase reversal must also be considered.

LDR'S

Cadmium-sulfide and cadmium-selenide cells are gaining in popularity as switching elements. In total darkness, a cell has a resistance of several megohms. When illuminated with a light source of 100 or more footcandles (the approximate intensity of a pilot bulb at the distance of a fraction of an inch) the resistance of the cell drops to several ohms. This huge dark-to-light ratio permits the design of switching circuits capable of achieving signal attenuation comparable to relay or manual switches under certain conditions.

An improperly designed circuit will surely produce distortion, lag in action, and provide insufficient attenuation of the signal.

There are advantages to the use of ldr cells. They are relatively inexpensive. They can be operated either from an electrically actuated light source or by mechanically moving a light shield separating the source and the cell.

An ldr cell should only be considered for low-impedance circuits (or high-impedance circuits not carrying any audio). The cell changes its resistance fast as it is illuminated. However, as the light source is removed the cell exhibits slow changes in resistance due to photon inertia. It may take the cell several *milliseconds* to come on to a low resistance of several ohms, but several *seconds* or more to return the resistance of the cell to the megohm region.

Yet another factor affecting the reaction time of the circuit is the time of the light source. Normally this is longer than the cell's. The only light sources which could qualify for extra-fast switching might be neon bulbs (with a d.c. power supply to prevent a.c. modulation), and light-emitting diodes. Both of these devices have fairly low light intensity; this is not enough illumination for low-impedance circuits.

The property of the cell itself is such that it can easily be voltage-saturated when illumination is low — something that creates problems when turning off the light source when an audio signal is present in the circuit.

The indiscriminate use of a single cell in moderate- and high-impedance cir-

cuits is a dangerous practice, particularly when the cell is used as a coupling element between the source and load. One of the largest manufacturers of electronic equipment built a console using cadmium-sulfide cells in this way. They ran into trouble. Not only was the impedance of the circuit high enough to cause slow turn-off times but during the switching cycle high-level signals were distorted. The design was intended to overcome the pops and clicks of conventional switches coupled with an ability to be remotely controlled. Instead of improvement, they inherited distortion.

While this could not have been entirely prevented, it can be minimized to the point where it is no longer a practical problem. Let us examine the circuit. In FIGURE 2 the cell is being illuminated by a bulb, completing the path for the signal from the transformer secondary to the input of the amplifier. While the bulb is on, the transformer is terminated by the input impedance of the amplifier. The cell, with low resistance, provides a loss-free path for the audio in the circuit where the impedances are of several thousand ohms. If the impedance of the amplifier input is 50k ohms and the cell has a maximum resistance of 10 megohms, the maximum attenuation of the circuit will be 46 dB. But it may take the cell several seconds to reach maximum resistance, though it will achieve a few thousand ohms in several milliseconds. Thus the signal is attenuated a few dB's quickly, and then attenuation is slowed down to a crawl. At the same time the termination of the transformer is being lifted, boosting the signal applied across the cell.

In FIGURE 3 a few dB of gain obtained from the transformer have been sacrificed. Now the situation looks like this: the secondary impedance of the transformer is 600 ohms with a fixed resistor connected across it. 1200 ohms is a proper value. The cell is still connected as in the circuit shown in FIGURE 2, but the amplifier input is now terminated by another 1200-ohm resistor.

What happens when the cell is on? The resistance of the cell increases insertion loss by a fraction of a dB. But when the light is extinguished and in a few milliseconds the cell reaches several thousand ohms, the attenuation of the signal is already over 20 dB and going further quickly to a maximum of 80 dB. Not only does the transformer maintain some termination, but the amplifier input remains terminated for lower noise and less r.f. pickup. When the cell is *on*, two 1200-ohm resistors in parallel produce the proper 600-ohm termination for the transformer.

There's a lesson to be learned from this example. With practically no change in circuitry, using data from experimentation and basic common sense, one can benefit from these new elements by obtaining optimum performance with minimized side effects.

Next month: some new elements for changing the gain of audio circuits.

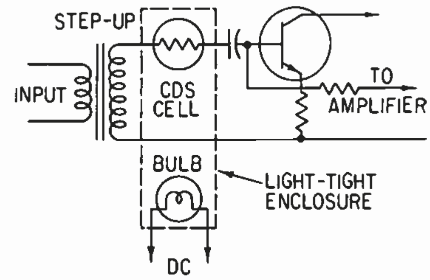

Figure 2. This illustrates the improper use of a cadmium-sulfide cell as a switch.

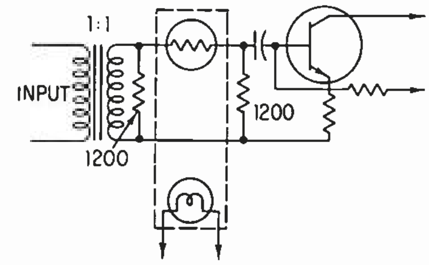

Figure 3. This modification of the circuit in Figure 2 significantly improves the characteristics of attenuation.

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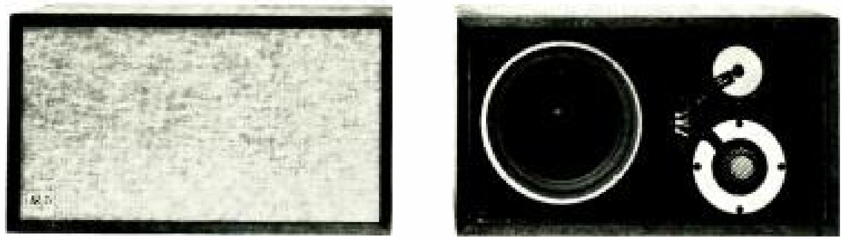

Theory and Practice

NORMAN H. CROWHURST

• If you try figuring out electronic crossovers and then building them, as I have done, you will stumble over problems in different orders. For the sake of clarity and brevity, I will group them based on the over-all picture.

In transistor circuits, you'll run into two troubles that make you re-evaluate your circuit. If you start by calculating a high- or low-frequency roll-off between stages, the circuit is different from tube circuits, in that the source resistance of the preceding collector is usually much higher than the load of the following stage, where in tube circuits, the grid load is higher than the plate source.

But you alter your calculations to take care of this, and it should work, shouldn't it? For the high-pass, possibly, but the low-pass is almost certain to give you trouble. If you shunt the combined circuit (collector impedance in parallel with base input resistance) with a capacitor to achieve the correct roll-off, you run into two troubles.

First is distortion: linear amplification from successive stages of transistors depends on the *current* transfer from one stage to the next. All the while coupling components hold the current linear, the amplification is linear, and over-all feedback can linearize it some more. But a shunt capacitor must work on *voltage* at that point. So a capacitor should never shunt a base input, either directly, or by a.c. coupling.

This means the base input must have a series resistance, to linearize current to the voltage at the point where the capacitor shunts (Figure 1). The circuit may prove even more fussy than that statement implies. When a collector is directly coupled to a following base, because the collector impedance is high, the voltage waveform may be highly distorted, even though current amplification is quite linear.

Inserting the resistance may reduce the distortion of the voltage waveform, but the reduction must be enough to make the voltage waveform (across which the capacitor is applied) essentially undistorted.

That's the first thing. The second applies to the achievement of the predicted frequency response, both before and after applying feedback. Unlike tubes, the input impedance of a transistor reflects the impedance of the load connected to the collector or emitter (according to whether the stage is grounded-emitter or grounded-collector, respectively).

In the grounded-emitter, reflection inverts the impedance. So a series capacitor at the collector reflects back as a shunt inductance, and a shunt capacitor reflects as a series inductance (Figure 2). The equivalent values will never cause peaking (which might prove convenient), they always augment the loss with the result that feedback can never sharpen up the response enough, to follow the predicted change.

This sharpening depends on the 90-degree phase shift coinciding with the 6 dB/octave slope point on the frequency scale. But when this reflection happens it never does. However much feedback is applied, the predicted sharpening never happens.

The remaining possibility for failure to achieve the desired result is not unique to transistor circuits. The resistor used to provide feedback is called the feedback resistor. But we forget that a resistor doesn't mind which way the signal goes.

As the crossover action cuts off response, a frequency is reached where the output is not bigger, but smaller

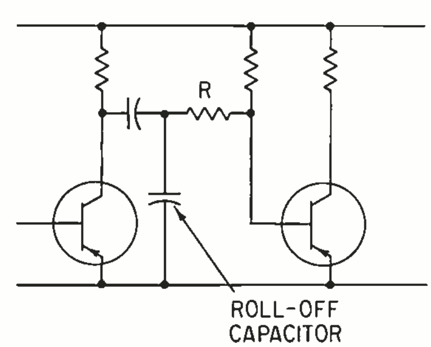

Figure 1. Achieving linear amplification with low distortion. The resistor R is there to prevent the rolloff capacitor from shunting non-linear base input resistance.

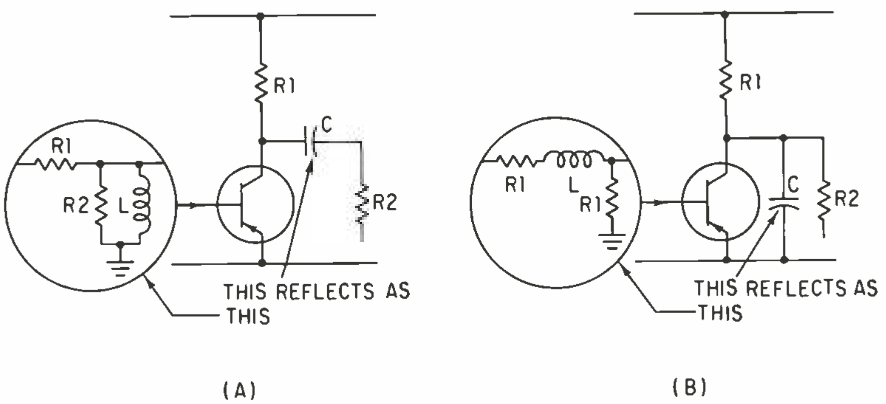

Figure 2. A series capacitor at the collector of a transistor reflects back as a shunt inductance (A), while a shunt capacitor (B) reflects as a series inductance.

than the input. When this happens, the feedback resistor can become a feed-*forward* resistor. providing leakage from input to output. If the circuit is designed so that forward gain is not very great—only enough to get good negative feedback action of the desired amount—then the point where feedback changes to feed-forward isn't very far down the cut-off slope (Figure 3).

Both feedback and feedforward depend on one other value beside the feedback resistor (Figure 4). Perhaps this aggravates the problem with transistors, as compared with tubes. If the source impedance from which the feedback is taken at the output is large, the feedback resistor may provide less attenuation to feedforward than it does to feedback.

The figures in Figure 4 show this (where both resistance values are equal). The impedance at both input and output end of the feedback resistor is 1k. The circuit is designed to use 12 dB feedback. midband, and net gain without feedback is 20 dB, so that feedback reduces it to 8 dB. The feedback factor must be (1 + AB) = 4 (for 12 dB). So AB = 3. As A = 10 (20 dB Gain), B must be 3/10.

With 1k at the input end the feedback resistor must be 7/3 x 1k = 2.33k. This is part of the output impedance, which combines to 1k. So the rest of the output impedance must be 1.75k, to make parallel combination with 2.33k come out at 1k for the output load.

Now, the mid-band gain is 8 dB, or output 2.5 *times* input. But when the crossover comes into action, the feed-forward will be from 2.33k into 1.75k, which attenuates by 1.75/4.05, or 7.4 dB. So the maximum change in level our crossover will give is from 8 dB gain to 7.4 dB loss, only 15.4 dB total, although the crossover aims at 12 dB/octave.

Having explored the unexpected theory failures, let's work on a design. Figure 5 shows a workable circuit to achieve 12 dB/octave. Using stages with appreciable resistance in their emitters does 3 useful things:

(1) It controls gain very nicely, so that performances is not affected by variation in transistor characteristics;

(2) It provides a linear base input impedance, reflected by the emitter resistor, through the gain of the transistor. This reflected emitter-circuit impedance also swamps any change reflected by the collector circuit.

(3) It linearizes individual stage gains, so less over-all feedback is needed for high linearity.

Assume the transistors used have a current gain of 100, with tolerances 70 to 140. First step is to provide terminations that will prevent precise external impedance values from affecting performance materially. At the output end, the two 1k resistors shunting the the 560-ohm emitter resistor, which is shunted by a much lower effective emitter source resistance, will provide a 500-ohm source for the output.

Assuming the external termination is 500 ohms, the 1 k in series with the 1k and 500 ohms in parallel provide 12 dB voltage attenuation and the emitter a.c. load is 394 ohms. With short-circuit termination (unlikely, but an extreme limit) it is 560 ohms in parallel with 1k. which makes 360 ohms. With open-circuit termination (again an extreme limit) it is 560 ohms in parallel with 2k, or 437 ohms.

We do a similar thing at the input, providing another 12 dB voltage attenuation, and similar reduction of effect of external termination. We need 6 dB over-all feedback to get the required sharpness, and if we plan on a resulting gain of 6 dB when we're through, that will give a margin for any losses. So the total gain needed is 12 + 12 + 6 + 6 = 36 dB.

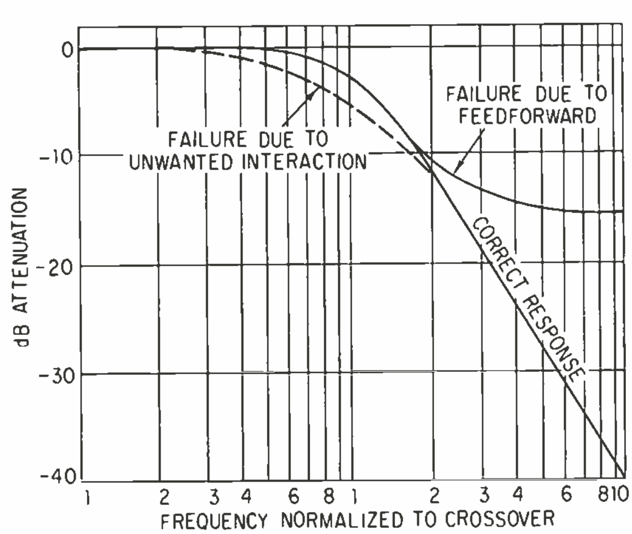

Figure 3. As the crossover action cuts off response, a frequency is reached where the output is smaller (rather than larger) than the input.

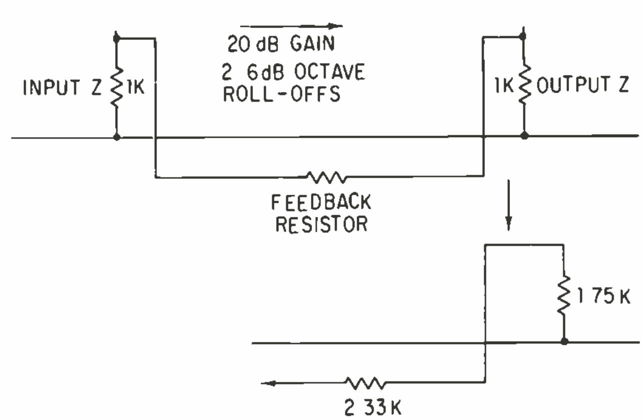

Figure 4. Both feedback and feedforward depend on source impedance as well as the feedback resistor.

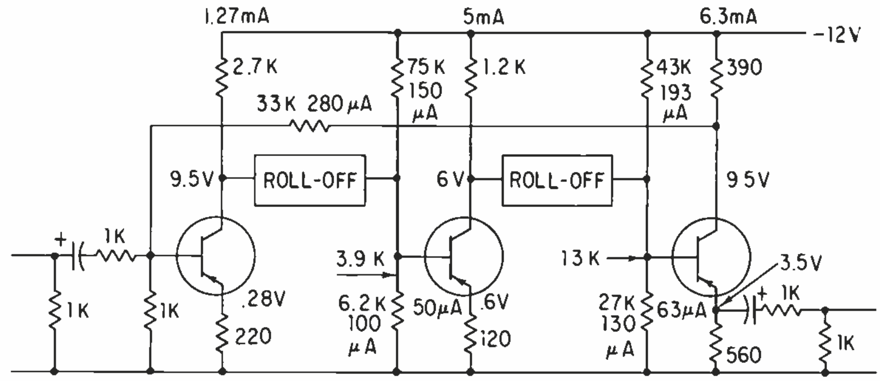

Figure 5. A workable circuit that will achieve 12 db/octave.

If we give each stage 18 dB (or 8:1) gain, and let the last stage provide 2 equal outputs, one (collector) for the feedback and the other (emitter) for the crossover output, this will further reduce feedforward effect, because of the high attenuation of signal from collector to emitter.

Now we work back, stage by stage. To make collector and emitter signal voltages of the third stage equal, average emitter load is 390 ohms, so use a 390 ohm collector resistor. Total d.c. load is 390 + 560 = 950 ohms. Assume a 12 V supply. 6 V across the transistor will yield maximum output swing, with the other 6 V across 950 ohms, yielding 6.3 mA. This sets emitter voltage at 3.5 and collector at 12 − 2.5 = 9.5 V.

Base current will average 6.3 mA divided by 100, or 63 microamps. Designing for twice this in the base-to-ground resistor will swamp transistor gain variations. 130 microamps at 3.5 volts requires a 27k resistor. The top resistor passes average 130 + 63 = 193 microamps, at 8.5 V, requiring 43k (nearest preferred value).

The base presents average impedance of 100 times 560 ohms, or 56k, and is paralleled for signal by 27k and 43k, making a combined impedance of 13k. Using a 1.2k collector resistor for the middle stage, and aiming for a stage gain of 8 (18 dB, half the 36 dB total required), the load is 1.2k in parallel with 13k, or 1.1k, requiring an emitter resistor of 1.1k divided by 8. If a 120-ohm is used, the gain will be 9.15, leaving a required gain of 64 divided by 9.15, or 7, for the first stage.

To get 6 V on the middle stage collector, this stage current must be 5mA, making emitter voltage 0.6 V. The base current will average 50 microamps. For the base-to-ground resistor to swamp gain variations, it should pass 100 microamps, which at 0.6 V requires 6.2k. For the top resistor, 11.4 V at 150 microamps requires 75k.

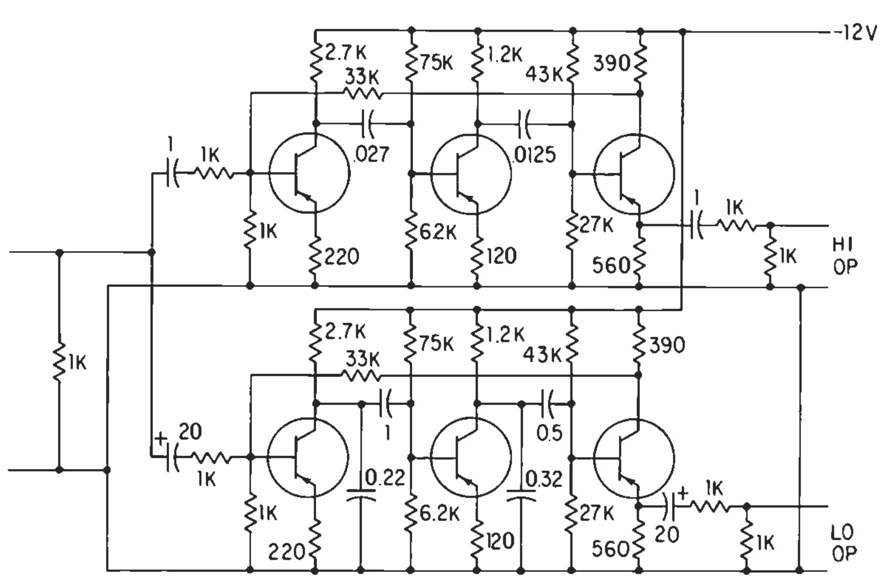

Figure 6. The complete crossover circuit that has been arrived at.

Reflected base impedance is 100 times 120 ohms, or 12k, and is paralleled for signal by 6.2k and 75k, which combines to 3.9k. Using a 2.7k collector resistor for the first stage, the load combines to 1.6k. Dividing this by 7, to get a voltage gain of 7, the emitter resistor should be 220 ohms (5 per cent).

Now the emitter reflects 22k into the base circuit, and the average base circuit impedance is 550 ohms, which parallels to about 537 ohms. The gain is 9.15 + 7 = 64, and for 6 dB feedback AB must be 1, so the feedback resistor needs to be (64 − 1) = 63 times 537 ohms, or 34k. A 33k, 5 per cent should be close enough.

With 9.5 V on the output collector, and a 34k d.c. path to ground, current is 280 microamps, and voltage at first stage base is 0.28 V. This emitter voltage, across 220 ohms, sets first stage current at 1.27 mA, and collector voltage at 12 − 3.5 = 9.5 V, which is satisfactory.

Now for the feedforward effect: the normal gain from input base to output collector, with feedback is 32 (half 64, due to 6 dB feedback). The ultimate feedforward attenuation is 33k terminated by 390 ohms, followed by the attenuation from collector to emitter. 33k and 390 ohms produces an attenuation of 1/86, and the collector to emitter attenuation will be at least 1/40 (meaning collector resistance is at least 20 times the combined collector-load value).

So the over-all level change, before leak forward starts to be noticeable, will be 32 × 86 × 40 = 110,000, or more than 100 dB, which should be enough! At 12 dB per octave, that will take more than 8 octaves.

For reactance values: First stage coupling, series reactance is 2.7k + 3.9k = 6.6k; parallel 1.6k. Second stage, series 1.2k + 13k = 14.2k; parallel 1.1k. Assume 630 Hz crossover, and coupling of low-pass also designed to roll off at 20 Hz. Frequency for high-pass design is 1.414 × 630 × 890 Hz, ω = 5,600. Frequency for low-pass is 0.707 + 630 = 445 Hz, ω = 2,800. For low cut-off, frequency is 1.414 × 20 = 28.28 Hz ω = 177.5.

The various capacitor values calculate to: High-pass: 0.027 mFd and 0.0125 mFd; Low-pass: 0.22 mFd and 0.32 mFd; Low roll-off: 0.85 mFd and 0.4 mFd. Probably a 1 mFd and a 0.5 mFd will serve that last function, making roll-off about 16 Hz, instead of 20 Hz. The complete circuit is shown at Figure 6. For the external coupling capacitors, the reactance should be low at the lowest frequency handled, for which 1 mFd is satisfactory for high-pass and 20 mFd for low-pass. The 1k across the combined inputs, with each input regarded as shunting the other, results in the average value (when terminated with 500 ohms) of 550 ohms.

Sound with Images

MARTIN DICKSTEIN

• Back a long time ago, or so it seems, discussion took place (along with experimentation) to develop a visual effect to complement voice or music being played at the same time. The effect of colored lights, changing in brightness and pattern, could be used to enhance the created mood, thus producing a fuller, more sensual experience. This began during the last few years of the 1950's and took place in Europe.

This device was different from the light organ first developed toward the end of the 1890's. In this later unit, the music is on one track and the light-control and triggering pulses are on others of a 4-track tape. A multi-channel drum programmer is rotated in steps by the trigger pulses and the lights flash on or off with their brightness controlled by the frequency pulses on the other track, driving motorized dimmers.

Each light unit, and there are as many of them as desired to create a wall-sized effect, consists primarily of a lamp with a mask/reflector, mirror, and rotating color filter. The original intent might have been to soothe ragged nerves or calm frightened or upset individuals. It is fairly obvious, however, what the effect could be if fiery music and rapidly moving vivid colors were to be used. Projection is intended to be from behind a specially made glass.

Although the original concept had application potential such as lobbies or reception rooms and show rooms or exhibit displays, the total use of the device was limited, perhaps due to its expense, or the complexity or the space requirements, or even the failure to visualize the effect or potential of such a technique. (A device similar to this unit, but much smaller, was developed for some commercial use with a short message flashed repeatedly between or mixed with varying color effects.)

With further technical development it soon became evident that music could be used to directly vary the light effects. Many will recall the application of this concept at the *General Electric Pavilion* at the '64 World's Fair in which a large "curtain" of lights was made to vary in color and brightness in direct relation to the audio heard by the visitors in the entry segment of the carousel. A similar but much smaller effect was also used in one segment of the *Tower of Light Pavilion*. In these instances, the total frequency spread of the music was divided into three or four bands with each of the segments controlling the brightness of a color (red, blue, green, yellow) in relation to the dynamics of that band of frequencies.

This unit also had limited application in fancy lobbies, show-rooms, large-company reception areas, and the offices of a few exclusive dental surgeons and psychiatrists.

Then, along came the discotheque, and the supreme application of changing lighting with the beat and tempo and dynamics of music became a demanded need. The wildest lighting imaginable is considered a minimum requirement.

Spotlights and over-all lighting are not to be used at all. The dance room is to be as dimly lit as possible allowing one person to see another vague shape at close proximity (although this is not the way they dance). This static, dim lighting is used only during the short pauses between the dance music.

During the playing, however, another world comes into being. The static lighting dims out completely and dynamic lighting takes over. The group on the platform or stage is illuminated by pulsating footlights of different colors keeping time and beat with the music and varying color by instrument frequency range. Under these conditions, the lighting can become rather garish, but it must be if it is to blend with the extremely high level of sound emanating from the many A-7's mounted around the room on the walls or in the ceilings and pushed by individual 40-watters. Levels of 110 dB's call for high-level lighting during peak playing/singing.

However, the ceiling and side walls are not illuminated in this manner except by spill of the light playing on the musicians. The front wall, either directly behind or above the playing group, is lit up by an overhead projector located somewhere on a second level. Instead of a static transparency, however, a plastic dish is used with a mixture of colored water, oil and a rhythmic strong arm. By gently maneuvering the dish in time with the music, the colored images projected on the wall change form and shape in sync with the pulsing lights. By super-imposing a smaller plastic dish with different fluids of other colors on top of the first dish, the combination of images on the wall really take on accented effects. Using several projectors with different types of "transparencies" enhance the imagery even more.

In addition to all of this, film projectors provide undulating colors and shapes on side walls and on the dancers themselves. The films have been specially made so that no definite image or object can be identified. The projectors use continuous film loops creating bright mixing and blending of shapeless colors, adding to the total effect.

Now, it seems, the concept of "seeing" the music is on its way into home-type application. Consoles including a fluctuating light unit will be on the market shortly (if not already available). Devices are also being sold which permit the user to connect his existing music system so that he may "see" the music. The techniques used vary from a rotating disc to a vibrating mirror or prism moved by the air in the speaker housing, or electronic control of different colored lights. Each provides its own type of audio/visual effect, and each will have its own price tag. Some units, looking like bookcase speakers, will undoubtedly take their place in the reception room, office or display area. Where do we go from here? Music classes in schools (Beethoven is the one that has as many reds and greens as the Monkees)? Maybe these devices and techniques can be used to help mental patients (or, create new ones).

Editorial

On October 23rd, Erich Leinsdorf, music director of the Boston Symphony Orchestra was awarded honorary membership in the Audio Engineering Society. The occasion was the 35th Convention Banquet. In an address following the award, Maestro Leinsdorf said, "In this era of recordings, f.m. broadcasts, and electronic music, musicians and engineers must work in ever closer relationships. You audio people and we musicians must always watch lest sound become noise. This danger is more iminent than ever before."

The question of noise pollution has become one of the major problems of our times, particularly in urban areas. It must be checked.

Noise pollution takes many forms. While the street-breaking jack hammer is an obvious affront, the sum total of the many sounds around us is just as devastating as the pounded decibels of the hammer.

Increasing medical reports link the constant bombardment of noise with adverse effects on our nervous system, our hearing, our very well-being. It always has been assumed, for example, that progressive high-frequency hearing loss with increased age is a normal consequence of aging. Today we know of societies in the world who live in comparatively silent environments in which they do not suffer these losses.

Erich Leinsdorf made an impassioned plea for a world where it is possible to escape into silence. Most of us are so accusomed to the noise pollution around us — accept it so passively — that we no longer are aware of it. Noise-battered urban residents transported to deep rural areas are almost universally disturbed by the lack of pollution.

In our industry noise is an outright danger. Those hours spent in control rooms with the monitors up much higher than they need to be take their toll. (We will shortly present an article that documents the *permanent* hearing losses suffered by audio men.)

But this, at least, is private noise. What about the public places that are flooded with "background" music systems.

Is the public mind so empty that we must fill the vacuum with layers of sound? There is no reason for every lobby, every barber shop, every office, to be infested with sound.

Every one has a *right* to silence.

Perhaps we cannot blame the increasing ills of the world on the increasing noise pollution. But if the doctors are correct there is a direct relationship.

Most urban areas today have local noise-pollution control organizations. Sound professionals must be in the forefront of these groups, offering their expertise to control rampant noise. Where noises cannot be eliminated, acousticians and sound men can provide acoustic perfumes to make them more psychologically palatable.

L.Z.

European Equipment and U.S. Audio

STEPHEN F. TEMMER

European level, impedance, and metering standards are at considerable variance with U. S. usage. The author explains these differences and offers remedies for the operation of foreign equipment in conjunction with domestic components.

THIS IS NOT THE FIRST TIME THAT I BEEN BEEN ASKED about the differences between the European and American concepts of circuit design with special emphasis on impedances and levels. Each time before I have begged off for I saw a grave danger in being misunderstood as offering a plea for change to the European system, or a criticism of the U. S. system of handling levels and impedances. Just as it is foolish to advocate changing the languages of Europe to English, so would it be of no advantage to try to change the U. S. system of audio design. Each has its deeply rooted engineering heritage; each has its good and proper reasons.

Just as we all speak fluent English and perhaps are not too familiar with the rules of grammar that dictate the way we speak, so is it in technical matters which are a matter of habit. I am sure that many engineers in the audio field follow engineering habits without questioning each time the underlying theoretical and practical reasons behind their every move. It would make the everyday work process much too cumbersome. As an over-all concept I would say that our audio engineering language is mostly *use oriented;* the European mostly *theory oriented;*

Of paramount importance, however, to the American audio engineering community is the interchangeability of equipment between the two systems. This applies equally to American exported equipment to Europe and to European imported equipment into this country. The past years have brought increasing trade in both directions; the future will bring even more.

IMPEDANCES AND LEVELS

Fundamentally, our system layout technology concerns itself with matching loads to sources, almost entirely with 600-ohm circuits. Specific exceptions are microphone inputs and loudspeaker outputs. In the former we use anything from 30 to 250 ohms and work with unterminated inputs; in the latter we speak of voice-coil impedances on the order of 4-16 ohms. In both cases we do little thinking, for we invariably work with clearly labeled devices which require only that they be connected as indicated in the instructions.

When it comes to the in-between connection points we hold fast to a system using 600 ohm inputs and outputs of equipment producing complete or almost complete power transfer from one device to the other. For example, a booster amplifier indicates that it has an input and output impedance of 600 ohms. If you actually measured these impedances you would find that these figures are only *nominal.* What the manufacturer actually meant to say in his data is: "To be connected between a preceding source impedance and a following load impedance of 600 ohms."

There you have the use orientation. You are told where to connect it and not what its actual impedances are. Should you want to connect two such inputs to one preceding output you will have to take steps to assure that the preceding output is not injured by too low a termination. This would result if you simply connected the two loads in parallel with a resulting impedance of approximately 300 Ohms. You therefore use a *splitting pad* which costs you some gain but which restores the required 600-ohm happiness all around. Should you wish to produce a bus across which you could then bridge any number of inputs, you will have to terminate the amplifier feeding such a bus with a 600-ohm resistor. Then you will make sure that all of your bridges are of a sufficiently high impedance so that when all of them are connected at once, the effective impedance will not tend to significantly reduce the 600-ohm termination of the bus. Lest we make the mistake of calling any high-impedance input a bridge, let me make sure you understand that this term is reserved for Hi-Z inputs which are connected across 600-ohm terminated busses only!

Stephen F. Temmer is president of Gotham Audio Corporation, New York City.

Ampex AG-440. For the recording engineer who wants to fly.

If you're moving out, taking off with the hot young groups who ride the charts, you deserve the multichannel recorder that will let you be your most creative. The one with the greatest versatility. The one with the fewest technical hangups. The one that will pay for itself fastest.

There's only one like that: the incomparable Ampex AG-440. Read why recording engineers who are helping to create today's big sound prefer to work on an Ampex. Then return to this thought: An AG-440 can be working for you in a matter of weeks; you can use our deferred payment or lease plans, and pay for it out of current earnings.

Now. Ready to fly? Call your local Ampex distributor. If his number isn't on your chart, call our world headquarters, (415) 367-4400 (collect).

The MOJO Courtesy Reprise Records Div. Warner Bros./7 Arts

ADD TRACKS ANYTIME, EVEN SPLIT YOUR SESSIONS

The Ampex-pioneered Sel-Sync® System made multi-channel mastering possible. It gives you the freedom to develop the right sound even after the talent has gone home, to add new dimensions with layers of experimental sound. And if only half the group can make today's session, let the other half do their thing later. It's easy. AND REMOTE SEL-SYNC SYSTEM makes your AG-440 even more versatile. It was previewed at the Spring AES Show. Plan for it on your new board.

LESS DOWN TIME, MORE ASSIGNMENTS, BETTER INCOME

The AG-440 series is the result of ten years of multi-track experience. We know that easy maintenance is a must . . . that's why it has up-front plug-in electronics, tiltable top plate, and the simple straightforward transport. And because you can switch easily from ¼" to ½" tape, you can assign a variety of jobs to this machine, make more money for the studio.

PERMANENT ALIGNMENT SAVES TIME, SAVES MASTERS

If you've ever wrestled with fuzzy sound caused by misalignment, you know how important our heavy die-cast frame is to the AG-440. It provides an inflexible anchorage for tape guides and heads so that alignment and tracking will not present problems to hamper your creativity. And the stable, drift-free electronics keep your signal clear and true.

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The AG-440's extremely low inherent wow and flutter give you full freedom to create the biggest or most delicate sound. Now, in addition, you can get rid of annoying midrange scrape flutter, which has plagued recording engineers in the past. One built-in jewel bearing idler cuts scrape flutter by as much as 75%. A second (optional) virtually eliminates it.

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Much of the excitement of today's young sound comes from driving the instruments off the top of the scale. To capture these wild experimental sounds, you need the widest possible dynamic range . . . and the AG-440's distortion free swing gets the most from the tape. Coupled with maximum S/N. This means you can mix down and dub through many generations without excessive degradation. And, your multi-channel AG-440 records and plays back an honest 30 to 18,000 Hz (±2dB @ 15 ips).

LET YOUR AG-440 PAY FOR ITSELF WHILE YOU USE IT

Ampex offers a unique choice of lease or extended pay plans. You could, for instance, get a one-channel AG-440 for as little as $50 a month. Then, because of its modular design you could build it to 2, 3, or 4 channels. Or you can get a complete AG-440-4 or AG-440-8 *now* and pay for it out of current earnings. It's the only way to fly!

Write for your copy of the AG-440 "Flight Plan." It contains full description and specifications.
Professional Audio Products Division, 401 Broadway
Redwood City, Calif. 94063

AMPEX

(*from page 17*)

MICROPHONES IN THE U.S. SYSTEM

When specifying impedances in conjunction with microphones we follow the same rule; i.e., a preamp which specifies a 150-ohm input winding must be fed *from* a microphone whose actual source impedance is 150 ohms. However, the actual input impedance is highly frequency dependent, over most of the frequency range it is several times the 150-ohm nominal value. As a consequence the resulting over-all response of the combination of linear source and nonlinear load will remain linear as long as the load stays high enough to not load down the source. At the point where the load equals the source impedance, the response will drop off 6 dB. The reason for not loading such sources at all times is that it is desirable to keep such low levels from dropping even further, thus adversely affecting the signal-to-noise ratio.

A very low source impedance and relatively high load impedance in microphone circuits has advantages. The low source impedance effectively shorts static interference, while the high load impedance produces an over-all high loop resistance (which in turn minimizes magnetic interference). The disadvantage is that with long audio lines the low source acts as a capacitative load. As a result, telephone lines are handled in Germany as they are here, with a well-matched circuit.

ARE WE ALWAYS FAITHFUL TO OUR SYSTEM?

Take an *Ampex* model 351 recorder, for example. It purports to have an output impedance of 600 ohms and indeed, features a terminating switch permitting a 600-ohm resistor to be connected across the output. Now, make this test: Disconnect the Ampex output, play a standard alignment tape 1000-Hz tone and adjust to read zero on the vu meter. Now close the output terminating switch. How much does the level drop? You'll find it dropping about 4 dB. If the source were truly 600 ohms, it would have to drop in half or *6 dB*. The fact is that most such source impedances (except for test equipment such as transmission sets, which must be accurate) are somewhat *lower* than the indicated amount, and inputs will tolerate such a range of impedances.

THE EUROPEAN SYSTEM OF IMPEDANCES AND LEVELS

The way all German and most other European equipment is operated is very much the same as the way we operate our a.c. power system. Nobody normally wonders what the source impedance of the electric wall outlet is, nor does anyone normally worry about the load impedance of the appliance to be connected (with the limiting proviso that it not be so low as to blow a fuse). The assumption made by everyone without thinking is that the source impedance of the outlet is significantly lower than the intended load; so much so that plugging a lamp into the outlet will not diminish the existing voltage. In other words, terminating the outlet with a reasonable resistance, one that will not blow the fuse, will not change the voltage appearing at the outlet.

The very same system is in use in European audio. The source impedance is always significantly lower than the input impedance of any subsequent device. Ask a German audio man about the input impedance of his mic preamps and he'll just shrug his shoulder. Press him for an answer and he'll probably say something like this: "I don't know exactly but it's something greater than 1000 ohms." Ask him about the output impedance of his mic preamp and he'll tell you that it's no more than 40 ohms. Ask him to feed one of his mic outputs to three console inputs simultaneously and he won't reach for a splitting network, he'll just help himself to a multi strip and go to it. The same way that you reach for a cube tap when you want to plug three lamps at once into the same wall outlet. No splitting pad needed there. He has never heard of any of the following expressions: *bridging*, *terminating resistor*, *splitting pad*, *terminated bus*, etc. He just doesn't think in those terms.

Now you'll perhaps ask why this system isn't in use in this country. The reason is that it is expensive in terms of both dB and in terms of the kind of transformers needed to accomplish this. The input specification on any device would have to allow it to be connected to any source between zero and 200 ohms, while the output must be able to operate into any load from 200 ohms to infinity (i.e., unterminated).

The basic component which makes the voltage system possible is the transformer. While we consider the price of an input transformer to be in the fifteen dollar neighborhood, German transformers for such use may run as much as four times that figure. The result is a transformer whose stray resonance lies well above the audio range, sometimes as high as 90 kHz, resulting in a transformer usable over a much wider range of impedances.

Loudspeaker distribution systems in Germany use a 100-volt system much like our 70-volt speaker line system.

THE DETERMINATION OF LEVELS IN EUROPE

The audio level in a European system is simply expressed as the dB equivalent of the voltage across the particular circuit point, regardless of impedance. Measurement is made using a high impedance vtvm. It is clear that shunting the input of such a vtvm with a 600 ohm resistor would not significantly change the reading, since we are looking at a source impedance of 40 ohms or less. Levels are expressed in dB as is the gain or loss of a device. This gain or loss refers simply to the *voltage* ratio of input to output. A 40 dB amplifier is one which will deliver 100 times as much voltage at the output as you fed to the input. Zero level is defined as 775 mV which happens to be 1 mW when related to 600 ohms.

Let's look at an example of a *Neumann* transistor preamp and discuss the relative American and European level and impedance relationships. If you wish to lose some level you don't have to get an expensive constant-impedance loss pad; you simply need a voltage divider to assure that the previous device sees more than 200 ohms (actually as high a load as you can manage) while the next device sees 40 ohms or less. You never need a table for that, but you do need a very easily obtained slide rule which converts voltage ratios into dB and *vice versa*. Such slide rules are available everywhere in Europe. The same slide rules also give you the absolute dB levels for any given voltage.

LEVEL INDICATORS

Most European broadcasting and recording systems use what is called a ***peak level indicator*** rather than a vu meter.

This device is usually in the form of a light-beam meter connected to a logarithmic amplifier whose input is high impedance. Integration time of such a device is 10 ms for the amplifier and about 60 ms for the meter movement. The result is called *quasi peak indication*. The universally accepted output line level is +6 dB (equivalent to 1.55 Volts); i.e., a voltage of 1.55 Volts across the line produces a zero reading on the peak indicating meter. Program modulation ridden to that zero mark, however, can be expected to spill over with its peaks by no more than 3-5 dB, while a vu meter aligned to read zero for the same +6 dB line level if used for gain riding will allow (due to its at least five times slower action) peaks to spill over by as much as 14 dB. For this reason we usually expect our line amplifier outputs to be capable of peak levels on the order of +18 to +22 dB(m) while European equipment, unless it is especially made to meet U. S. specs, will only permit about +10 to +12 dB levels at the same distortion rating. You must be cognizant of this fact whenever you interpret output levels given in European technical specifications. Maximum output capabilities are given only by manufacturers who realize the problems of the vu-meter markets. As an example, the EMT140 reverb unit is capable of an output level of only +10 dBm. It is specified as a "+6 dB output." This level would be insufficient to allow monitoring with a standard vu meter, but then again, this is never done in the reverb return. On the other hand, the EQ-1000 universal equalizer and the *Neumann* transistor console components and *Studer* tape recorder are all made to deliver +22 dBm maximum output levels, perfect for U.S. use. Output source impedance of these devices are always less than 40 ohms but they are perfectly delighted with 600-ohm or higher terminations; even high-impedance loads without termination necessity.

The inputs of such devices are usually quite a bit higher than 1000 ohms; some as high as 5000 ohms. Connect these right across your U. S. equipment outputs but remember to switch on the terminating switches or use 560-ohm resistors to satisfy your equipment's need to see a 600-ohm load.

CONDENSER MICROPHONE CONNECTION

European condenser microphones operate just as any other European outputs, but U. S. microphone inputs are unterminated and *must* be fed from their nominal impedance. The microphones are published to be switchable in output source impedance to either 50 or 200 ohms. These are the actual measurable source impedances. Just like any other European sources, however, their source impedance may *not* be matched, but must be terminated with no less than five times their source impedance.

Let's see what happens if we set the microphone for 50 ohms. Plugging this into a 250-ohm input will be satisfactory with the microphone but the preamp input will be underterminated and will produce a high-frequency rise. This was the reason so many of the early *Neumann-Telefunken* microphones were criticized for being peaky. If we set the microphone for 200 ohms, it will be loaded too much by the preamp even though the preamp input will be satisfied as to its source requirement.

The answer is that you must resistively raise the 50-ohm source impedance of the microphone by simply installing a resistor in each of its output leads. We use 56 ohms on each side, making a total source of 162 ohms. This will just come within range of the 150- and the 250-ohm preamp inputs. Some of our microphones have so high a sensitivity that a voltage divider must be used to reduce level. by some 12 dB.

What does the preamp input see? It sees the 200-ohm resistor paralleled by two times 270 and 50 ohms (microphone source) in series. Effective total source is 149 Ohms. Fine for the preamp. The microphone output sees 740 ohms with 200 of those paralleled by some undefined medium impedance of the preamp input. Plenty high to make the mic happy.

INTEGRATING U. S. AND FOREIGN EQUIPMENT

As far as impedances are concerned (microphones and speakers excluded), worry about the U. S. equipment only. 600 ohms is good in every case with the foreign equipment. Terminate circuits *only* if the U. S. equipment needs it. Never load foreign equipment at all if you can help it. Make sure you do not exceed output-level capabilities of European equipment or for that matter, input-voltage capability either. Use a voltage divider if necessary to drop voltage in front of the input. U. S. outputs usually deliver more level than is needed; European inputs are more sensitive than you will need.

ATTENUATORS

In this country, attenuators are traditionally labeled *in* and *out*, referring to those terminals across which you are expected to connect inputs and outputs. European attenuators leave that to the situation at hand. T attenuators are symmetrical as to input and output so that you may use them forwards or backwards. Ladders have a constant impedance side and a non-constant impedance side. In U. S. terms, the non-constant side, the side which reduces in impedance to some 300 ohms, is marked *in* (European: *S* for *slider*) and the constant side is marked *out* (European: *K* for *chain*). This is the way the attenuator is normally connected in a mixer.

There are, however, applications under which you may wish to reverse this order. As an example: a tape-machine line output feeding into a ladder attenuator with a vu meter across the tape-machine output. You use the attenuator input as marked. The result is an alignment tone reading zero on the vu meter with the fader open that will drop some 4 dB when the fader is closed. That's a rather difficult situation in practice, for you may wish to check levels of program material with fader either open or closed. For the solution reverse the attenuator so that constant impedance *out* connection faces the tape recorder.

CONCLUSION

Both systems work; both systems have their relative merits. There is no doubt that the advent of transistor technology will bring with it ever lower source impedances and higher input impedances so that eventually we too will speak in voltage rather than power terms. We will probably never drop the m from dBm in our language and resistor manufacturers will go on getting rich selling 560-ohm resistors long after their need has disappeared. But then, tradition persists in every aspect of life; even in engineering.

An Electronic-Music Synthesizer

ROBERT C. EHLE

The design of a synthesizer for the electronic-music studio is discussed. Along with design parameters, the author covers the musical and electronic requirements of a synthesizer.

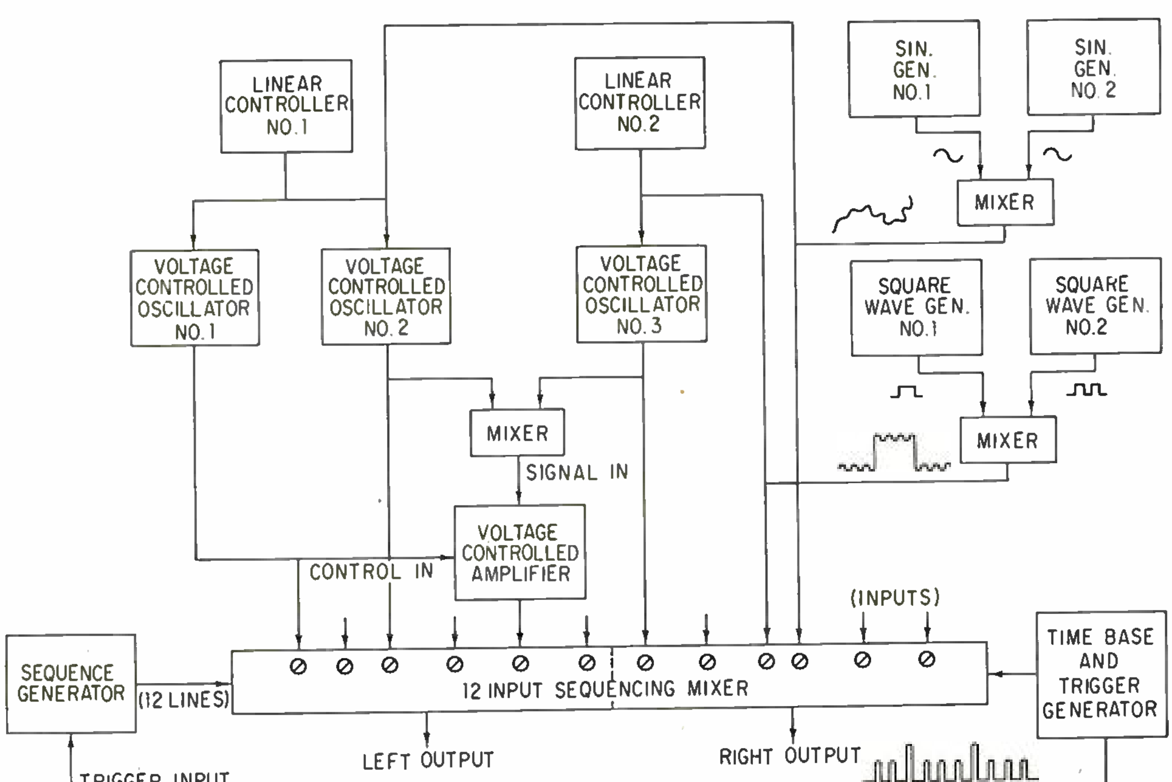

Figure 1. A block diagram of a synthesizer.

THE USE OF ELECTRONIC EQUIPMENT for the purpose of synthesizing sound and composing music has gained considerable popularity in the twenty or so years since serious efforts began. Of course, for many of these years serious work was done almost entirely by faculty members of universities. These efforts were closely allied with avant-garde music, suffering the fate of that type of music. More recently, commercial possibilities have been discovered; thus today, much activity is being done by commercial-sound and music studios in applying electronic music techniques to movies, television, and popular music.

For many years, the only electronic-music equipment was commercial test gear: signal generators, and so forth. More recently, with the development of the *synthesizer*, we have seen efforts to put all the necessary components into one package and to determine that they are all interchangeable or compatible. Thus, this assemblage of components becomes an instrument in itself and is commonly referred to as a synthesizer.[1]

The ideal synthesizer is a complete electronic-music instrument in itself (less amplifiers, speakers and [perhaps] tape recorders). So the important question is what constitutes a complete and functional synthesizer for music.

The largest manufacturer of commercial synthesizers today, the R. A. Moog Co. builds many different models. This indicates that there is no universal agreement as to what must be included (let alone, how it should be built). On the other hand, all synthesizers contain certain basic components: oscillators, modulators, filters, mixers, reverberators, and control devices. This article describes one possible arrangement of components and deals with a design actually built by the author.

FIGURE 1 is a block diagram of the synthesizer built by the author. The blocks represent items mounted in a common case. However, since most of the wiring is done on a patch panel, the interconnections do not represent the only possibility but simply one found convenient and, actually, frequently varied for different functions.

The synthesizer contains seven audio oscillators, more than most commercially-available synthesizers. This is due to a preference for rather complex sound patterns on the part of the author and is strictly an individual preference. The various oscillators are of three different types: voltage-controlled multivibrators, voltage-controlled sine generators, and conventional multivibrators. The multivibrators generate variable-width pulses and sawtooth waves. The conventional multivibrator generates a clean pulse on the collectors of the transistors and a sawtooth at the bases of the transistors. A switch allows the selection of waveforms.

Selection of circuitry is not particularly unusual. The voltage-controlled method has been devised by Moog[2], his method of generating currents proportional to control voltages has been employed. Essentially it involves a differential operational amplifier with a negative feedback loop from one collector output to the input. The other input receives the control voltage while the unused output provides a control current. The current may be used to operate many different circuits including ramp generators (such as those in relaxation

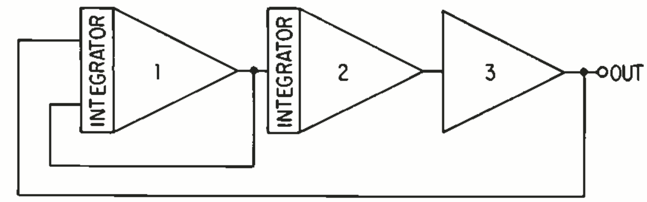

Figure 2. An analog computer, sine function generator.

oscillators or other operational amplifiers). In fact, the basic transistor amplifier is controlled by its base current, thus it may be directly controlled by the current source.

The balanced modulator circuit is used for a voltage-controlled amplifier and a special mixer is provided for sequencing functions. Completing the synthesizer, two linear controllers are provided for direct control of voltage-controlled devices.

Accompanying circuit diagrams are FIGURE 2 — a basic sine-wave oscillator as used in analog computers and FIGURE 3 — the method employed for converting it into a voltage-controlled oscillator. This method replaces fixed resistors in the integrators with light-sensitive resistors (cds cells or photo diodes). The lamps illustrated are driven by the two phases from the emitters of an operational amplifier. The light emitted by a tungsten filament lamp is essentially a function of the voltage across it.

FIGURE 5 is the circuit of the voltage-controlled amplifier and FIGURE 6 is a block diagram of the sequencing mixer. These two units are essentially original in design with the present unit. FIGURE 4 is a block diagram of a voltage-controlled multivibrator.

Many different circuits could be used to implement the block diagrams given in this article. What is required are

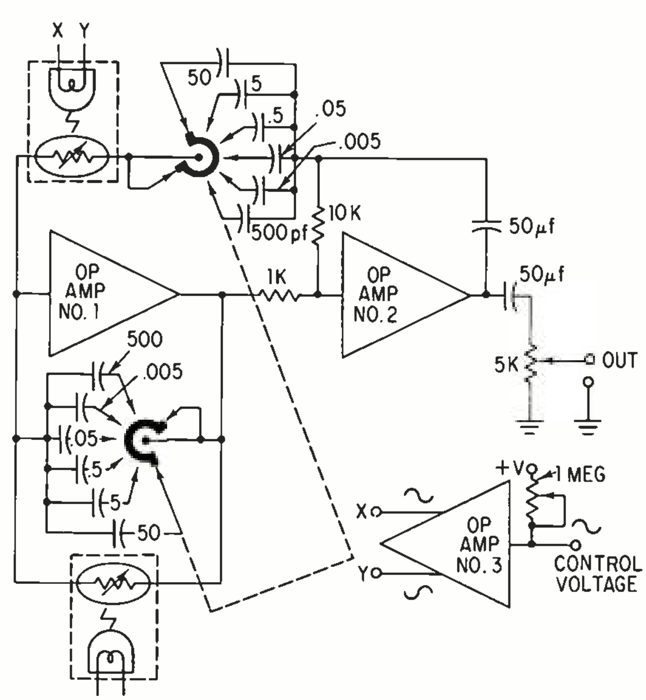

Figure 3. This partial schematic shows the implementation of a voltage-controlled oscillator.

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circuits capable of generating various functions in the audio and sub-audio range. Any circuits which fulfill this requirement may be used. A recent method gaining increasing popularity is function generation with the digital computer. The block diagrams still remain, but they describe the function only. The computer actually computes discrete points on the curve of the function which are then converted to the actual analog function with a digital-to-analog converter.

CONTROL

At least as important as the means for generating the various functions is the provision for *controlling* the various function generators. Analog function generators in a synthesizer, as in analog computers, employ patch panels, potentiometers, and switches for basic control. Digital function generation is controlled by computer programming. Analog function generators are limited by the functions provided by the manufacturer while digital generators are only limited by the programmer's ability (assuming a computer of sufficient size). Still, analog synthesizers maintain many links with traditional musical instruments, particularly in the control devices used. FIGURE 7 is the schematic of the author's linear-controller unit for use with voltage-controlled devices. This device is similar to a guitar fretboard in that frequency may be selected by pressure at selected points on the metallic ribbon. It has a fixed selection of frequencies available through fixed resistors rather than a continuous band of resistance wire as is often used.

The most pronounced link between synthesizers and conventional acoustical instruments is the frequent use of a keyboard in both. The keyboard is still a very popular control device in analog synthesizers. It can be built to work in the same fashion as the linear controller illustrated in FIGURE 7 simply by installing switches on the keys and selecting the points in the resistive voltage divider with them.

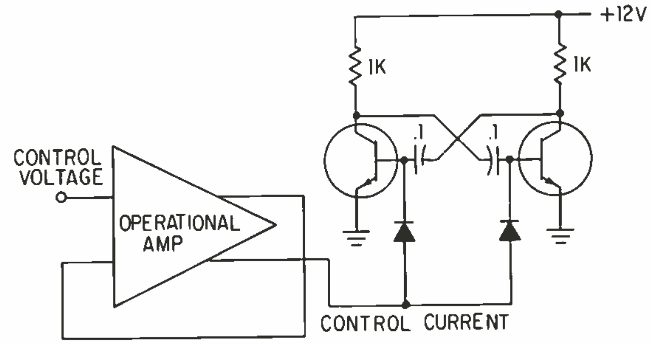

Figure 4. Schematic for a voltage-controlled multivibrator.

Probably the area of controllers is the most fertile for development at the present time. The reason is the natural attraction of a performable instrument, that is, one which can be played in *real-time* concert performances, rather than simply generating sound for tape which must be later spliced to assemble the piece. Among the parameters of the various function generators desirably controlled by the player are the simultaneous frequencies of several oscillators. The individual amplitudes of each, and the envelope applied to each oscillator should be controllable. It is also desirable to have control over tone color, attack height, and other new param-

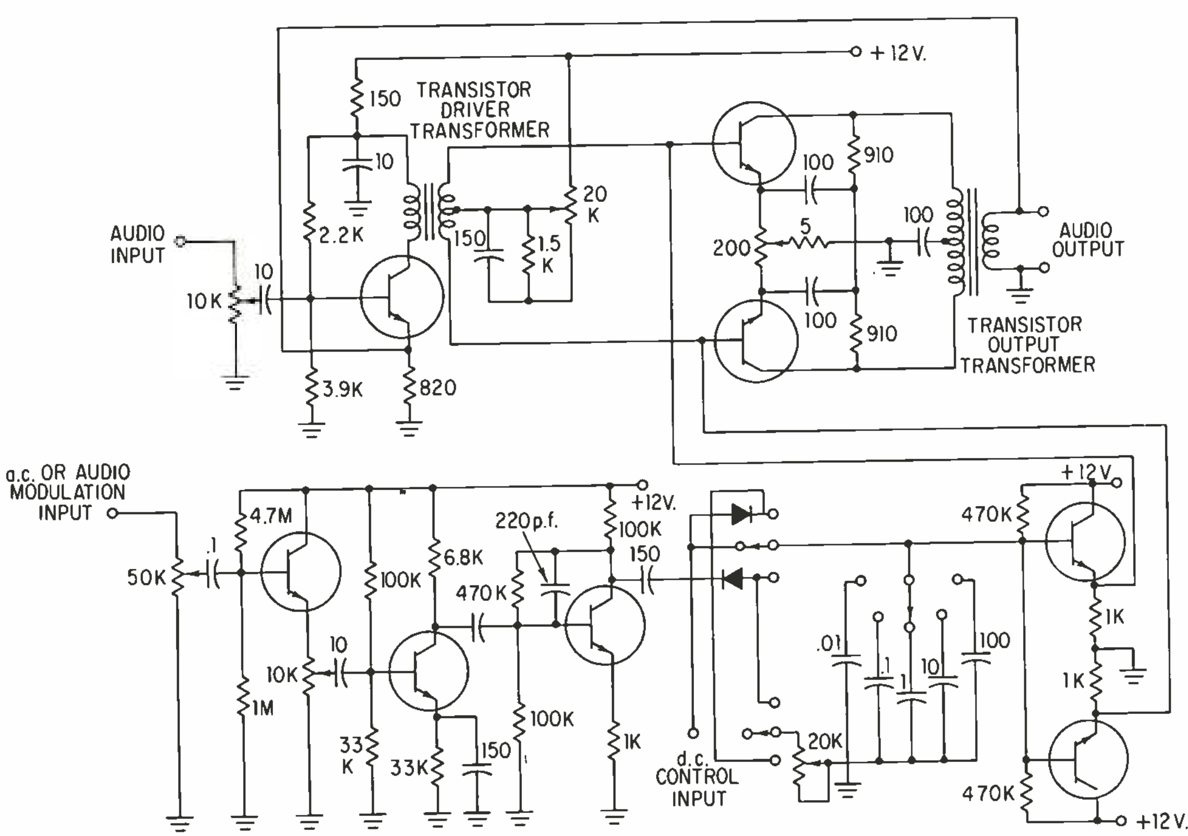

Figure 5. A voltage-controlled amplifier. All transistors are 2N 1304.

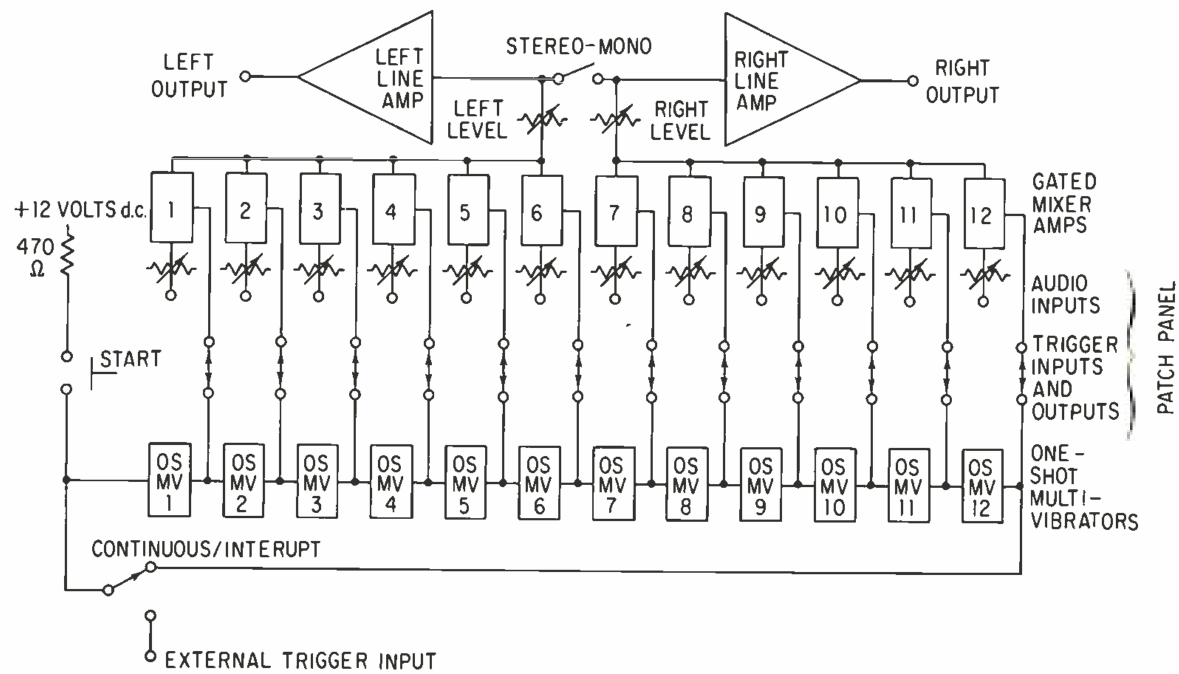

Figure 6. Block diagram of a sequencing mixer.

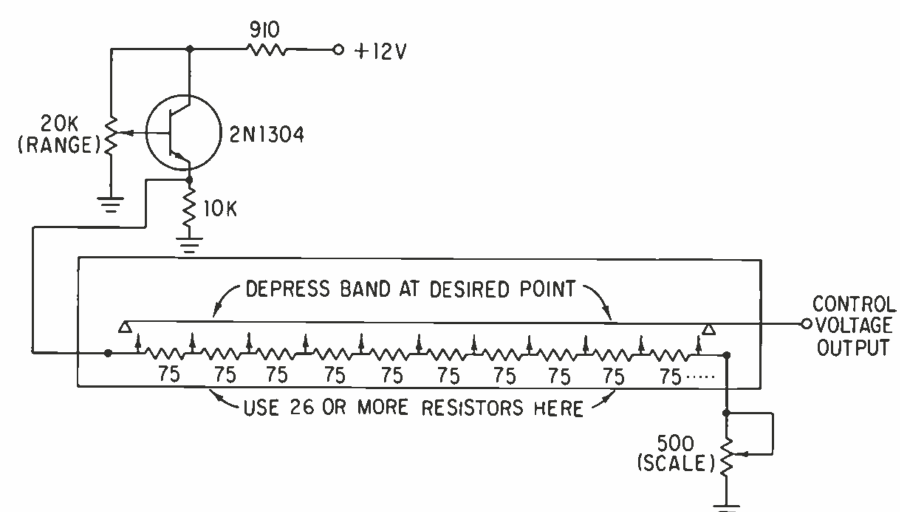

Figure 7. A linear controller with steps.

eters such as reverberation and resonance. No musical instrument in history has ever provided all of these parameters in a real concert-performance instrument. This is, perhaps, the greatest challenge facing electronic music at the present stage of its development.

One unique unit in the present synthesizer is the sequencer. This is actually an automated control device useful in rapid articulation of musical structures. The sequencer is a type of envelope-control device which works through a twelve-input sequencing mixer. The principle of operation is that each of twelve one-shot multivibrators generates a pulse and in turn triggers the next one-shot. The twelve sequential pulses are used to gate as many as twelve input audio signals through the sequencing mixer channels. FIGURE 6, a block diagram of the sequencer and sequencing mixer, illustrates the basic operation. The one-shots are at the bottom of the figure and the mixer amplifiers at the top. Interconnections are brought out on a patch panel.

The sequencer is one device for increasing the flexibility of control available to the electronic music performer. At the same time, it eliminates some elements of choice because of automated control of some parameters. This is often a condition of flexibility and it is often a difficult choice to make. Greater automation makes complexity easier but it takes many decisions out of the hands of the operator. He then decides many parameters by means of the blanket decision — To use the sequencer, for example.

REFERENCES

1. R. C. Ehle, *A Complex Tone Generator for Electronic Music, Audio*, October, 1966.
2. R. A. Moog, *Voltage-Controlled Electronic Music Modules*, Audio Engineering Society Preprint, 1965.

Test Tapes

ROBERT K. MORRISON
and JOHN G. McKNIGHT

Anyone that uses a tape machine sooner or later must use a test tape. The test tape is a valuable tool with certain limitations and restrictions. The authors provide a clear understanding of the values of this instrument.

Reproducer test tapes are used by professional audio technicians to standardize the azimuth, frequency-response characteristics, and recorded levels of magnetic sound-recording systems. Nearly 40 per cent of test tapes are sold to sophisticated consumers, who wish to test their audio systems at least through the speaker terminals. Professionals in radio and recording and manufacturers of tape-recording equipment account for the rest of reproducer tape sales.

Test tapes (also called alignment tapes or standard tapes) make possible the practical measurement and adjustment of magnetic-tape reproducers, for recorded material must reproduce satisfactorily not only on the machine which made the original recording, but on all similar equipment.

CARE OF TEST TAPES

Tape intended for repeated use in standardization work must be properly cared for if full usefulness is to be maintained. Tapes should not be stored in strong fields, such as those of permanent magnets or motors. Heads and tape guides should be demagnetized. When a reproducer test tape is used for continuous check-out purposes, as in production line work, age and wear often become the primary sources of inaccuracy.

Certain errors arise in test tapes after their manufacture. The most common are accidental erasure damage, mechanical deformation, magnetic damage and normal wear.

Mechanical and/or magnetic damage will destroy the accuracy of test tapes. Such accidental damage may occur in playing, rewinding, or storing tapes. The shorter wavelength (high-frequency) tones on the test tapes are most easily damaged.

Prolonged contact of the tape pack with one reel flange may result in irreparable edge damage. Mechanical deformation of the tape usually will damage the edge of the tape, causing uneven tracking and a constantly changing relative azimuth of the recorded tone. Edge damage can be prevented by winding the tape smoothly under moderate tension and evenly spaced between the reel flanges. High temperatures and temperature and humidity cycling aggravate mechanical distortion of a test tape, particularly when the wind of the tape on the reel is uneven.

Effects listed as accidental damage may be small enough to cause a type of accumulative damage not apparent from one use of the tape. But repeated usage will result in gradual loss of accuracy. The tape surface also will wear (lose oxide), even if the tape transport is perfect. Loose oxide also may become welded to the tape surface, causing increased spacing loss.

When the tape passes around small radii, the mechanical bending may cause some loss of megnetization, especially at short wavelengths. This loss depends on the tape used, but is usually about 0.5 dB at 0.5 mil wavelength.

As an example, a test tape which has been carefully handled and played 50 times will likely have a loss of 0.5 to 2 dB at 0.5 mil wavelength (15kHz at 7½ inches per second). After 100 passes, the loss may be as much as 3.5 dB at short wavelengths. Additional plays and/or slightly defective reproducers will cause the loss to approach 5 dB or more.

Ampex prolongs the useful life of test tapes by compensating for some of the losses experienced in normal usage by recording the shortest wavelengths at a level slightly higher than that prescribed in the National Association of Broadcasters' Standard: a boost of 1.25 dB is used at 0.5 mil. To reassure the purist, it should be noted that this boost is within the manufacturing tolerance attainable and simply means that Ampex carefully controls the flux to be on the plus side within its manufacturing tolerance. Further, only by this means can Ampex be sure that the user ever receives a test tape with the standard flux at the short wavelength.

In the case of 7½ in./sec. tapes, the average test tape at first play will have a short wavelength flux of 0.6 dB above the standard value, while the 15 in./sec. test tape will be 0.3 dB high. Within the first dozen plays, the test tapes will pass through the standard flux level and then drop below the standard value. Recent experience with certain European test tapes for 150 mil cassette machines has shown that, in the

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case of very short wavelength tones where no boost was involved and where the tapes may have been purposely underbiased to facilitate the making of the level at the high-frequency information, the losses encountered were so great as to make the tapes worthless as tools of measurement. These tapes were at speeds of 1⅞ in./sec. and were demonstrated to be down 7 or 8 dB at the 10kHz region, even after careful use.

It is apparent that test tapes must be recalibrated or replaced periodically, no matter how carefully the tapes are handled. Even though a system with short wavelengths may be adjusted to have flat over-all response, true flux response on an interchangeability basis may be difficult to obtain due to errors in the test tapes. Also, even when a system is set up to be flat on an interchangeability basis, the short wavelengths recorded on the tapes are fugitive, just as those on the test tapes, and a flat recording at slow speed (3¾ in./sec. or less) which is flat today may well be lacking high frequencies after storage or after a number of playings.

USING TEST TAPES

In order to avoid measurement errors during the use of a test tape, it is essential to observe certain special precautions, in addition to the use of ordinary good engineering practices. High- and low-frequency errors of measurement amounting to 3 to 6 dB or more are a likely result of improper procedures or the failure to observe needed precautions. Common errors of procedure cause an apparent rise of low-frequency response or a decrease of high-frequency response, or possibly both at once. Therefore, when measurement errors occur, the response will almost always fall with increasing frequency.

Test tapes may be inappropriate for use on a given system in any of the following areas: *rated tape speed, flux-frequency-response* (equalization), *track format,* and/or *recorded test frequencies.*

It is obvious that a 15 in./sec. reproducer must be tested by use of a 15 in./sec. test tape. It is not so obvious that a multiple-speed reproducer must be tested and adjusted at each of the speeds, for a response measurement at one speed guarantees nothing about the response at the other speeds. Since the NAB equalization curve is identical for both 7½ and 15 in./sec., it is commonly assumed that an adjustment at one or the other tape speed suffices for bath. Because of the 1:2 ratio of wavelengths involved this is only approximately true. Accurate response measurements require the use of both 7½ and 15 in./sec. test tapes.

FRINGING

All proposed NAB Standard test tapes, and most Ampex test tapes are recorded across the full width of the tape. A very common error is that of using a full-track test tape to measure the low-frequency response of a multi-track reproducer. When these tapes are reproduced by narrower-track heads (e.g., half-track, stereo, or multiple-track heads), a low-frequency measurement error occurs because of the fringing effect — at long wavelengths (low frequencies) the reproducing head core receives effective flux from the recorded track area outside of the area actually contacted by the reproducing head core. This error is a function of the recorded wavelength (which equals the tape speed divided by the recorded frequency), the particular design of the head shielding, and the geometry of the tape wrap over the head face. With certain heads the fringing effect may cause a rise of up to about 5 dB in the apparent low-frequency response of the reproducer. (See FIGURE 1.) Actually, it is due to the use of the wrong test tape, since a multi-track test tape must be used for accurate low-frequency measurements of a multi-track reproducer.

No satisfactory solution has yet been found for producing test tapes for all of the various multi-track configurations of reproducers. At least 11 standard (but different) flux-frequency responses are used for four common speeds. If each flux-frequency response were produced in each track configuration, a catalog of 88 reproducer test tapes would be required. Others such as 1⅞ in./sec., and Ampex mastering equalization at 15 in./sec., could undoubtedly be added. It just is not possible to manufacture, catalog, and distribute reproducer test tapes for every track configuration, because to do so would make their cost prohibitive. So reproducer test tapes are all made in the full-track configuration, but relatively few test tapes are made in multi-track configurations.

Another means by which fairly acceptable data on multi-track reproducers could be determined even though using a full-track test tape would be for the tape recorder manufacturer to include in the instruction manual of each model of multi-track reproducer the response of that reproducer to a full-track test tape, when the response is flat with the correct track-configuration of the test tape.

MECHANICAL ADJUSTMENT OF THE HEADS

The gapped reproducing head is a flux collector which gathers flux from the recorded track. Proper flux collection depends on having intimate contact and correct alignment between the recorded track and the gap area. Any imperfection of this alignment or contact will result in losses at some or all frequencies. In principle, errors of contact and orientation can be corrected in the reproducing equalization. The practical problems are:

1. *A system with faulty contact and/or alignment is usually unstable, i.e., the response is variable during the measurement, and from one measurement to another.*
2. *The usual equalizer is incapable of correcting for faulty contact and/or orientation, i.e., the range and shape of equalizer responses do not generally match the response of the system in which there are contact and/or orientation losses.*
3. *Although faulty contact and/or orientation reduces the signal and the tape noise by approximately the same amount, the head and amplifier noises remain constant, so that the signal-to-noise ratio is usually degraded.*

Because of these reasons, the contact and orientation should be mechanically adjusted in order to prevent errors

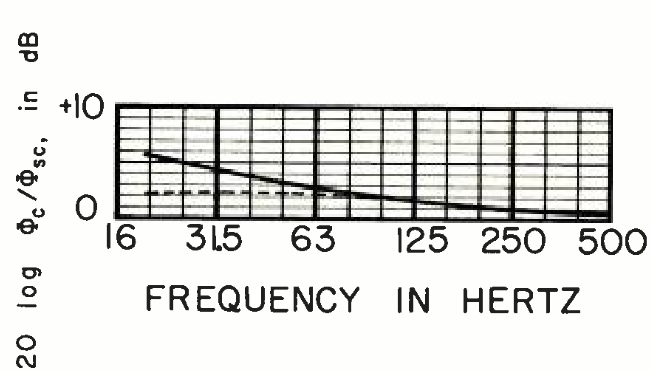


Figure 1. The fringing response rise when reproducing a full-track recording with a half-track reproducer (solid curve and a stereo reproducer (dashed curve) at 15 in./sec. Φ_c is the flux in the head core, and Φ_{sc} is the short-circuit tape flux.

in respect to a number of variables.

Height of the reproducing head should be adjusted accurately so that the reproducing head cores will coincide with the recorded tracks. Head height is especially critical with narrow tracks, such as four and eight-track systems. A misalignment of the reproducing head, causing the recorded test tape track to contact only a portion of the reproducing head core, will cause a reduction of the reproducing core flux at medium and short wavelengths, but, due to the fringing effect, not at long wavelengths. Thus there is both a level-setting and a frequency-response error. Misadjustments of recording and reproducing heads also may cause recorded levels which are too high or too low, depending on the nature of the misalignment. Too high a recording level will in turn cause high distortion. Head height errors escape detection when full-track test tapes are used with multiple-track reproducers.

The gap of the reproducing head should be parallel to the gap of the recording head. Standard practice is to make both of these gaps perpendicular to the edge of the tape. These relationships will be affected by the azimuth and zenith adjustments, and the tape guiding.

Practical azimuth adjustment is made by reproducing the azimuth adjustment section of a reproducer test tape. As the azimuth angle of the reproducing head is changed, the signal output will rise and fall. The adjustment must be made to the peak with the maximum output (FIGURE 2). If, before adjusting the azimuth from the test tape, the reproducing head is approximately parallel to the edge of the tape, the chances of setting to the wrong peak are greatly reduced.

Proper guiding requires that all tape guides and the front faces of the heads be parallel to the axes of the tape reels. These, in turn, usually are perpendicular to the top plate. This adjustment of the heads is called the zenith adjustment—all axes should point upwards. If any of the elements are not parallel, any change in tape tension will cause the tape to bow, producing an apparent azimuth change. Since in this case the azimuth depends on the tension, unstable high-frequency response results. A simple technique will aid in setting the zenith adjustment:

1. Paint the face of the head with dye, using a red felt-tip marking pen. (Wax pencil is too messy, while a layout fluid takes too long to wear off.)

2. Play a piece of scrap tape until the dye is worn off the head face where the tape runs.

Observe the wear pattern of the dye on the head face. If the zenith adjustment is correct, the right and left edges of the wear pattern will be parallel; if they form a V, the zenith is incorrect.

Changes of position of the tape edge also can cause the tape to bow, resulting in the apparent azimuth changes mentioned above. If the tape guides are too wide, the tape edge will wander. If guides or heads have a slot worn in them, then different widths of tape will lie in them differently. This change is especially noticeable if heads are readjusted after having been allowed to wear in an incorrect adjustment.

Very small spacings between tape and head cause large losses: the slope is not "6 dB octave," for expotential, i.e., the slope increases with frequency. Unintentional spacing may come from several sources.

Material (usually loose tape oxide, or loose scraps of oxide and base material from the slitting process) may accumulate on the head face in use, causing spacing loss. Heads should be cleaned carefully before measurements and/or adjustments are performed.

Tape-to-head force which causes tape-to-head contact is commonly obtained in professional recorders by having the heads deflect the tape path between two guides. In this case the force which holds the tape in contact with the head is proportional to the hold-back tension. Low contact force may therefore be caused by a hold-back tension that is too low because of improper adjustment, or because of the use of large reels with the reel size (tension) switch set for small reels (low tension). This low force again allows separation of head and tape.

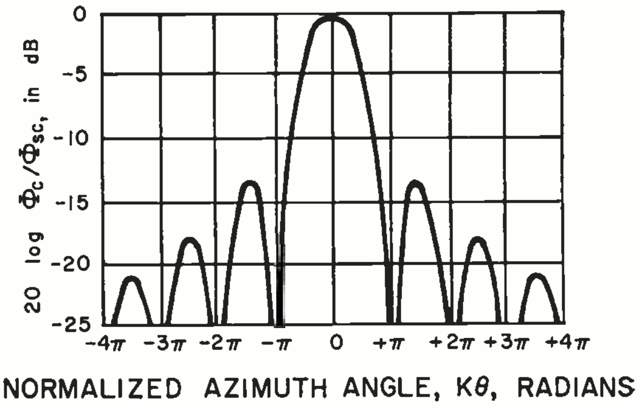


Figure 2. As the azimuth angle of the reproducing head is changed, the signal output will rise and fall. In this graph, output versus azimuth angle, where θ = azimuth angle in radians, $\pi K = x/\lambda$, w = track width, λ = wavelength (in the same units), and Φ_c/Φ_{sc} = (s in $K\theta$)/$K\theta$, for $\theta < 0.2$ radians (11°).

In the design of a head assembly or in replacing heads in an adjustable head assembly, it is possible to have too little wrap of the tape around the heads. For a given tape tension, head-to-tape force is a function of the wrap angle; angles of less than about 12 degrees total deflection of the tape at each head are unlikely to give adequate contact force for the elimination of the spacing loss.

In the machines described above which obtain tape-to-head force by having the heads deflect the tape path between two guides, the gap of each head must be at the vertex of the angle so formed in order to have best tape-to-gap contact.

Mention was made earlier of guiding problems due to a head which has a slot worn in it. Such a slot causes a further difficulty: when a head wears, the cross-section usually is stepped. The steps often cause the edge of the tape to be lifted out of contact with the head face, with the resulting spacing loss.

EVOLUTION OF "STANDARD" FLUX-FREQUENCY RESPONSES AND LEVELS

Standardizing organizations such as the NAB have been prolific in their recommendations, with the result that 27 different reproducer test tapes are available from Ampex, the firm that produces almost all test tapes used in the U.S. Old standards cannot be dropped simply because a new standard is adopted by one, or even all, standardizing organizations. Audio equipment and tape libraries suited to an old standard still exist, and the manufacturer must provide corresponding test tapes for many years.

In 1953 the NAB published a description of a standard reproducer for 15 in./sec. use, based on an idealized ferromagnetic reproducing head and an amplifier of specified response. The reproducing flux-frequency response had time constants of 3180 microseconds and 50 microseconds. At the 15 and 30 in./sec. speeds then prevalent, the wavelength remained relatively long even at high frequencies — a 1 mil wavelength (15 kHz at 15 in./sec.) was the shortest encountered in general audio usage. Even without a test tape, the error might be small if the reproducing amplifier was adjusted to the specified response, and the reproducing head was assumed to be ideal. But the test tape was definitely needed to adjust head azimuth and operating level.

It was shown by experiments that a 15 in./sec. test tape suffered very little degradation when used carefully. Carefully stored primary and secondary reference response tapes were employed to calibrate test-tape production equipment over long periods of time.

Because several different reproducing frequency responses were in use for 7.5 in./sec. in the early 1950's, the NAB did not standardize a reproducing frequency response for this speed. Eventually the frequency response originally used for 15 in./sec. (3180 microseconds and 50 microseconds) also became generally accepted for 7.5 in./sec., thus becoming a standard by usage.

No mention of the recorded level was made in the 1953 NAB Standard. The *Ampex Operating Level* was at first determined by measuring distortion, and thus would be a function of the particular tape used. A 15 mil wavelength signal producing 1 per cent third harmonic distortion on a selected batch of then-current 3M-111 tape became the flux reference for the *Ampex Operating Level.* It was soon realized that the operating level of Ampex test tapes had to be held at a constant absolute flux rather than at a constant distortion, so that practical compatibility of levels could be produced in the recording industry. Therefore, even though tape has changed, the *Ampex Operating Level* flux of test tapes has remained extremely constant to the present time. Fortunately, the tape selected in the original determination of long wavelength operating level had greater distortion at a given flux than present-day tapes. Thus the amount of distortion experienced today with most available tapes at operating level is less than the original 1 per cent.

A new standard was published by NAB in 1965 which reaffirmed the reproducing flux-frequency response time constants for 15 in./sec. and also recognized past industry usage of the same reproducing frequency response for 7½ in./sec. However, the new NAB standard does call for a change in the reproducing frequency response time constant at 3¾ in./sec. from 120 microseconds to 90 microseconds. A compromise "even value" of 100 microseconds has been proposed, also, and is now used in Ampex consumer products and Ampex stereo tapes.

AZIMUTH DETERMINATION

Azimuth determination is of course crucial when using reproducer test tapes. Correct head azimuth is obtained when the head gap is exactly at right angles to the edge of the tape. When correct head azimuth is obtained, it in turn produces a recorded track which is exactly perpendicular to the tape edge. (If tape is improperly slit, wound and/or stored, the edge naturally will be wavy and the relative azimuth will vary.)

Some very early test tapes were made with incorrect azimuth. Although not too noticeable at then-current 30 and 15 in./sec. speeds, it was corrected because it became quite apparent at slower speeds.

Several methods are used to determine the azimuth of a recorded track. The simplest in principle is to make the recorded track visible by means of carbonyl iron powder, by softening the binder with amyl acetate, or by using a tape viewer. A toolmaker's microscope then can be used to check the visible track's perpendicularity to the tape edge. Unfortunately, tape with a perfectly straight edge does not exist; therefore many measurements are required in order to determine the average angle.

Until recently this method was cumbersome and not too helpful. However, greatly improved optical measurement devices now make this method very useful with an excellent repeatability factor. (This method also is helpful in demonstrating edge curl, which may occur when imperfect tape winding allows the tape to rest against one reel flange.)

If tape is simply turned end-for-end, the azimuth error will be in the same direction, for it does not reverse. Some means thus are necessary for producing what may be termed a mirror image. A most satisfactory method consists of these six steps (See Figure 3):

1. A full-track, short-wavelength signal is recorded, using a combination recording and reproducing head with an arbitrary azimuth setting. An indicator and an angular scale should be added to the azimuth adjustment screw of the head assembly to show the relative azimuth settings: the angle for this first recording should be noted.

2. After the signal is recorded, the full-track tape is rewound with the oxide surface in contact with a blank piece of tape.

3. The two tapes are then run through the machine, with a small amount of bias current applied to the second head. (The erase head is of course disconnected.) This step causes a mirror image of the original signal to be printed on the adjacent blank piece of tape.

4. The printed tape is reproduced, the head adjusted for this new azimuth, and the new setting noted.

5. The azimuth is set halfway between the two azimuth angles measured in steps 1 and 4.

6. Steps 1 through 5 are repeated until masters and prints both show maximum response at the same setting, thus indicating perpendicularity of the head gap to the guided tape path.

Another method for producing the mirror image involves turning the tape over and reproducing the signal through the base of the tape. The adjustment method is similar to that just explained. The problem, though, is that with the short wavelengths necessary for accurate azimuth adjustment, the loss of signal through the backing of the tape is so large that it is difficult to find the signal at all.

This principle of azimuth determination may be made more practical by using a two-track combination head. The two gaps must be exactly coplanar (there must be no "gap scatter"). Phase of the two signals is compared, instead of seeking maximum amplitude. If the two signals are combined out of phase, the position of correct alignment results in a very sharp null of output even when a mid-frequency signal (such as 3 kHz at 15 in./sec.) is used. Therefore the signal amplitude can be large enough even when reproduced through the tape back.

Another method for determining azimuth has been tried using heads with a gap at the front and another at the back, and special guides for the tape. The front and back gaps of the head must be parallel to one another, and both surfaces must be lapped for good tape-to-head contact. The tape is recorded at the front of the head, rewound oxide-out (B wind), then reproduced by the back gap of the same head (Figure 4). Thus the head rather than the tape produces the mirror image. Adjustments are made as described above until perfect agreement is achieved between back and front gaps. Guiding problems have to date made this method unsatisfactory.

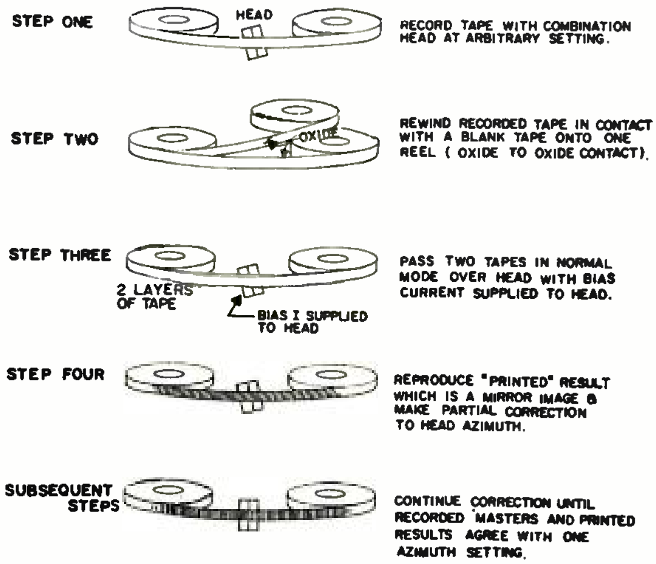


Figure 3. Azimuth determination using the master and transfer print method.

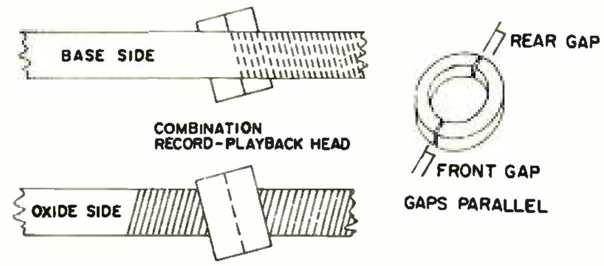


Figure 4. Azimuth determination using front/back gap method. Above, the recording is made on the front face of the head. Playback is accomplished on the back gap. The head is adjusted until the recording made on the front gap plays back in agreement with the rear gap.

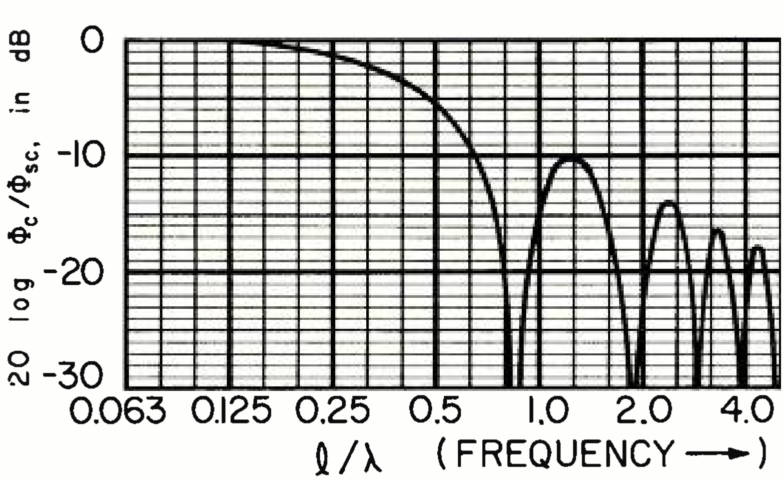
CALIBRATION OF THE FLUX VERSUS FREQUENCY

Procedures detailed in the 1965 NAB standard are generally followed for calibration of the reproducing channel used in making reproducer test tapes. First the response of the reproducing head is determined along with the electronics from a "constant flux available" input. (A flux-inducing loop is attached to the front gap of the head, and a constant current signal vs. frequency is applied through the loop.) The amplifier can then be adjusted to provide the correct response for the head, assuming a flat wavelength response. See Figure 5 and 6.) (In this measurement, resonance effects of the head and cable are included. It simplifies matters, if the playback head resonates well outside the bank of interest; this requires a low-inductance head, with fewer turns of wire than usual.) A de-emphasis network may be used with this flux-inducing method after the reproducing amplifier to produce a flat reading on a voltmeter when the reproducer response is correctly adjusted to the appropriate standard. A different response is of course needed for each of the many standard curves in use.

Wavelength response errors will exist with a practical head, and these must be measured and taken into account. Gap losses and contour effects are measured as outlined in Annex C of the 1965 NAB Standard. These measurements are somewhat involved, and it is best to doublecheck them. Additionally, several known heads should be measured for later cross reference.

A curve may be drawn showing the deviation from ideal of a particular reproducing system after the losses inherent in the reproducing head are determined. A recording then may be made which will reproduce in agreement with the calibration curve. In other words, test tape should ideally play back with the same response as the calibrated reproducing system. The inherent deficiency of the calibration reproducer must be known so that the corrections will be applied to the calibrated *reproducer*, not to the test-tape recording.

After a system is calibrated, a tape may be made for use in adjusting other test-tape production machines, for test-tape production on a comparative reference basis. Each machine must produce tapes which have recorded flux identical to that of the tape made on the calibrated system. In production



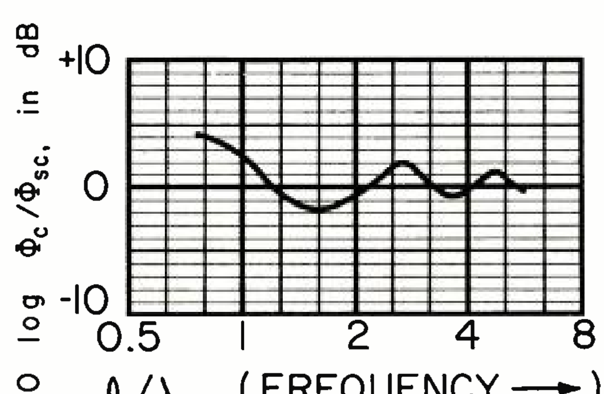


Figure 5. (Left.) The theoretical short wavelength response of a reproducing head with a long gap, where l is the mechanical gap length. Figure 6. (Right.) The theoretical long wavelength response of an unshielded reproducing head with rectangular pole pieces, where L is the length of the head core face and λ is the recorded wavelength.

practice, several tapes are made on the calibrated reproducer and used as setup tapes. But it is very important to note that setup tapes do not last long, and must be recalibrated frequently on the original test setup or else replaced entirely.

SUMMARY

Although the reproducer test tape provides the most satisfactory means known for measuring and adjusting the azimuth and frequency response of tape reproducers, there are numerous opportunities for making errors in the measurements. If care is taken in performing the measurements and these errors are avoided, measurements with a test tape should give accurate indications of the frequency response of a reproducer. Used correctly, test tapes are a valuable addition to both professional and amateur audio systems.

Readers seeking additional information on Ampex test tape applications will find their 12-page brochure Test Tape Applications useful. It is available without charge by circling number 100 on the reader service card bound into the back of this issue.

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● This fourth generation of Neumann stereo cutterheads brings technical characteristics long sought by the recording industry. The SX-68 is made to satisfy the most stringent demands for smooth response, low distortion, high channel separation, high recorded level, and extreme ruggedness. As with its predecessors, this head is fitted for helium cooling. Important characteristics are: secondary resonance above 75 kHz; high feedback capability with a reserve to prevent oscillation resulting in less than ±1 dB frequency response deviation, and extremely low distortion over the entire audio range; channel separation is 35 dB over the full range; vertical cutting angle is 15°; and construction is rugged. An important design parameter is the ability to cut (with helium cooling) lateral velocities at 10 kHz of 40 cm/sec for ten seconds, or 26.5 cm/sec continuous sine wave.
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● This Pilotone version of the model 11 portable uses a two-track head with the second channel used for standard motion-picture pilot-sync signal recording. Camera and synchronizer interconnections are provided. An indicator on the front panel of the recorder shows that the pilot tone is being connected. The model 11-1-P records at 7½, 3¾, or 1⅞ in./sec. There is seven-inch tape-reel capacity. For audio there are three heads, a 40-16,000 Hz response ±2 dB at 7½, and a 56 dB s/n. Distortion is under 0.5 per cent. A servo-type speed control is used, there is a built-in mixer and limiter, and cannon-type mic connector. Ten "D" cells provide power. Weight is approximately ten pounds.

Mfgr: Tandberg of America, Inc.
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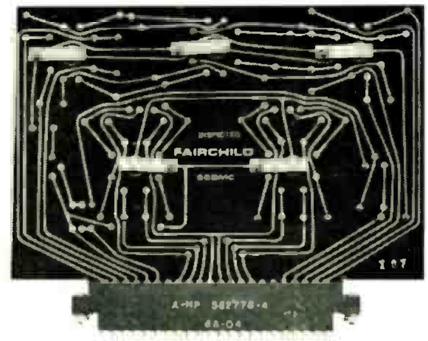

● This single plug-in card, model 668MC, contains four interlinked faders capable of attenuating four audio channels simultaneously with less than 1 dB of mistracking between channels. The card's light-dependent circuitry allows remote control with a minimum of crosstalk and a high degree of reliability. With plug-in light sources that can be replaced instantly, the card is available in 600- or 150-ohm impedances. The card can be driven off a slide actuator or rotary potentiometer. Several cards may be connected to one actuator so that as many as 32 or more channels can be handled simultaneously.

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People, Places, Happenings

● **Benjamin B. Bauer,** an audio research scientist for **CBS Laboratories,** has been elected president of the **Audio Engineering Society** for the coming year. Mr. Bauer has been responsible for many innovations in audio and acoustics, including highly advanced microphones for the broadcast field, phonograph sound pick-up devices and tape-recording heads. He has spearheaded many CBS Laboratories developments including the recent introduction of a loudness-level monitor which analyzes sound frequencies and converts them into indications of loudness. The device then adjusts them to a pleasing level for radio and television broadcasting.

● **Frederick B. Bundesmann** has been named broadcast systems sales manager by **Philips Broadcast Equipment Corp.,** manufacturer of **Norelco** closed-circuit and broadcast-television equipment. The announcement was made by **Anthony R. Pignoni,** director of marketing. Mr. Bundesmann formerly was eastern regional manager, **F & M Systems Company,** and prior to that was a sales engineer for **MGM Telestudios.** In his new post he will coordinate the efforts of the company's field sales engineers on major broadcast systems installations.

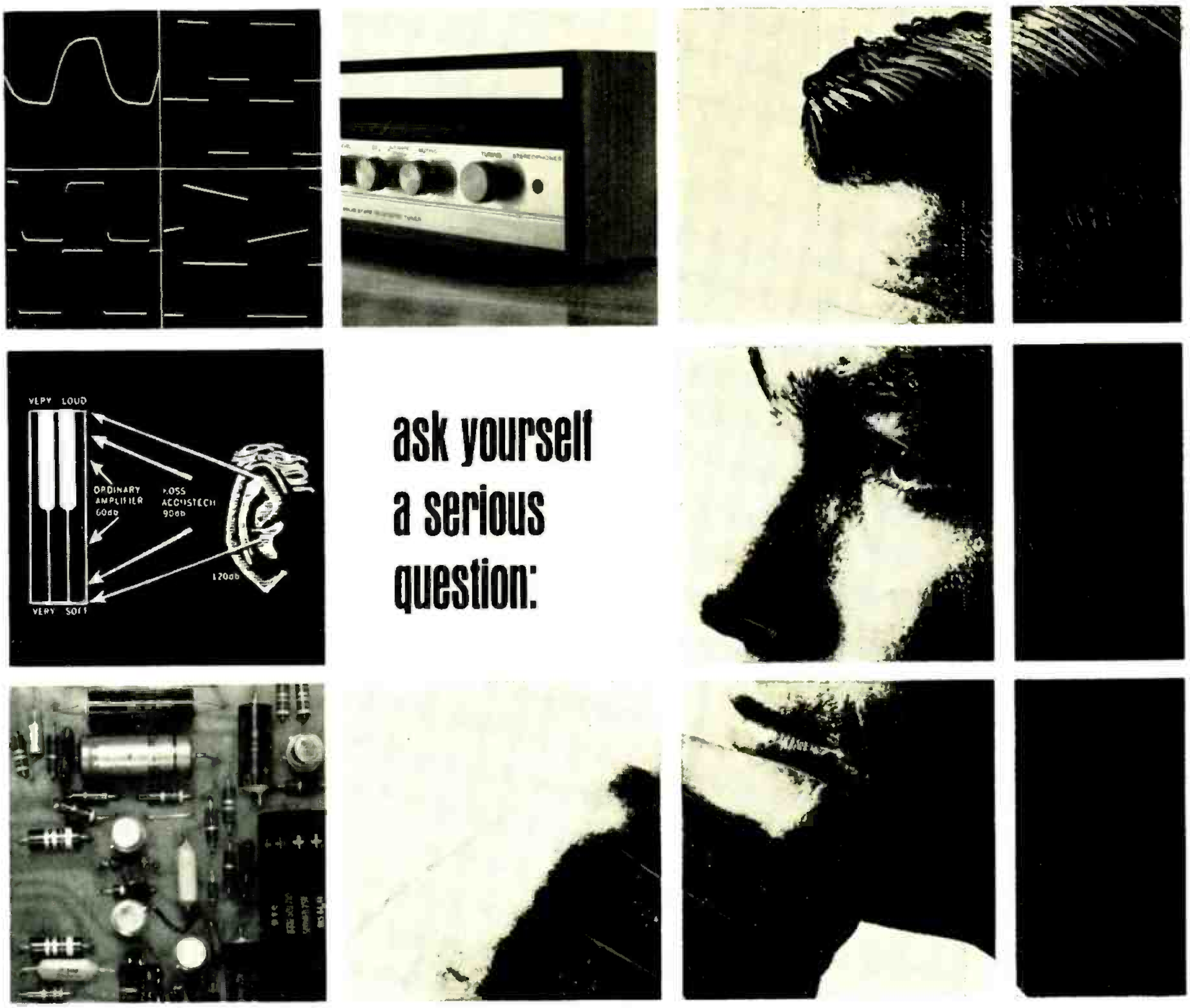
● **Recorded Publications Laboratories** of Camden, New Jersey has completed installation of a new automatic stereo facility for the production of stereo disc masters. According to **David Goodman,** executive vice president of the firm the new equipment was custom designed and constructed by Dr. Georg Neumann in West Berlin and incorporates the new Westrex 3D stereo disc cutting system as well as Ampex playback equipment for program input. The system is computor controlled to handle every facet of the audio and mechanical requirements of disc mastering. According to Mr. Goodman, the combination of a skilled mastering technician and the computer system assure high standards in consistency and quality of the finished masters.

● An announcement by **Walter Stanton,** president of Stanton Magnetics Inc. calls attention to the return of **George Petetin** to the organization. Mr. Petitin is a former employee of **Pickering** (pert of the Stanton group), having served as dealer sales manager from 1950 to 1962. In 1962 he moved to **Reeves Soundcraft** as distributor sales manager. Since January of this year he has been manager of the special products division of **JBT Instruments, Inc.** In his new position he will fill the post of sales manager of Stanton Magnetics Inc. According to Mr. Stanton this move is one of several designed to broaden the marketing approach of Stanton Magnetics Inc. and Pickering and Company, Inc. **C. Ray Bennett** has been advanced to the newly created post of corporate sales administrator. Stanton Magnetics Inc. is a supplier of professional products for the recording and broadcast industry.

● From **Fairchild Recording Equipment Corp.** comes word that **David Bain** has been appointed as manager of application engineering, a newly created post. In a statement by **Edwin J. Everitt,** president of the firm, Mr. Bain's addition to the staff is called another major step in the total expansion program of the firm. Mr. Bain has an extensive background in professional audio. As a partner in the firm of **Joseph & Bain,** he acted as sales representative for professional and broadcast-equipment firms. Prior to that he was general manager of the product division of the **Muzak Corporation.** Before that he was in the broadcast and television division of **RCA** serving in various sales and management posts in Camden, Chicago, Kansas City, and Washington. He has also been chief engineer of several radio stations.

● Several moves have been announced at **Ampex** divisions. **Charles S. Dolk** has been named marketing manager for the Ampex Corporation industrial and educational products division headquartered in Park Ridge, Illinois. In his new post Mr. Dolk will be responsible for marketing sales, and advertising of the division's line of closed-circuit videotape recorders, t.v. cameras, and associated equipment.

From Redwood City, California comes word that **Lawrence Weiland,** general manager of the Ampex Corporation video products division, has been elected a vice president of the corporation. Mr. Weiland has been with Ampex since 1960 when he joined as manager of video engineering. He also was video marketing manager of the audio-video group prior to assuming his present position.



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