db

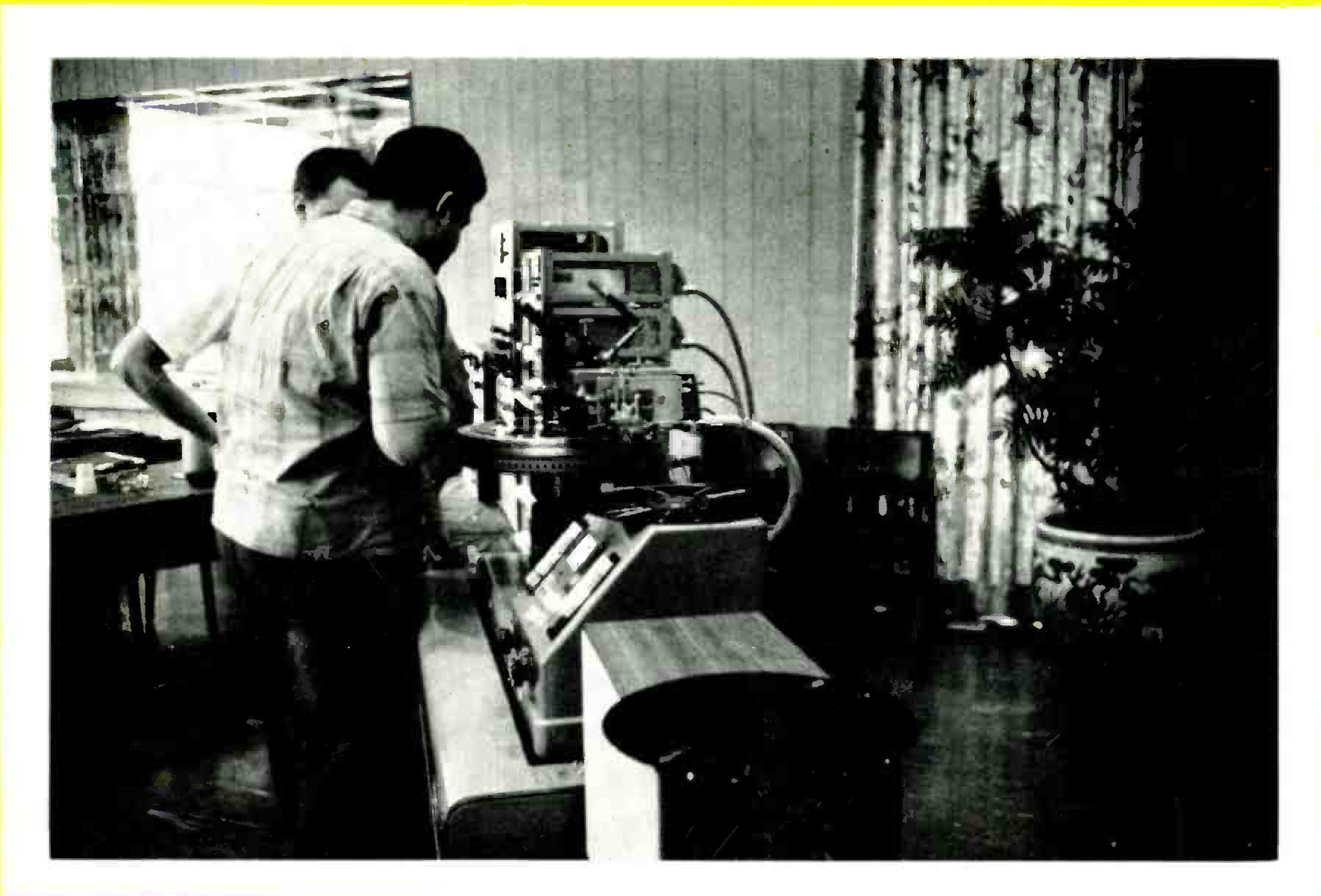
THE SOUND ENGINEERING MAGAZINE

May 1970 75c

A Disc Recording Seminar

Phonograph Record Compatibility

Hearing Loss and Audio Engineering



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Coming Next Month

● The concluding part of HEARING LOSS AND THE AUDIO ENGINEER by W. Dixon Ward tells you haw to determine your own TTS, talk about pre-existing hearing losses and concludes on the question—*is Music Dangerous?*

POWERING CONDENSER MICROPHONES by Dr. Gerhard Bore of Georg Neumann of Berlin will explain the advantages and disadvantages of the various means of powering transistor-equipped condenser mics.

H. van der Wal of Philips in the Netherlands describes a new tape drive system that permits multi-speed operation of the capstan under very precise control of the operator. The article is appropriately titled a MULTI-SPEED CAPSTAN DRIVE SYSTEM.

Next month will also feature a picture-gallery collection of equipment that was seen at the NAB exhibition recently held in Chicago. While much of the equipment is pointed toward broadcasters (naturally enough) there is still much that was new for others of the professional audio fraternity.

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

About the Cover

● A Neumann disc-recording system as was displayed at a recent disc-recording seminar held by Gotham Audio Corporation. The story begins on page 28.

GRT (Cov. II) Circle 10 on Reader Service Card

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THE SOUND ENGINEERING MAGAZINE

MAY 1970 • Volume 4, Number 5

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Letters

The Editor:

The coverage in the literature of, on, and about four (or more) channel stereo reproduction has set me to thinking. I wonder if we are not aimed off in the wrong direction? The problem, it seems to me, is to discover and develop a *practical* system of stereophonic sound reproduction. *Tetraphonic Sound* and it's four channels is one approach. (dB, 12/69) Trouble is, it involves twice as much of everything as the consumer now has—and the duplicates should be of a comparable quality level. I wonder if a *three* channel system has been researched enough: I refer to a conventional two-channel stereo mic setup, with an additional cardioid located at the rear of the hall—or an electrical equivalent—making the third channel.

This third channel could very well be matrixed in with the usual a&b signals, feeding a speaker *behind* the listeners for the surround effect. With tape, it is simple—three tracks instead of two. Discs may represent a magnitude or so of greater difficulty, although the little experimentation I have done has convinced me that the surround channel need have only limited frequency response, 200 to perhaps 5,000 cps (and a Hertz to you, too!) and reproducing equipment of lesser quality than the two main stereo channels. This system could readily be adapted to f.m. broadcast by utilizing the 67 Kc sca channel to transmit the surround channel while a+b(a−b) takes its usual course through the modulator. This looks like a good field of investigation for a younger man—anyone interested?

F. C. Hervey, MAES
Charlotte, N. C.

The Editor:

In his letter in the April 1970 issue, Mr. Odom is correct in attacking Mr. Smith's application of the noise formula to his preamplifier, but is wrong in his analysis of the advantages to be obtained with a high impedance pick-up. In fact he has understated them.

Mr. Odom's statement that one may overcome amplifier noise by raising source impedance, and hence voltage by the square root, is correct only if

(*Continued on page 33*)

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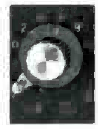

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The Feedback Loop

ARNOLD SCHWARTZ

● A typical day's broadcasting includes a drama review, a report from a local Congressman, closing prices on the stock exchanges, helicopter reports of traffic conditions, a consumer report by a former Miss America, a discussion of new record releases, music, and full news coverage—yet this station does not spend a penny for programming. We are talking about WNYC-AM/FM/TV, the only municipally owned non-commercial station in the United States. WNYC-AM went on the air July 7, 1924, only four years after KDKA in Pittsburgh. I had the opportunity of visiting the station when I appeared as a guest on WNYC-FM's weekly program, *Men of Hi-Fi*. Produced and moderated by Harry Maynard, editor of the magazine *FM Guide*, the program is a panel discussion which on this occasion dealt with the relative merits of disc and tape as a home playback medium. Interestingly enough, the topic was suggested by last month's Feedback Loop, and many of the points made in the article were discussed in greater detail on the show. A reprint of the broadcast is scheduled to appear in *FM Guide* in the near future.

WNYC, EXPERIMENTAL STATION

Seymour N. Siegal, WNYC's director, has served under six mayors since he was first appointed to the job in 1934. He describes WNYC as the leading station in the United States in the field of broadcast experimentation. WNYC was the first FM station in the New York City area, broadcasting on the old FM band. It was the first station to play an lp record on the air. The Municipal Broadcasting System got into the TV business when, in cooperation with the FCC, an experimental UHF station was established to determine the severity of the interference caused by tall buildings. The tests confirmed the feasibility of UHF transmissions in large cities, and the City TV station stayed on the air.

Currently, WNYC-FM is participating in experimentation with the most recent development in the high-fidelity field, four-channel stereo. In conjunction with WKCR, a local FM station operated by Columbia University, several four-channel programs have been broadcast. A four-channel tape recorder is located in the WNYC

"The Dolby System is essential

for the recording of chamber music,"

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

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

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

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Figure 1. New York City's Mayor Lindsay and WNYC Director Siegel inspect the FM transmitter, located in the Empire State Building.

Figure 2. WNYC's fleet of mobile units.

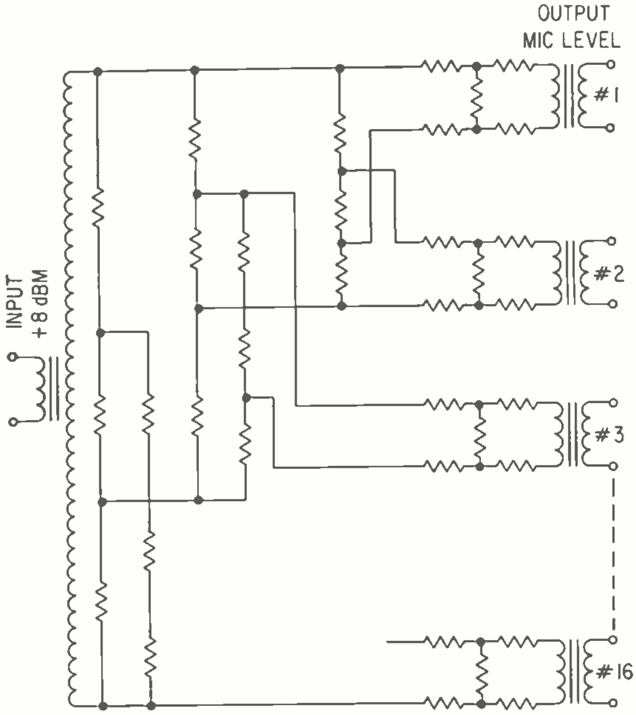


Figure 3. The WNYC microphone feedbox schematic.

studios, and the two front channels are fed to the WNYC-FM transmitter. The rear, or reverberation channels, are fed to WKCR-FM on telephone lines and broadcast simultaneously.

On the day I was at WNYC there was another first. Metropolitan area residents were able to see an opera on TV, and listen to the music in stereo. Channel 13 (NET Station) played a video tape of *My Heart's in the Highland*, an opera written by Jack Beeson, based on a short story of William Saroyan. The audio was synchronized to the video tape, and relayed to WNYC-FM for simultaneous broadcast.

WNYC FACILITIES

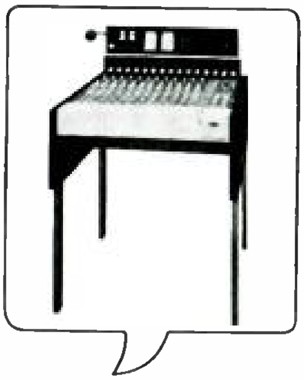
WNYC's studio facilities are on the 25th floor of New York City's Municipal Building. Because the station is municipally owned and supported by taxpayers, purchases for new equipment cannot be made directly. The need for the protection of public money dictates that all appropriations be reviewed by appropriate City agencies, with consequent delays. Salaries are set by the City government and the resulting wage differentials, compared to private broadcasters, make it difficult to hold on to trained and competent personnel. WNYC's large turnover is viewed philosophically by Mr. Siegel who sees his station serving as a kind of training ground for broadcast personnel. Judging from the many people I've met in the broadcast industry who got their start at WNYC, the Municipal Station has, in fact, performed this function. While this turnover may create problems for WNYC, it certainly has proved a benefit to the broadcast industry at large.

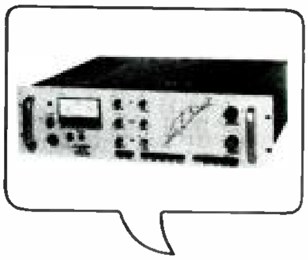
Mr. Hom Hong Wei, chief of radio operations, showed me around the studio facilities. Mr. Wei knows his station inside out, having been at his job for the past 29 years. The equipment presently in service varies from some of the latest and modern to some of the most ancient I've seen, with some items possibly qualifying for a display case at the Smithsonian Institute. One of the monitor amplifiers still in service is a Radio Receptor, model MA-11, serial #6—can anyone tell me how old that is? Despite these handicaps, WNYC does an excellent job with a complex programming schedule. The radio studios consist of two control rooms, one has a two-channel console, and the other is a single channel facility. In addition there is a master control area with switching equipment, jack fields, and tape recorders which can feed the program line directly. Expansion and modernization of both radio and TV audio are constantly going on. New York City's Mayor, John V. Lindsay, and Seymour N. Siegel, WNYC's Director, are shown in FIGURE 1 inspecting the FM transmitter which was installed in 1966. WNYC also has a fleet of mobile units which are shown in FIGURE 2.

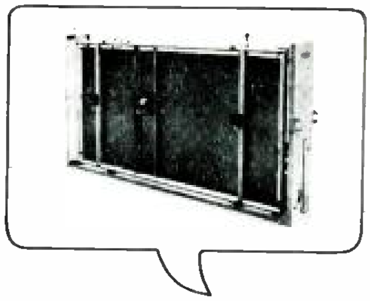
CITY HALL COVERAGE

WNYC has very active coverage of City Hall activities. There are frequent news conferences, ceremonies for visiting dignitaries, and broadcasts of important hearings of various municipal governmental agencies and committees. With more than forty radio stations and seven TV stations in the metropolitan area, one can imagine the forest of microphones that might sprout in front of the mayor on some of these occasions. To avoid this kind of a confrontation, only one WNYC microphone is used. The output of this microphone is fed to a Collins 212Z portable mixer. The +8 dBm output of the Collins is fed to what Mr. Wei describes as a *feedbox*. This is a passive network which splits the feed into sixteen isolated microphone-level outputs. Where more than sixteen outputs are required, additional feedboxes are used, and all radio and TV audio requirements are supplied by the feedbox outputs. The resistive splitter network used here is one that I have not seen before. The basic scheme of the circuit is shown in FIGURE 3. The feedbox input, as well as each of the outputs are isolated by transformers.

What impressed me most about WNYC-AM/FM was its ability, despite operating limitations, to maintain a high standard of technical excellence in the face of increasing demands upon it as a source of information, education, and culture. ■







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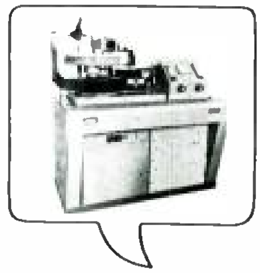

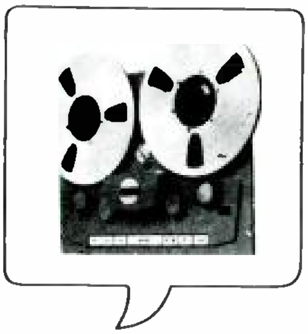

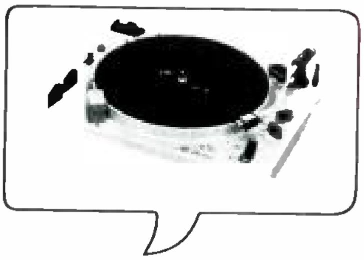

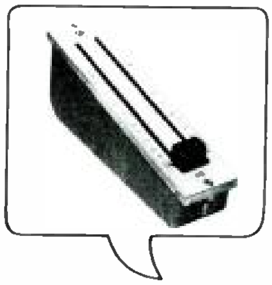

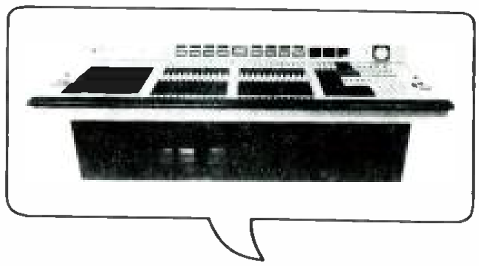

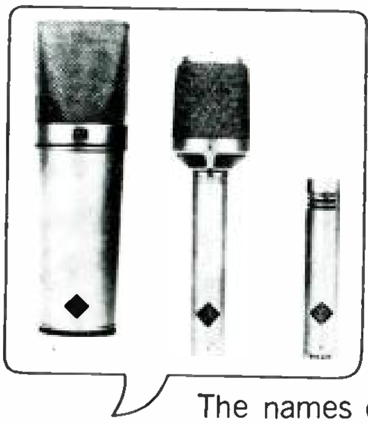

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The Sync Track

JOHN M. WORAM

STARTING AN ELECTRONIC MUSIC STUDIO

● With the new-found popularity of electronic music, most conventional recording studios sooner or later require the use of an electronic music synthesizer. Depending on the particular situation, there are several alternatives. One can rent time at an existing electronic music studio and do the required work there. Or, in some cases a synthesizer may be rented as required. And of course, you can always purchase your own synthesizer if extensive work in this area is anticipated.

Whether a synthesizer is rented or purchased, there is often some confusion about how to best use it in the studio. For simple keyboard-type effects, the synthesizer may be treated like any other electronic keyboard instrument and played on session along with conventional instruments. However, when complicated sounds are required, a lot of time (money) can be spent if studio musicians are kept waiting while the synthesizer operator makes his set-ups.

In this case, it is usually more practical to add the synthesizer parts later. If this is done in the recording studio, there's usually no problem, except that the studio is being tied up by what can become a lengthy process. It would certainly be more economical if this type of work could be done in a tape-mastering (mix-down) room, so that the studio could be free for live recording. Also, mastering time is usually cheaper than studio time, which is a consideration for those paying the bills.

The usual mastering room is set up to mix down a multi-track tape to two tracks. However, for involved electronic music, one needs to be able to record onto a multi-track machine, one (or more) tracks at a time. And, of course, a flexible monitoring system is a must in this type of recording just as it is in conventional live studio sel-syncing.

At first, it may seem that the requirements of involved electronic music work are beyond the capabilities of the usual tape-mastering room. However, with a few modifications, most mixing consoles can be used for this type of work. The amount of modification required depends on how much flexibility has already been built into your console. In most cases, very little rewiring has to be done.

Figure 1 is a (grossly) oversimplified flow diagram of a typical tape-mastering facility. The units indicated by broken lines show modifications that

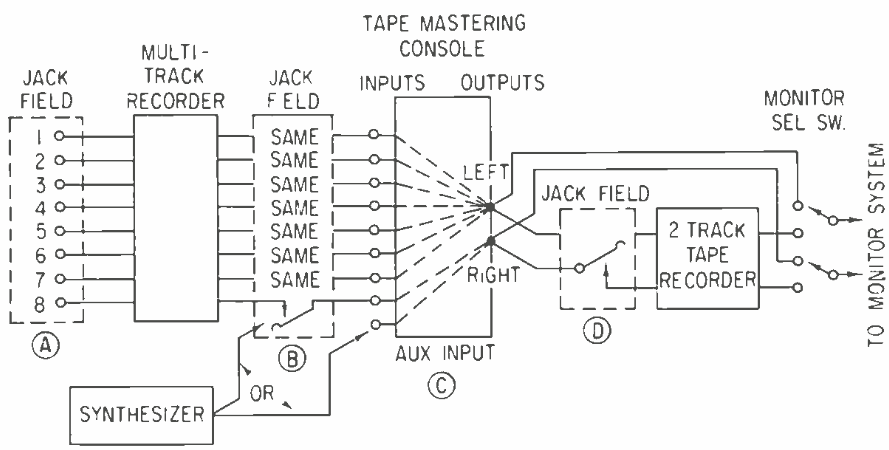

Figure 1. A representation, oversimplified to be sure, of the audio flow in a typical tape-mastering facility.

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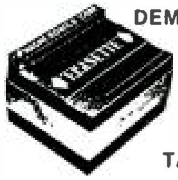

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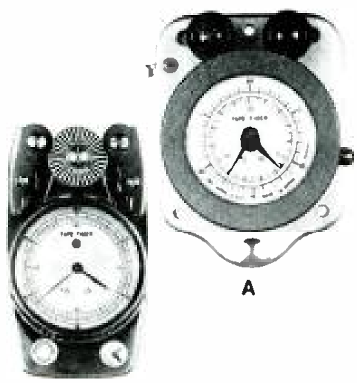

A

B

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may be required if they do not already exist. Presumably, there is not yet any provision for recording onto the multi-track machine, so the first step is to provide for Record inputs. (A) Probably the most practical method is via a jack field terminated by whatever type of plug your recorder requires.

With this done, you are immediately in business. To record on say, track 8, the synthesizer output is plugged into track 8 on the jack field. Outputs from previously recorded tracks, if any, are put in sel-sync position, and are monitored through the console. To hear track 8, the selector switch on the tape machine is placed on input and the mixer assigned to track 8 is raised.

As long as no additional processing of the synthesizer output is required, this arrangement is sufficient. However, if one wants to equalize, limit, or otherwise modify the synthesizer output before recording, a second jack field (B) or auxiliary console inputs (C) will be required, plus provision for diverting one of the console outputs. Again, a patch point (D), on the right output for example, is required.

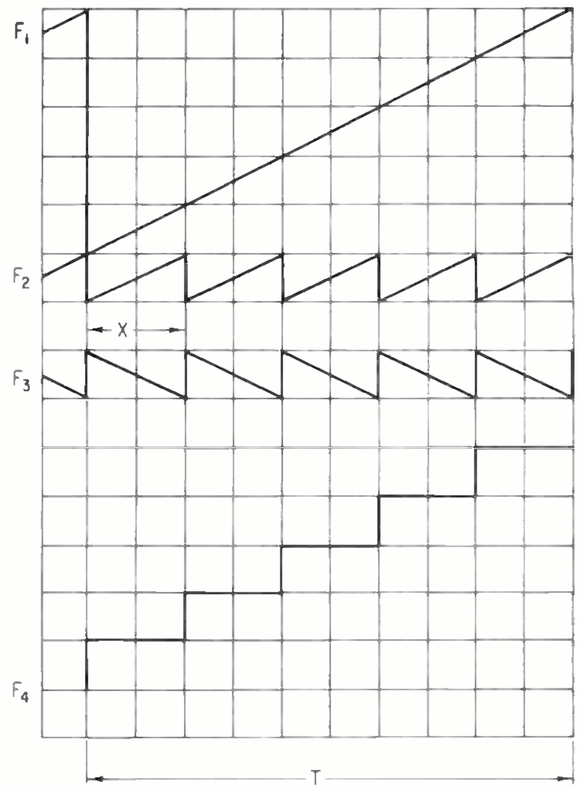

Figure 2. Constructing a step voltage using sawtooth outputs from two oscillators.

Now, the synthesizer output is patched into the track 8 input at (B) or into an auxiliary input at (C). This input is panned (or switched, depending on your console) to the right. All previously-recorded material on tracks 1 through 7 *et al* is put on the left. You may now process the synthesizer's output with whatever facilities are available at your console and pick up the right console output at (D) and feed it into the multi-track input 8 on the (A) patchfield. If you are using an aux input, the mixer assigned to track 8 must be off while recording to avoid confusion. For playback the mixer must of course be raised. If the 8 mixer itself was used as the input from the synthesizer, this automatically lifts the regular 8 tape output, so that the patch must be removed every time you want to listen to a playback.

Figure 1 also shows that the left side of the console is used to monitor previously-recorded material while the right side records the synthesizer. If there is leakage from the left side (tch-tch) it may be minimized by keeping these mixers relatively low and raising the left monitor system accordingly.

With these few modifications, even the most complex electronic-music projects are possible. Eventually, if it turns out to be worth the expense, sophisticated switching circuits may be built in to eliminate the need for patching, however for most applications this will probably not be necessary.

Naturally, the more extensive synthesizer systems offer the most flexibility. However, with imaginative use, even the smaller units can produce a wide variety of sounds. A popular accessory in the complete electronic music studio is a sequencer. But until one can be afforded, don't overlook some of the simpler sequences that can be set up on a fairly small synthesizer. Since most synthesizers are voltage controlled, a simple sequence can be set up with a step voltage.

How does one construct a step voltage? It's easy, if you have an oscilloscope handy. For example, on a Moog Synthesizer, tune two adjacent oscillators (sawtooth outputs) to very low frequencies, F_1 and F_2, where $F_2 = XF$ (See Figure 2). The amplitudes should be 1 volt and $\frac{1\ \text{volt}}{X}$, respectively. F_2 should be inverted (F_3) and combined with F_1 to produce step voltage F_4. This waveform can then be used as a frequency control voltage to vary the frequency of another oscillator, which will now sweep through an octave per T seconds in a series of X discrete steps. The slightest variation in any of these parameters will significantly alter the sound of the sequence.

This is just one of the simpler set-ups possible with a basic synthesizer in an average tape-mastering room. From here, one can progress as far as need, budget, and imagination permits. ■

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Circle 22 on Reader Service Card

Theory and Practice

NORMAN H. CROWHURST

● The deceptively simple circuit evolved at the end of the last discussion in this column certainly works very well, when your output load is a resistor whose value is the theoretical nominal, and with plenty of dissipation rating. But we really need amplifiers to feed loudspeakers and similar devices, which makes matters slightly different—enough to create some more problems.

Without going any further, loudspeakers come in a variety of impedances, which tube amplifiers met by providing different output tappings, usually, 4, 8 and 16 ohms. Professional amplifiers may have had a few extras. Some of the early transistor amplifiers catered for the same thing by designing the circuit to provide its maximum power into 8 ohms, and "cutting their losses" for the other impedances.

Of course, no harm comes to an amplifier designed to feed an 8-ohm load when a 16-ohm speaker is connected to it instead: it just delivers half as much current, at the same output voltage, which is half the power it delivers to 8 ohms. But connecting it to a 4-ohm load can cause trouble: now it wants to deliver twice the current at the same output voltage, and that is twice what it is designed to do.

The early remedy for this was to use a high-wattage 4-ohm resistor in series with the 4-ohm output terminals (FIGURE 1). If the impedances were always connected to the correct terminals, this looked after the output transistors. If the maximum power into 8 ohms was 50 watts, the amplifier would only deliver 25 watts into 16 ohms and, with a 4-ohm load connected to the 4-ohm terminals, it would deliver 50 watts, of which only 25 would be delivered to the output load, the other 25 going into the fat internal resistor.

The real trouble is that people do not always know what impedances mean, or if they do, they may not know what impedances they have. So a fairly inevitable process is that someone tries connecting speakers to different output terminals, to see which ones give the loudest sound.

With the old tube amplifiers, they might or might not land up on the correct terminals that way, but if they

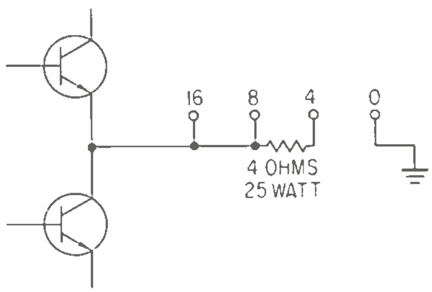

Figure 1. A method of providing multiple-impedance matching used on some earlier transistorized amplifiers.

STEREO REVIEW, JAN. 1969

(Advertisement)

EQUIPMENT TEST REPORTS

By Hirsch-Houck Laboratories

REVOX A77 TAPE RECORDER

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

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

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

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

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

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

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

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

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

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

The Revox A77 is housed in a teak cabinet with a fold-away carrying handle. It is one of the handsomest, as well as best-performing, tape recorders we have seen. We have never seen a recorder that could match the performance of the Revox A77 in all respects, and very few that even come close. It sounds as good as it tests, which speaks for itself. The Revox A77 is offered in a variety of configurations. It is available with either half- or quarter-track heads, in either the teak cabinet or a portable carrying case. The price of the deck in a wood base is $569; the deck with built-in power amplifiers is $599.

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didn't, no serious harm was done. But with this circuit, connecting a 4-ohm speaker where it would give the loudest sound, which would be to the 8- or 16-ohm terminals (which are actually the same connection, internally), can result in twice the maximum output current: bang goes the output transistor(s).

We could complicate this by talking about how loudspeaker impedances vary with frequency, and what the nominal impedance really means, relative to the impedance curve of the thing, and we shall later. However, this is trouble enough to show designers that something needs safeguarding, or someone is going to have to pay for replacing some fairly costly output transistors.

The simplest form of protection, and one that was used by some amplifier makers for a while, is a fuse in the output lead designed to blow before the maximum current rating of the transistor(s) is exceeded. This protects the transistors all right, but can be a nuisance. Even if a replacement fuse is handy, replacing it takes time, which can be aggravating.

A circuit breaker to operate on audio current is hardly feasible, without using up some of the output power, at least. But there is one other way (at least) an example of which is shown in FIGURE 2. In each output transistor lead a resistor is inserted that will reach slightly more than the contact potential for a silicon transistor, when maximum output current is reached.

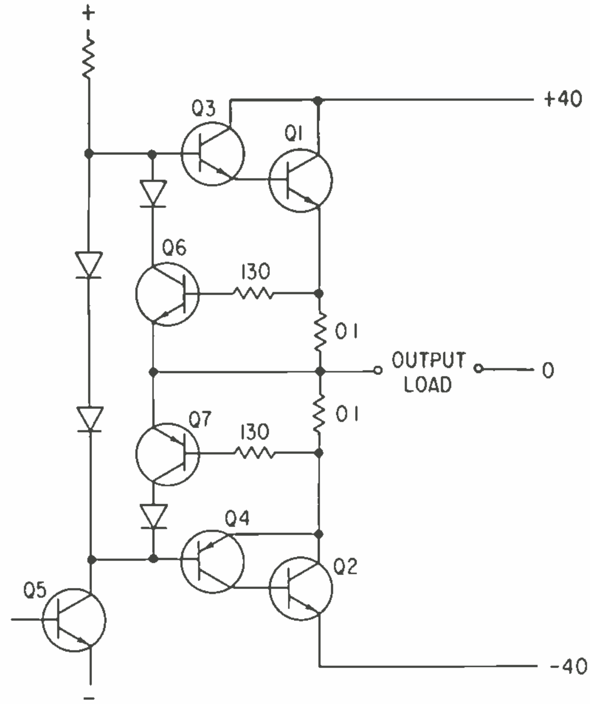

Figure 2. Adding over-current protection to the series push-pull output circuit shown in last month's issue.

Assume the supplies are plus and minus 40 volts, and 10 amps current is the maximum the output transistors can handle safely. A drop of 1 volt will only lose 2.5 per cent of the available output voltage, which is not an unreasonable sacrifice. A resistor of 0.1 ohms will produce a drop of 1 volt at 10 amps.

This voltage is used to trigger a couple of shunt transistors to bypass, by saturation, the base inputs of the drive stage. Output transistors have a gain of a little over 10, and drive transistors capable of this much drive current have a gain of about 30, so current gain for the two stages will be about 300. In an actual design, this would be measured.

So the drive stage bases need about 30 milliamps peak input. The shunting transistors should be capable of shunting away at least another 30 milliamps. Probably 100 milliamps would be a good figure, allowing for protection against a potential 4 times overload. At this collector current, as a saturation value, a gain of 30 would be reasonable (again depending on actual transistor type, which should be measured).

So the current available at 1 volt drop across the 0.1 ohm, assuming the contact potential of the transistor is 0.6 volt, making a difference of 0.4 volts, should be about 3 milliamps. This current and voltage combination dictates about 130 ohms for the series resistors.

In practice, values can be juggled until proper current protection is achieved. But this provides only one form of protection. If the load impedance is only a little too low, and in the form of a pure resistance, this protection is fine, but what if it is much too low, or highly reactive? Then a variety of other things can happen.

If the impedance is only a little too low in value, the current would only be a little over maximum when full voltage would be reached (the transistor then behaving as a short-circuit). Conversely, with current limiting in action, maximum current will be reached a little before maximum voltage, and thus the voltage drop across the transistor during this limiting period will be small (FIGURE 3).

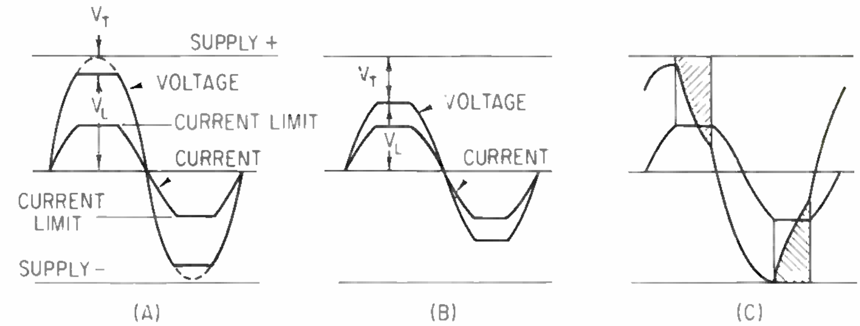

Figure 3. Why over-current protection is not enough. V_L is the voltage across the load and V_T that across the active transistor. At (A) the voltage drop during the current-limited period is small, because the load value is only a little below nominal. At (B) the voltage drop across the transistor is larger, because the load value is lower than at (A). At (C) the load value is correct, but reactive, so the voltage drop during the maximum current period is represented by the shaded areas; at one point almost the full supply voltage is across the transistor while it is at maximum current.

But suppose the impedance is much too low, or that it is reactive, so that maximum voltage does not correspond with maximum current (at the same time): then the voltage drop across the transistor can be much more, while this maximum current flows. This means that more than the current rating may be in trouble: maximum dissipation could be exceeded, too.

Dissipation is a function of voltage *and* current, both in the transistor. The transistor sustains somewhere approaching maximum voltage when it is cut off, and the other transistor is "doing the work," but that does not matter, because it is not carrying current. And in normal operation, maximum current may be reached when there is almost zero voltage across it, with no problems.

The dissipation is apt to be dangerous to the transistor when it reaches, or sustains maximum current, which can be controlled by this limiting action, and at the same time drops appreciable voltage. At this moment, with the kind of load impedance that would cause this combination, dropping voltage across the transistor could only be achieved by raising current, and *vice versa*. So there is only one thing to do: drop or stop the input signal, until the condition is corrected.

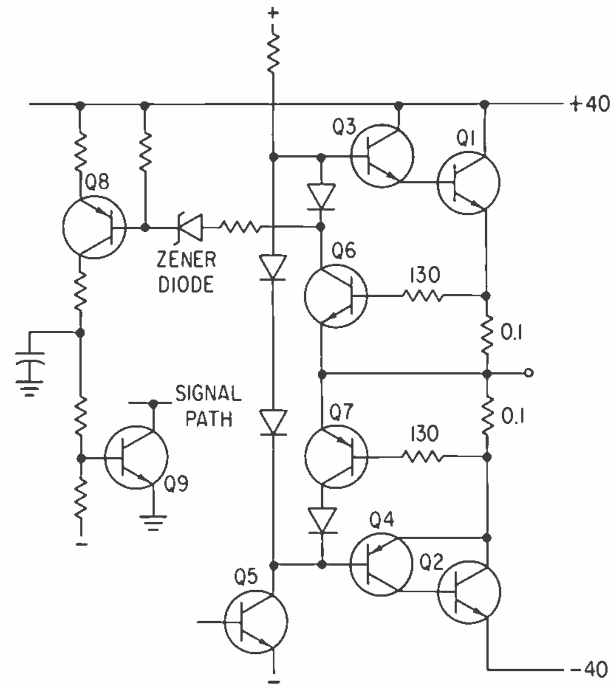

Figure 4. The beginnings of a circuit to protect against over-voltage, and thus over-dissipation, as well as over-current.

This is achieved by working out a circuit that checks voltage across the transistor, only when maximum current is reached. FIGURE 4 shows one way of doing this. When maximum current is not reached, transistor Q6 (or Q7) is not conducting at any time. One of these transistors only conducts when limiting takes place. The diode in series with its collector likewise only conducts then.

So, in the absence of current limiting, the zener diode has no voltage applied to it. But when current limiting occurs, the zener diode is put in parallel with the drive and output transistor voltage drop. If the voltage is not as much as its zener voltage, still nothing happens: transistor Q8 remains non-conducting all the time. But if, when current limiting occurs, the voltage across the drive and output transistor exceeds the zener voltage of the diode, Q8 receives base current, making it conduct and charge C1, which in turn saturates Q9.

Q9 can be placed so it short-circuits the input signal for long enough to allow things to correct themselves. When C1 discharges to the point where Q9 ceases to conduct, signal will again be admitted, and the circuit will try again. Most probably the dangerous condition will have passed. If not, it will again stop the input signal.

This circuit is an improvement over one with only current limiting. But it still has some problems left. It triggers regardless of how long the over-dissipation occurs for, either during a particular audio cycle, or as a sustained effect of several cycles. The output transistor mounting has a thermal capacity that will enable it to sustain an instantaneous over - dissipation, provided it is very short.

So it is difficult to set the voltage for complete safety, without having it too touchy for some purposes. If the voltage is that at which the transistor would not blow, if the dissipation is sustained, a quite safe, slightly higher, short duration dissipation will trip the circuit needlessly.

On the other hand, if some leeway is allowed for this, a sustained signal could blow it. The circuit needs some kind of intelligence to allow it to discriminate, so it will allow the short-burst higher dissipation, while tripping at a satisfactory value for a longer duration effect.

We will tackle that in the next issue, as well as dealing with ways to protect against over-voltage due to reactive loads. The basic design assumes a resistance load, with the transistor acting as a current modulator, and the voltage across the transistor, from collector to emitter and base, always in the same—the inherently non-conducting direction.

A reactive load can cause voltage kicks from the load to actually exceed supply voltage momentarily. When this happens, the output transistor's collector voltage is momentarily reversed, and this way the transistor conducts like a diode, with possible damage to itself or the drive transistor. Some way must be found to prevent this happening. ■

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Sound with Images

MARTIN DICKSTEIN

THE EYE

● The recent total solar eclipse reminded many of the times when, as youngsters, they were cautioned against looking up directly into the sun during the eclipse, but to use blackened film. This time, however, more had been learned about the eyes and the radiations from the sun, and the public was told not even to look at the sun with black film. In fact, people were warned by every medium of communication not to look at the sun directly at all, but to watch television or to look at the eclipse indirectly by using two pieces of of cardboard, the one closer to the sun with a tiny hole in it allowing the image of the sun to come through onto the farther piece.

An interesting paradox came to light during the recent gigantic visual presentation. Many people, not aware of how the eye works, or of the different solar radiations, thought it might be alright to look up at the eclipsed sun at the moment when the sun was completely hidden. However, the exact opposite is true. The moment of totality is the *worst* instant to look at the sun as that is the time when the eye could be most seriously damaged.

The eye operates as an automatic camera with a diaphragm (the iris), a lens (crystalline lens) and a film or screen (retina). The iris, controlled automatically by muscles, has a variable diameter opening (pupil) which regulates the amount of light which is allowed to fall on the retina. When the eye sees bright light, the diaphragm opening is small. When the light intensity decreases, the opening becomes larger, so the retina gets more light. During the moment of totality, the area within the umbra is almost in total darkness so the iris opens quite wide. However, instead of getting only light, the eye is also receiving infra-red rays which are radiated by the *corona*, that portion of the sun's outer atmosphere which is not blocked by the shadow of the moon. The infra-red burns the retina and causes permanent blind spots and the loss of ability to read and do fine seeing for close work. Incidentally, the retina is unaware that damage is being done because the retina is insensitive to pain.

The eye has the unique ability to focus on an object at any distance. This is accomplished by the changing of the relative curvature of the crystalline lens thus changing the focal length (unlike a camera). This feat, called accommodation, is automatic by muscular action and is more accurately and readily accomplished by the eyes of the young who can see objects as close as five or six inches from the eyes. Generally, the distance of ten inches is considered as the average distance of most distinct vision, for the normal eye.

As people get older, the muscles of the lens do not act as well and the reading distance grows longer until it seems that the arms are too short for comfortable perusal of anything but the headlines in the papers. This becomes evident, generally, in persons over the 40 mark and the situation does not improve with age but gets worse. Even telephone booths seem too small since the head can not go back far enough to make the dial numbers or the phone number come into focus, and when the numbers seem to be coming into focus they seem to dwindle even quicker into tiny indistinguishable dots. This loss of the power of accommodation is known as *presbyopia*.

Those who must keep objects relatively close in order to focus on them have the condition of myopia or nearsightedness which is correctable by glasses—lenses which compensate for the inoperation of the eye's focusing muscles. The opposite condition, farsightedness, is similarly corrected with lenses acting in the reverse direction to focus the image in the eye on the retinal surface. *Myopia* focuses an image in front of the retina while the image in the condition of *hypermetropia* is focused behind the retina. Lenses can then be computed to cause the focused image to fall directly on the retina when the object is held approximately 10 inches or 25 centimeters in front of the eyes.

The image of the object the eye is looking at is actually in sharp focus at the point where an imaginary axis through the lens crosses the retina.

At this point, minute cones are present and only cones. The areas of the retina proceeding away from the central focal point have less cones and more rods, and the farther away from the focal point, the lower is the ratio of cones to rods.

Through these cones and rods, the image is relayed to the optic nerve, which then transports the image to the brain. The actual image focused on the retina is upside down from the original object as is true with all lenses. However, from infancy, the human brain has learned to compensate for this situation and no further thought is given to the reversal. It becomes just another item on the long taken-for-granted list. It is interesting that at the point where the optic nerve connects to the retina, there is a blind spot. This is not noticeable under normal circumstances, but it can easily be found by marking a paper with two black dots about 8 cm. apart, holding the paper in front of the eyes at about the distance of most distinct vision and looking at the left dot with the right eye (with the left eye closed). The right dot will disappear.

As the eye travels over a page of copy (such as this one), the eye cannot read while in motion. It can only focus when it is at rest. The eyes, therefore, travel along a line in jerky movements and are guided along by six tiny muscles on each eye. Both eyes move in unison and after a full day's reading have traveled about a mile in approximately 100,000 steps. If the feet moved each time the eyes did, the distance covered would add up to about 50 miles. The eyes have focused on more than 100,000 words made up of between one-half and one-million letters. Any wonder they are tired? However, since man discovered printing a couple of centuries ago, he has learned to use this tool for education and pleasure in spite of the burden and hardship on the eyes. (It is this same capability of the eye to focus only when at rest that requires, conversely, that the images projected by a motion-picture projector momentarily stop at each frame, 16, 18, or 24 frames to a second, to fool the eye into thinking it is seeing a smoothly moving "movie").

The normal eye has fantastic adjustment capability for varying intensities of illumination. In the dark of night, the eye can see larger objects with illumination as low as 1 or 2 footcandles, and in broad daylight, the eye has to compensate for as much as 10,000 footcandles. In between these extremes, moderate office and factory tasks can take place in 10 to 20 footcandles, normal reading in 20-50 footcandles, proofreading, drafting, and watch repair in 50-100 footcandles. To indicate the normal illumination available, a house lamp provides about 40 footcandles at about 1 foot away; about 500 footcandles are available under a porch in the sun and about 1,000 footcandles in the shade of a tree. Interestingly, the eye (which finds difficulty in focusing at the normal reading distance in normal lighting) seems to improve in higher illumination, something people find out by moving the printed page or the needle they are trying to thread, closer to a high-wattage lamp.

When light becomes too bright for comfort, this is considered to be glare. Bright headlights at night, a match struck in the dark of a theater, appear to blind the observer. However, the same light under other ambient lighting conditions would not seem as severe. A bright slide image does not bother the viewer, but an overly bright motion picture will appear to have a flutter that can become objectionable.

The eye is a marvelous and complex device of which we only get two. They should be taken care of. A look at the eye and its workings, however, brings up other interesting subjects such as lenses, image brightness in relation to ambient lighting, and so on, which will be subjects for study in subsequent discussions. ■

Hearing Loss and Audio Engineering

W. DIXON WARD

A great deal of misinformation exists in regard to the effects of loud sound and hearing loss. The author, an expert in this subject cuts through the mystique to present the state of the knowledge as it exists today.

W. Dixon Ward is Professor, Department of Otolarngology and also in the Department of Speech Science, Pathology, and Audiology — a joint appointment — at the University of Minnesota. He is a Fellow of the Acoustical Society of America, and the American Speech and Hearing Association. He also finds time to be a member of the executive council of the NAS-NRC-sponsored Committee on Hearing, Bioacoustics, and Biomechanics (CHABA) and chairman of its Subcommittee 7_2 damage-risk criteria.

NEARLY TWO-THOUSAND YEARS AGO, Pliny the Elder recorded for posterity in his *Natural History* the observation that inhabitants of a certain village near one of the cataracts of the Nile gradually became hard of hearing. Although deafness associated with metal crafts had doubtless been known many millenia earlier, this seems to be the earliest surviving reference to a problem that is of ever greater concern in our increasingly noisy society. The realization is gradually spreading that not only industrial noise but also certain noises of everyday life — power mowers, chain saws, firecrackers, outboard motors, and even rock-and-roll music — are all able to damage the ear more or less gradually. Since audio engineers are occasionally (or perhaps even frequently) exposed to rather high sound levels in the process of monitoring recordings, editing tapes, and listening for flaws in the final sales product, it might be worthwhile to discuss the possibility of hearing loss in the field of sound engineering.

Though the number of people partially deafened by industrial noise and gunfire has been steadily increasing, only since World War II has any serious attempt been made to determine just how much noise the average ear can stand. This happened because industrial hearing loss severe enough to cause difficulty in understanding ordinary conversation finally came to be regarded as an occupational disease (like silicosis in miners, or radiation poisoning in watch-dial painters) and employers were required to indemnify their workers for hearing losses caused by noise in their factories. Where previously, for example, the drop-forge operator had regarded the gradual onset of deafness as a necessary evil, perhaps even a badge of distinction attesting to his long experience in his trade, he now became aware that this hearing loss was not at all necessary and that he could sue his employer for the damage. Legal action always hastens social reform, and so beginning about 20 years ago, various limits for noise exposure (damage-risk criteria, or DRC) have been proposed.

RISK CRITERIA FROM INDUSTRIAL DATA

The attack on the problem of DRC has followed two general lines. The first consisted of gathering actual data on noise-induced hearing loss (NIHL) in industry, either by estimating its growth indirectly by means of a cross-sectional analysis of all the workers, both new and old, or by actually determining the rate of increase through a longitudinal study, that is, testing the same individuals over a period of years. The second and more recent approach was more indirect; in this case, data on temporary hearing losses (more often termed temporary threshold shifts, or TTS's) were used to estimate the noxiousness of noises concerned. Noises that produced equal TTS's were assumed to be equally dangerous, and obviously noises that produce *no* TTS cannot cause NIHL.

With the accumulation of data from cross-sectional and longitudinal studies of industries having relatively constant noise levels, a few generalizations began to appear. For example, it soon became clear that low-frequency noise (below 500 Hz) was less hazardous than high-frequency noise (1000 Hz and above), so even the first DRC specified tolerable 8-hour exposures to steady noise in terms of the over-all sound-pressure levels in specific octave bands. FIGURE 1 shows a collection of DRC for 8-hour exposures proposed between 1950 and 1963,[1] and the relatively lesser noxiousness of the low frequencies is clearly evident (except for a few DRC that simply ignored frequencies below 300 Hz).

In general, of course, these criteria did not pretend to try to protect *every* ear from *any* damage (except perhaps for the USSR criterion). Because of individual differences in susceptibility to hearing loss, one is always dealing with distributions of losses under the stated conditions. Usually the goal, if stated explicitly, was to prevent a handicapping loss (more than 25 dB of average loss at 500, 1000 and 2000 Hz) in perhaps 90 per cent of the workers. However, the disparity among many of the criteria illustrated in FIGURE 1 stems from different assumptions of how much loss is tolerable.

Unfortunately, the noises of industry all tend to be quite similar. There are not enough noises that differ radically in spectral distribution to allow us to determine in detail the relation between the SPL of a particular octave band and the resulting NIHL. In addition, only seldom can one find an industry whose noise has a regular cyclicity in order that we may infer the trading relation between time and intensity for workers whose exposures are interrupted or intermittent, a condition that certainly applies to exposures of audio engineers. The earliest DRC simply avoided the issue by assuming, quite without evidence either pro or con, that all that mattered was the total energy absorbed during the course of a day. Thus, an 8-hour exposure to 85-dB SPL was taken as equivalent to 4 hours at 88 dB, 2 hours at 91 dB, etc. Furthermore, a single 2-hour exposure was taken to be no more hazardous than, say, eight 15-minute exposures with an hour between successive noise bursts.

RISK CRITERIA FROM LABORATORY TTS DATA

On the other hand, the temporary effects of noises with different spectra and distributions in time could be studied in the laboratory relatively easily. Intensive study of TTS, therefore, had by 1963 established (1) that TTS, like NIHL, was more readily caused by high than by low frequencies; (2) that the relation between intensity and time for constant

TTS was curvilinear; and (3) that the TTS at the end of the day was much smaller if the exposure was intermittent instead of continuous. In 1966 a set of rather complicated DRC based on TTS data was published[2] by a special working group of the NAS-NRC Committee on Hearing, Bio-Acoustics and Bio-Mechanics (CHABA). In addition to the assumption that all noises that produced a certain TTS were equally dangerous, it was the consensus of this working group that it was more important to protect hearing for the so-called "speech frequencies" (below 2000 Hz) than for higher frequencies. Thus we somewhat arbitrarily decided that we would try to limit the average TTS_2 (TTS measured 2 minutes after exposure) to 10 dB at 1000 Hz and below, 15 dB at 2000 Hz, and 20 dB at 4000 Hz and above. From latoratory data on TTS from steady and intermittent noise, a set of graphs were derived, summarizing the present knowledge of the exposures that would just produce the criterion TTS_2. The solid lines in FIGURE 2 portray one of these: the DRC for a single steady exposure as a function of octave-band SPL and frequency, with duration the parameter. It is clear that the shape of these equal-TTS contours changes as a function of duration, so that one cannot describe the relations among level, duration and frequency by any simple mathematical formula. The other eight graphs in the CHABA DRC were equally complicated; they provided limits for exposure to the four most important octave bands of noise (from 500 to 4000 Hz) when the exposure was intermittent.

Because of the difficulty soon experienced by safety engineers who had to use these graphs, several suggestions for simplification were made. The similarity of the DRC curves for 8-hour steady exposure (both in FIGURE 1 and 2) to the weighting factor for the A scale of sound-level meters (the dashed curve in FIGURE 2) suggested that perhaps an octave-band analysis of the noise might not even be necessary, provided that measurements were made in dBA.

Since the A weighting of the sound-level meter was deliberately designed in an attempt to compensate for the fact that not all frequencies affect the ear equally, this correspondence is not really surprising. Actually, certain investigators had argued for quite a few years that the relative noxiousness of a noise was given by its dBA value and had therefore made all their measurements in dBA. W. L. Baughn, for example, showed in 1966 that noises of less than 80 dBA produced no loss whatsoever in workers.[3] That is, in a group of men employed in 78 dBA, the number of persons with handicapping losses after 20 years of exposure was no greater than one would find in a non-noise-exposed group of men of comparable age and socio-economic status. On the other hand, data on men who worked in higher levels implied that at about 95 dBA, the incidence of handicapping loss would be about twice as great as in a comparable non-noise-exposed group.

In addition, the use of a constant trading relation between time and intensity was so much simpler than the eight graphs of the CHABA DRC that it was felt that the saving in effort was worth the loss in accuracy. Since it was evident by this time that a trading relation of 3 dB per doubling time was grossly over-conservative, especially for intermittent exposures, and that 6 dB per doubling time erred in the opposite direction, the use of some intermediate value was suggested.

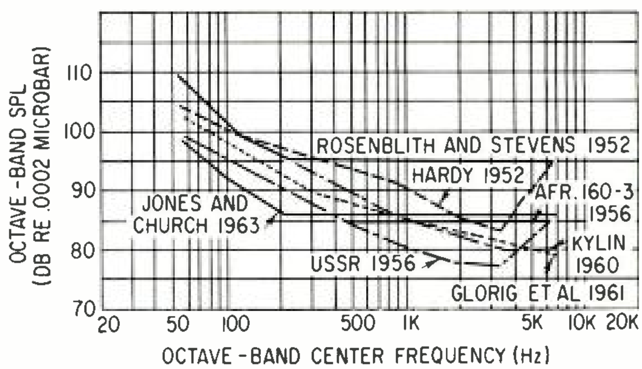

Figure 1. Seven different early damage-risk criteria for continuous 8-hour daily exposures (from Action[1]).

THE WALSH-HEALEY DRC

Thus was the stage set for the development in 1969 of what is rapidly becoming the most well-known of the modern DRC: the exposure limits specified by the Walsh-Healey Public Contracts Act.[4] Fundamentally, this DRC is as simple as one could possibly wish for: an 8-hour daily exposure limit of 90 dBA. If, as is usually the case, the maximum energy is distributed over several octaves, this 8-hr exposure does not differ materially from the 85-dB-per-octave-band limit of several of the earlier criteria shown in FIGURE 1 or the CHABA DRC of FIGURE 2. Only when the noise happens to have all its energy in a single octave band in the 1-4-kHz region will the Walsh-Healey (WH) DRC allow a 5-dB greater level than the others.

For exposures other than 8 continuous hours, two alternatives are given. If only dBA readings are available, then a simple 5-dB-per-doubling-time trading relation is used. That is, 4 hours of 95 dBA, 2 hours of 100 dBA, 1 hour of 105 dBA, 30 minutes of 110 dBA, and 15 minutes of 115 dBA are allowed. When the noise varies in level, then the total

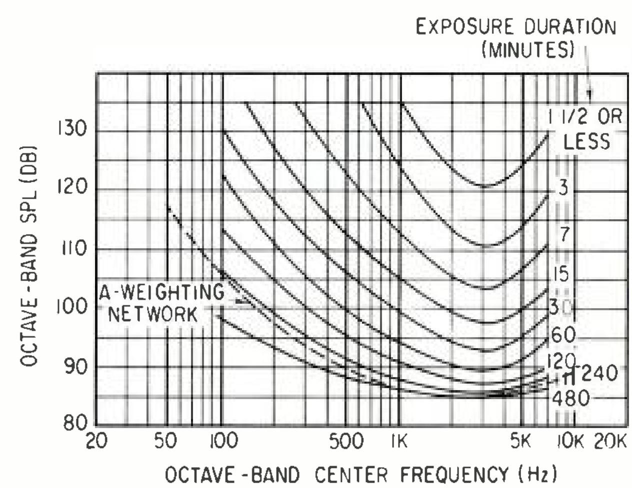

Figure 2. Damage-risk criteria as a function of spectral distribution of energy, according to CHABA[2]. The parameter is the daily tolerable single exposure, expressed in minutes. The A-weighting characteristic of a standard sound-level meter is indicated by the dashed line.

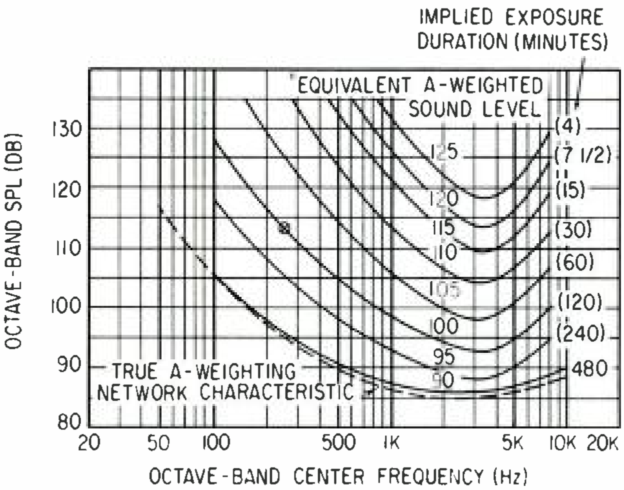

Figure 3. Suggested exposure limits for noise when octave-band levels are known, as recommended by the Walsh-Healey Act[4]. Note the similarity of the shape of the curves to those in Figure 2, although the durations permitted by the two sets of curves are quite different.

equivalent exposure must not exceed 8 hours of 90 dBA; for example, 4 hours of 90 dBA plus 30 minutes of 105 dBA would be tolerated, or 2 hours of 90 dBA (one-fourth the daily limit) plus 15 minutes of 105 dBA (again, one-fourth) plus 15 minutes of 110 dBA (one-half). The WH DRC makes no distinction between a single noise burst of 30 minutes and 10 3-minute bursts spread throughout the day.

Now in regard particularly to the relative effect of low-frequency noises, the strict dBA criteria just outlined tend to be over-conservative. Therefore the founders of the WH DRC provided a chart to be used to determine something they call "equivalent dBA" when a full octave-band analysis of the noise is available. This set of curves, whose contours are remarkably like the CHABA DRC of FIGURE 2, is shown in FIGURE 3. One plots the octave-band levels of the noise in question on this chart, and the highest penetration of that spectrum into the labelled curves would determine the "equivalent A-weighted sound level" to which the 5-dB-per-doubling-time trading relation would be applied.

I hope it is clear that when one uses the first option described above — that is, relying only on the dBA values actually measured and applying the 5-dB-per-doubling-time rule — then the procedure is equivalent to using a set of curves similar to those in FIGURE 3 but in which the curves are all parallel to the "true A-weighting network" and are separated at all frequencies by 5 dB. The point can be graphically demonstrated by considering a hypothetical noise that has nearly all its energy in the octave band centered at 250 Hz, and whose true sound pressure level is 114 dB re 0.0002 microbar. Because the A-weighting network discriminates against noise elements in this range by about 9 dB, the dBA value read on the meter will be $114 \times 9 = 105$ dBA, so that by the first method above only 1 hr of the noise would be tolerable. However, FIGURE 3 indicates that a 114-dB octave-band SPL at 250 Hz (point X on the chart) falls right on the contour marked 100, so that this noise has an equivalent level of 100 dBA, hence 2 hours of exposure are permitted.

The astute reader will note that in either case the allowable exposure is greater than that prescribed by the CHABA curves; FIGURE 2 indicates that only 30 minutes of 114 dB at 250 Hz is tolerable. On the other hand, however, if the noise were presented in bursts, the CHABA criteria for *intermittent* exposure—not shown here—indicate that this particular noise should be tolerable for a full 8 hours, provided that 7-minute bursts were separated by about 8 minutes of rest—which would give a total *cumulative* exposure of some 3½ hours in a day! This fact makes it clear that for all levels except 90 dBA, the WH DRC are for neither a single continuous exposure nor one interrupted in such a fashion that the greatest amount of energy can be tolerated, but rather something in between—perhaps they had some "typical interruption rate" in mind.

PRESENT DRC ARE INADEQUATE

If you are all by now thoroughly confused, I have achieved my goal. I could compound the confusion by citing German, French and British proposals (all of which differ from ours), but I think my point is clear. The fact of the matter is that we really do not know what a reasonable DRC should be; because of a lack of sufficient data, DRC such as the WH are more political than scientific in character. All one can be sure of is that whenever octave-band levels exceed 75 dB SPL, or the A-weighted value exceeds 80 dBA, some eventual loss after many years of exposure in the most susceptible individuals is possible. The exact amount by which the level that is permitted should increase as a function of (a) shortening the exposure from 8 hr, or (b) making the exposure intermittent, is still to be determined, as is the precise risk associated with continuous exposures at higher levels.

However, even if these *were* known, application of such curves and correction factors to the problem of the sound engineer would probably be extremely tedious, since no two days of exposure are apt to be alike. Therefore, while it does no harm to measure a typical day's exposure (if a typical day can be found) and consult either the CHABA or the WH criteria, a more direct method of estimating whether or not a particular exposure of a particular person is dangerous seems to be indicated. Instead of measuring the noise, one can measure the TTS it actually produces.

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(to be concluded next month)

Phono Disc Compatibility

SIDNEY FELDMAN

Ten years after the introduction of the stereo disc, the question of a compatible-stereo . mono disc is not yet fully resolved. This is where we currently stand.

WITH THE PHASING-OUT OF MONO records by most record companies, compatibility of stereo records becomes a major consideration again. There are many factors affecting the degree of compatibility of a stereo phonograph record. Will the record reproduce properly on *all* mono phonographs, or only on *some* machines? How will it sound?

How do we define compatibility? What does the stereo FM broadcaster do about stereo records that don't have the proper musical balance when heard by the mono FM listener? Do we want to make stereo records that are compatible? Is it possible to do so?

These are some of the topics that were discussed at a recent meeting of the N. Y. Section of the Audio Engineering Society. The meeting was held at the studios of RCA Records, and various effects were demonstrated from tape and disc' This article is based on the remarks made at this meeting, with additional material supplied by the author.

Participants in the panel discussion were John Eargle, then of RCA Records and now with Mercury Records; Floyd Harvey of Bell Telephone Labs; and Howard Holzer of Holzer Audio Engineering Co. The writer was chairmen.

Almost ten years ago, Pickwick Record Company released a series of stereo lp's which they claimed were compatible. What was then meant by a "compatible-stereo record"? Looking beyond the advertising agencies copy, we find that the term meant a stereo record that reproduced optimally in mono, with an esthetically pleasing sound balance, and no sacrifice in stereo quality. Also, the wear on the record should not be excessive when played with a mono pick-up. Several dozens of compatible-stereo records were wear tested, at that time, on automatic turntables equipped with counters, and all the popular cartridges available were utilized for the test. After 100 plays, it was concluded that the compatible-stereo cuts held up as well under the mono cartridges, as they did with the stereo pick-ups, except for one or two very-old-design mono cartridges which had high vertical and lateral impedance. These latter pick-ups ruined *both* the stereo and compatible-stereo pressings. So the compatible stereo record was launched by a small independent record company, but was not looked upon with much favor by the industry. There was talk of "compromise", "degradation of quality", and much harsher language at an A.E.S. meeting held in New York City on February 24, 1960. At this meeting various musical selections were played in mono, stereo, and compatible stereo. (Perhaps this was the first A-B-C test as distinguished from an A-B test?) Individual copies of this 12-inch pressing were given to *everyone* in attendance at this meeting, for further study at home.

In the intervening years, much sophisticated testing and evaluation has taken place, and the interested reader should refer to the literature on the subject. The early concept of a compatible stereo record was that the grooves had to be capable of being tracked properly by all mono pick-ups. If the mono cartridge would track the compatible-stereo cut satisfactorily, *i.e.* without distortion, skipping. or excessive wear, then the record buyer would not have to worry about purchasing stereo records even if he did not have any stereo playback equipment at that time. He could update his equipment to stereo later on, and know that his collection of stereo records would still be usable. For the record manufacturer there would be definite economic advantages if he would only have to produce one record—the compatible-stereo record, rather than the stereo and mono versions of all new releases, as was the common practice at that time.

The first compatible stereo records were mastered with a Fairchild stereo cutting system which had some signal

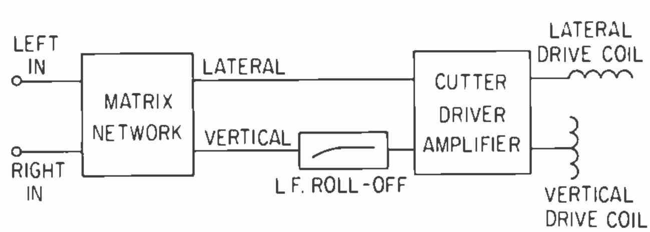
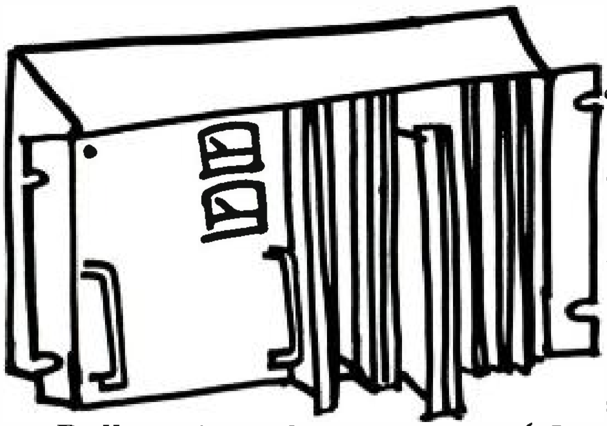


Figure 1. The Fairchild lateral/vertical stereo disc cutting system showing the signal processing in the vertical channel.

Sidney Feldman is chief engineer of Mastertone Recording Studios in New York City.

Down with dB's!



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

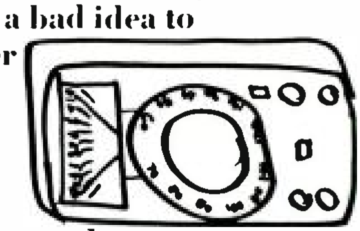
And to help fight the battle against those evil dB's, we now carry a rather large arsenal of noise-reduction equipment.

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

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

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

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



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

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

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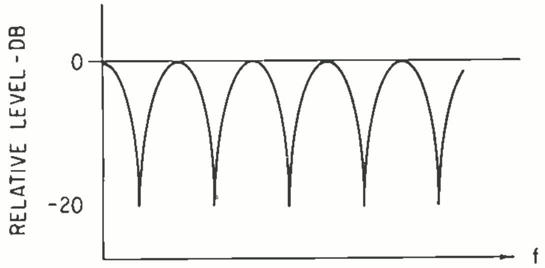

Figure 2. The comb-filter effect due to widely-spaced microphones. The resulting amplitude of the signal in mono shows the missing portions of the audio spectrum.

processing: a switch-selected low-frequency roll-off in the vertical, or difference signal, channel. (See FIGURE 1.) As the Fairchild stereo cutter was of the lateral-vertical type, it was an elegant idea to incorporate a low-frequency roll-off in the vertical channel, so that the degree of l.f. out of phase component could be controlled. Many times, though, poorly-recorded stereo tapes were run thru this system and cut *compatible*, with no particular attention to how the record sounded in mono. This may have been one of the contributing factors to the poor acceptance of the buying public, for many years, of this type of record. As time went on, various small record companies were producing compatible-stereo records with excellent results and good response from their customers, but these companies were producing the tapes from start to finish under controlled conditions, with a compatible-stereo record as their objective.

As more of the record buying public updated its equipment, it became evident that the difficulty of producing a compatible-stereo record was not in the ability of the cartridge to track the record with acceptable wear characteristics, but how the record would sound when played on a mono system. Many mono systems were being sold with stereo cartridges already in place, so that the conversion to stereo could be made quickly, when the customer was ready. The main concern now, in producing a compatible-stereo record is that the spectrum of sound add properly in mono, so that on a stereo FM radio station, the listener to the program in mono will receive an esthetically acceptable musical balance. As many record companies are releasing in stereo only, the AM radio station has a problem, too, as does the record buyer remaining with mono equipment.

There are many techniques that can be utilized in the recording studio to produce a compatible-stereo record, and there are some practices that should be avoided, as note the following panelists:

John Eargle: The marketplace for records now is stereo only, except for the stereo single which *must* be playable on the "$19.95 special." The stereo single requires special treatment: the grooves must be kept as deep as possible, to ensure trackability. A number of factors determine trackability, and certain techniques can be utilized to this end: below a certain bleed frequency, keep the low-frequency energy in the center channel, or in-phase. This will avoid excessive modulation in the vertical direction, which would lead to thin grooves and difficulty in tracking. Also, the level, as a rule, must be kept lower than in a comparable mono single, to avoid excessively thin grooves and high modulation velocities, which can introduce tracking difficulties.

When listening to a stereo FM broadcast, in mono, we get L + R. The L − R sub-carrier has the stereo information, which is lost to the mono listener. It is important to realize that what makes good stereo can also make excellent mono, good mono, or poor mono, depending on what choices and decisions were made by the producer and engineer at the original recording and re-mix sessions, and what was done in transferring the mixed tape to the master lacquer.

It is possible to enjoy stereo with mis-aligned tape heads, gap scatter, out-of-phase signals, etc. But when listening in *mono*, to a stereo tape, these all introduce problems which affect the sound. Gap scatter produces nulls which cancel part of the audible spectrum. We hear swishing or comb filter effect on piano records that were made in stereo with excessively wide spacing between microphones. (FIGURE 2) There is no way to correct for mis-aligned heads on a tape recorder, except to go back to the earlier mixes and make corrections—adjust the equipment and re-mix.

Another problem encountered when going to mono from a tape that was mixed for stereo only, is that of the center-channel build-up. Howard Holzer produces a device that by means of a quadrature split (phase-shifting network) can be utilized to reduce the center-channel buildup when reducing a stereo tape to mono. By introducing a 90-degree phase shift in one channel of a two-channel stereo signal, the center channel buildup of information can be reduced by 3 dB. This is the amount of buildup obtained when a coherent signal is fed to both left and right channels, in-phase and equal amplitudes. This effect is utilized to great benefit in stereo recording when you want an instrument or vocalist to be more prominent—that signal source is panned to the middle. But in mono we may find that the vocalist or instrument is too loud for a pleasing balance. The quadrature shift will take care of this problem when mixing down to mono from that same stereo tape. If the producer and engineer check the balance both in stereo and mono, when recording and mixing, this difficulty can usually be avoided. A satisfactory balance or mix will be obtained for both the stereo and mono listener.

When mixing from multi-track tape to stereo, there is a tendency to play the speakers too loud in the control room. This gives a false sense of balance between mono and stereo mixes. When the same mix is played at a lower volume, the balance will not sound the same. Almost everything seems to sound better at a high monitor level, and this leads to a false sense of security.

The M/S (mid/side) system of recording yields the least difference between mono and stereo mixes. This technique of recording is rarely used in the U. S. as it lends itself best to producing two-track stereo originals, and most of our present-day recording is being done on 8-track or 16-track machines, and then reduced to two-track.

Howard Holzer: We have found two problems in attempting to make a compatible-stereo record: *Tracking*. You must eliminate vertical lifts, otherwise portions of the groove are missing and you will not track properly. The low-frequency cross-over device will eliminate this problem. Below the cross-over frequency the groove being cut is essentially mono. This produces a groove that is easier to track with both mono and stereo pick-ups. You must sacrifice some of the stereo effect in the original recording to have it playable on mono cartridges.

Esthetic considerations. The record should sound the same in mono as it does in stereo. The Holzer CSG unit provides a means to achieve a mono mix, from a stereo tape, that will sound the same as the stereo tape, and is the best way to mix from stereo to mono. No claim is made as to the degree of trackability of the stereo groove with a mono pick-up, using the CSG device, as the nature of the operation of the quadrature split is such that the vertical component is increased, and the lateral component may be reduced in amplitude.

If AM and mono-FM radio stations were to play all records with a stereo pick-up, they would eliminate the problem of trackability and wear, but we still would have to consider the esthetic balance of the mono mix produced from the stereo tape.

Floyd Harvey: The telephone company is concerned with certain aspects of the transmission of stereo signals from one place to another, be it intra-city, inter-city, across the continent, and now between continents. Land lines, satellite, and cable are used for these transmissions. The problem is how much phase shift can be permitted between a pair of communications channels when used for stereo program transmission. Telephone lines need to be good to within 10-microseconds phase shift between pairs of lines in order to transmit a stereo program without losing the stereo effect. When a stereo FM broadcasting station orders a pair of telephone lines from their studio to transmitter, they have to be concerned with the quality of the lines obtained. If there is excessive phase shift between lines, stereo separation is reduced, and the program may become mono by the time it reaches the transmitter!

A question and answer period followed, with much interest being expressed in the problem of the stereo FM broadcaster: they are unable to check all new records for balance in mono, and may play records that do not balance properly for the mono listener.

A compatible-stereo record has definite economic advantages: the record company only has to produce one version, with the attendant lowering of production costs, and the record dealer simplifies his inventory system. As far as the listener is concerned, a compatible-stereo record does not have to provide any compromises if the record company will take the responsibility, through the use of good engineering practices, to produce a stereo record that also sounds good in mono.

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A Disc Recording Seminar

LARRY ZIDE

What follows is a first-hand report on a company-sponsored seminar that explored the practical methods to achieving better disc recordings in the studio.

LAST FALL Gotham Audio Corporation sponsored a seminar on disc recording held in the Sterling Forest Conference Center complex. This is a country setting some convenient 55 miles outside of New York City.

Gotham Audio is, of course, interested deeply in disc recording. They are distributors of Neumann lathes and equipment and much ancillary merchandise. But the seminar was not designed to sell Gotham's products; it was designed, and it succeeded, as an exchange-of-information discussion by, and for, professionals involved in disc recording.

A total of 42 people journeyed from all over the world (Japan, South Africa, Mexico, and Canada were represented —as well as the U.S.). The form of the seminar was a workshop conducted by Stephen Temmer, president of Gotham, with guest speakers. These were John Eargle, then quality control manager at RCA Records (now chief engineer of Mercury Records); and Lee Hulko, president of Sterling Sound, Inc., a New York City-based disc-mastering studio.

Most of the discussions centered on the very concrete problems of daily operation, with theory kept at the minimum required for proper understanding of the processes involved. The emphasis was on the most modern techniques of disc mastering.

I visited the conference on the middle of the three days it lasted. Gotham had set up a model mastering lab in the conference room. The interest of the listeners was obviously keen — and they were prepared to dispute anything that did not sound right to them. But mostly they listened. In my questions to several, there was no doubt left that they were receiving tremendous value from the seminar.

It was a relaxed (due to the fine fall weather and the absolute beauty of Sterling Forest) but busy seminar.

On the next page are a group of photos made during the seminar.

The Sterling Forest Conference Center. Located on a lake (behind the camera position), the center proved ideal for the seminar. The center also includes living facilities

In an informal session, Stephen Temmer, Gotham Audio's president, talked about disc mastering and the way he sees it as the distributor of Neumann mastering systems.

Between the scheduled formal sessions, participants closely examined the Neumann system that had been set up (and operational) in the conference room.

Lee Hulko, president of Sterling Sound, an outfit devoted entirely to disc mastering, speaks about his practical experiences.

In this group, John Eargle, now chief engineer of Mercury Records, but then with RCA Records, explained what happens when a record is cut. As you can see, it was both a lecture and an exchange-of-information session.

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● The MKS-10 head assembly permits cassette - drive reverse systems. By means of a lever, the assembly is swung to the proper position for either set of stereo tracks. In addition, guides used as part of the assembly permit other openings in the cassette to be used so that an erase function is available in either direction of travel. The entire system is rigidly mounted to eliminate the need for service or oem alignment.
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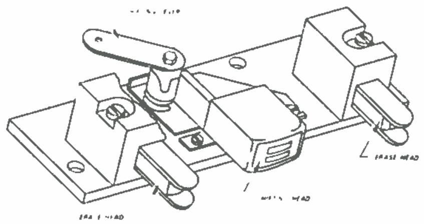
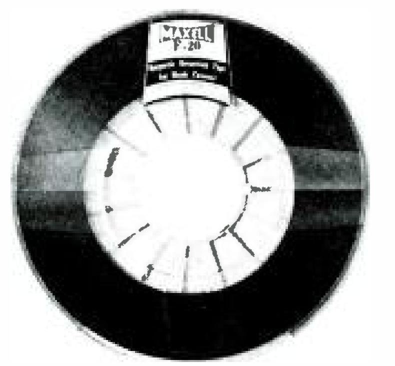


CASSETTE TAPE

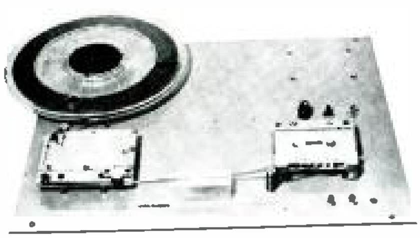
● A new-design magnetic cassette tape is indicated for both recording and duplication uses, particularly where high-frequency application is important. The frequency range of the tape is 25 Hz to 18,000 Hz. A 10 per cent increase in tensile strength over conventional tapes is claimed. Detailed information is available.
Mfr.: Maxell Corp. (Hitachi, Ltd.)
Circle 57 on Reader Service Card



DROPOUT COUNTER

● Used in conjunction with any standard audio recorder the model 521 dropout counting unit instruments the batch testing or cirtification of magnetic audio recording tape for a quantitative measure of instantaneous amplitude variations (dropouts) occurring in a given sample. The unit is of particular value to all slow-speed recording operations where minor tape oxide inconsistencies result in audible dropouts.
Mfr.: GRT Corporation
Circle 54 on Reader Service Card

CASSETTE EDITOR



● Here is a high-speed production-type cassette editor that loads recorded tape from a bulk supply reel to a pre-leadered cassette. Automatic electronic sensing of the model MN1001 and high-speed dynamic brakes provide instantaneous stoping action precisely locating the tape for cutting. The tape runs over a vacuum splicing block which is automatically activated when the brakes are applied so the tape is held ready for splicing.
Mfr.: Island Magnetics Electronics, Corp.
Circle 53 on Reader Service Card

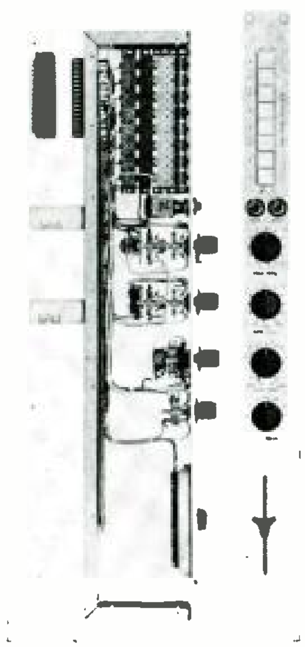
CASSETTE DUPLICATOR

● A new reel-to-cassette duplicator has been announced that is stated to meet professional standards and still be priced within range of even the smaller commercial duplicators. Model 235CS consists of an open-reel master transport and cassette slave modules. Each slave module contains three cassette transports. Both master and slave feature two-speed hysteresis-synchronous motor drives with the master operating at 15 and $7\frac{1}{2}$ in./sec., and the slaves at $7\frac{1}{2}$ and $3\frac{3}{4}$ in./sec. Wide-band solid-state electronics are of modular plug-in design. Up to six or nine slaves can be accommodated by one master without additional electronics. The systems will be available in half-track single and dual channel, quarter-track dual channel, as well as quarter-track four channel (simultaneous) configurations.
Mfr.: Telex Communications Division
Price: (half-track two-channel with three slaves) $2070
Circle 58 on Reader Service Card

VARIABLE-SPEED TAPE TRANSPORT

● Specialized audio applications will welcome this variable-speed drive system which is capable of precise, variable speed from a standstill to 120 in./sec. Discrete steps at standard speeds are also available. The unit has independent reel servos and constant tape tension in all modes. The direct-drive capstan servo can be controlled from its own oscillator or from an external source. A rewind speed of 500 in./sec. gives the instrument excellent search capability. Up to 10½-in. standard NAB reels are accommodated. Of course, full professional standard tape capability also exists.
Mfr.: Magnetic Recording Systems
Circle 52 on Reader Service Card

MIC MODULE KIT

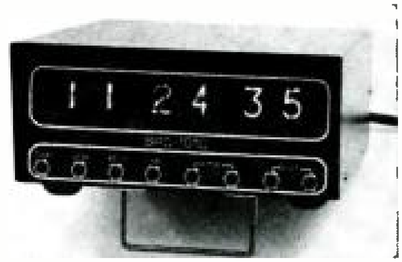


● Model 201 is a microphone input module kit 1¾-in. wide by 24-in. long. Eight push-button output assign positions (push on, push off), momentary solo and mute pushbuttons, high- and low-frequency shelf equalization with up to 12 dB of boost or cut at each, echo send with six pre and post assign positions, input level select attenuation, are among the included features. Octal plug-in amplifiers are also available to increase the versatility of the strip. These include a mic preamp, equalization amplifier, and line amplifier.
Mfr.: Opamp Labs.
Price: $200 without octal amplifiers
Circle 55 on Reader Service Card

MASTERING TAPE

● A new music mastering tape with greater dynamic range, increased tape durability, and fewer handling problems is promised in Scotch brand 206-207 tapes. An over-all increase of 3 dB in dynamic range over Dynarange tapes is stated. The base is polyester and uses a new binder formulation to hold the oxide. Special back treatment of the tape resists windowing, cinching, and scratching even when roughly handled as in shipping. In addition, the tape's backing surface is electrically conductive, greatly reducing the static attraction of dirt and debris, minimizing contaminant-caused dropouts associated with extended use of tape. Scotch 206 is 1½-mil in thickness while 207 is 1 mil. The tape is available in ¼; ½; 1; and 2-inch widths.
Mfr.: 3M Company
Circle 59 on Reader Service Card

DIGITAL CLOCK



● Solid-state design is used to provide an accurate readout clock that may be used for normal clock, stop watch, or elapsed-time indication. The six-digit display gives, hours, minutes, and seconds. All operational controls are on the front panel, including start-stop and reset. The standard case is 7½ by 3½ inches, although a rack-mount is available. Options for tenth-second display, rear-projection, remote function, battery operation, etc. are available.
Mfr.: Broadcast Products Co., Inc.
Price: $395.00
Circle 50 on Reader Service Card

SPECTRUM SHAPING

● Filter set 8056A consists of twenty-four 1/3rd-octave filters connected in parallel. Each filter channel has an individually adjustable attenuator that lets the user set the instrument's response to each 1/3-octave band of frequencies. Center frequencies range from 50 Hz to 10 kHz. Optional sets of filters, each spanning a 200:1 frequency range down to 2 Hz or up to 40 kHz are available. The filters have flat characteristics in the passband, but with steep skirts. Peak-to-valley ripple in the passband is less than 0.5 dB; response to adjacent-channel center frequencies is 20 dB down; and center frequencies two channels away are 50 dB down. Filter responses overlap at the 3 dB points. Each attenuator is adjustable 40 dB (+20 to − 20 dB).
Mfr.: Hewlett-Packard
Price: $2025
Circle 56 on Reader Service Card

(from page 2)

amplifier noise is not raised by raising source impedance. This is not the case in Mr. Smith's preamp. Let us look briefly at the kind of amplifying devices available and how they are affected by source impedance. In the audio-frequency range the noise produced by a tube is not affected by source impedance, and the same holds for a field-effect transistor. Hence one makes the source impedance, and hence input voltage, as high as practical. At rf frequencies capacitive-based noise currents cause best noise figures to be obtained at finite source impedances. The ordinary transistor at audio frequencies is entirely different. In the common emitter mode, best noise figures are obtained by using a source impedance of several hundred ohms. If one is interested in verifying this, one need only look in *Design of Low Noise Transistor Input Circuits* by William A. Rheinfelder (Hayden Book Company, New York). The UTC microphone transformer, used in all Gates preamplifiers, has a 385-ohm secondary, a correct application of the transistor in the common-emitter mode. Mr. Smith's amplifier does not use the common-emitter mode; the circuity is arranged such that best noise figures are obtained by using the lowest possible impedance, and thus the lowest possible voltage. His implication that he can obtain significantly better signal-to-noise ratio than with high impedance pick-up is not true, of course. His signal voltage is lowered by the same number of dB as the noise; he must use more gain and the noise will be right back where he started from, or within a few dB. Mr. Odom's statement that noise will be overcome by using high source impedance is in this case wrong. In his reply, Mr. Smith appears to reverse his position on the part of the issue in which he was correct. I find this inexplicable. In summary, to obtain the best noise figure when considering amplifier noise, one must use the proper source impedance for the device. Even when this is done some devices will produce more noise then others, on the order of several dB. Whoever can find the lowest noise device and then use it properly wins the low-noise game. Only properly applied precise measurements can determine this.

There is more to be said for the high-pedance pick-up, however. The output voltage will be the sum of the voltages induced in each turn. As long as we do not wind on so many turns that we are losing a significant amount of flux in the outer ones, the voltage will be the same for each turn. Thus output voltage increases linearly with the number of turns. However, we must be careful not to put on so many turns that we lower the resonant frequency enough to cause high frequency loss. Let us consider operating the pick-up into an infinite impedance, noiseless, preamplifier. Noise voltage will be produced only by the resistance of the coil (let us ignore any possible effects from quantum mechanics, such as quantized electro-magnetic fields). Signal voltage increases linearly with the resistance, keeping within the limitations given above. Noise voltage increases only with the square root of the resistance. Thus noise from the pick-up itself is least significant with the highest impedance. The Stanton 500 cartridge, a *high-impedance* model, has a *resistance*, which is what contributes the noise, of 800 ohms. Using a bandwidth of 20 kHz, the noise is 0.51 microvolts. Since the pick-up has an output of about 3 millivolts, we have an ideal signal-to-noise ratio of approximately 76 dB. Let us consider a pickup with a 2-ohm resistance and the same flux density from the magnet structure. Noise voltage will be about 0.02 microvolts. Signal voltage will be 0.01 millivolts, or about 54 dB signal-to-noise. Of course the Ortofon SL-15 may have a considerably stronger magnet than the Stanton. Mr. Odom will have to use a very sensitive pick-up to obtain the 80 dB he claims. Mr. Smith's Ortofon uses a moving coil; very high flux densities may be obtained, but the pick-up must be low impedance. I have not seen the pick-up specs; it may be possible for him to obtain 80 dB also. In summary, if one is given a source of some impedance which one does not wish to transform, one must use the proper device in the preamplifier to obtain the best signal-to-noise ratio. Mr. Smith has done this; however this is all he has done. He has not improved signal-to-noise ratio as he claims. The ability to run long leads between pick-up and preamp is of very little practical importance.

Until someone produces a record with much better that 70 dB signal-to-noise ratio this whole issue is of little practical importance. The usual studio tape recorder is limited to 68 dB. Record surface noise certainly cannot stay below this after a short exposure to the atmosphere. It appears to me that existing broadcast preamps and ones found in good hi-fidelity systems are good enough.

Michael P. Sulzer
Chief Engineer
WESU AM-FM
Wesleyan Broadcast Assoc. Inc.
Middletown, Conn.

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People, Places, Happenings

● **Marvin Headrick** has joined **Quad-Eight Electronics** in the position of marketing manager. He has been active in the professional audio industry since 1946. He was with ABC a.m. radio and t.v. operations from 1949 to 1956. After that, until early last year, he was a manufacturers representative selling key professional audio lines. In late 1968 he was appointed national audio sales manager of **Langevin** where he was until this present move.

● The **CBS Electronic Video Recording System (EVR)** continues to expand into practical ways. CBS has announced EVR sales offices in Montreal according to **Robert E. Brockway** president of the CBS EVR division. He commented that "So much interest in EVR has been generated in Canadian industry and education circles that we have found it mandatory to locate in Canada." CBS recently successfully demonstrated their color EVR system, with expectations of both hardware and software to be available later this year.

● **Harold "Bud" Jackson** has been appointed to the position of product manager, consumer sound tape for **Audio Devices.** In the announcement by company v.p. **William Goldstein,** it was stated that Mr. Jackson has been with Audio Devices for over five years, most recently as district manager of the Dallas, Texas area.

● Americans staying at Tokyo's New Otani Hotel during Expo '70 will see English language movies on Japanese commercial t.v. with the original sound tracks, even though Japanese audiences will hear dubbed audio. Multiplexed t.v.-sound tracks is the method used by the **Japan Broadcasting Corporation.** A simple adapter is used to pick out the English track. Normal sets without adaptation will only receive the standard Japanese track. Japan Broadcasting is working extensively with the system and sees application in educational broadcasting, guide commentary, and (you guessed it) stereo transmission.

● **Christopher F. Coburn** has been named to the newly-created post of executive vice-president of **GRT Corporation.** In this position he will be responsible for all daily operations of the corporation, including marketing, engineering, and manufacturing functions. He will be located at corporate headquarters in Sunnyvale California. He has been with GRT for a year and a half in executive marketing positions.

● **I. R. Stern** has been named executive vice-president of **James B. Lansing Sound,** according to a recent announcement by **William H. Thomas,** president of the company. Mr. Stern comes to JBL from an active career as a representative on the west coast for a number of high-fidelity manufacturers.

● **Richard O. Ware** has joined **Superscope, Inc.** as general sales manager of its recorded tape division. In the announcement by **Fred C. Tushinsky,** v-p sales and marketing, it was stated that Mr. Ware will be in charge of sales and new business contacts with major record companies who are potential customers for Superscope's custom tape duplicating service. He comes to the company with over ten years of experience in the tape duplicating field. He was most recently affiliated with the **3M Company,** Detroit, in various sales capacities.

● A new company has been formed to sell products and to service the broadcast industry. The company, **International Tapetronics Corporation** will have its product line announced shortly. Principals of the company include **Elmo Franklin, Jack Jenkins, Andy Rector,** and **Merle Wilson.** All are former executives of Gates Radio Companys, ATC Division at Bloomington, Illinois.

● From **Sonocraft** comes word that **Elie C. Katz** has been appointed as vice-president in charge of the video tape recorder division. From 1965 to the present position, he was with **ITV, Inc.** (which became **Riker Information Systems, Inc.**), in several managerial posts.

● **Jensen** loudspeakers have been selected for use in Boeing 747 aircraft. Both public-address and music system speaker arrangements are being used. As an example, model C12RS speakers are being placed above the ceiling in a baffle arrangement as a sectionally-controlled environmental system. Other models will be used as wall units. A total of 61 speakers will be used in some versions of the 747.

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The biggest changes in the RE55 are mechanical. For this microphone is even more rugged than the 655... long known as one of the toughest in the business. There's a solid steel case and new, improved internal shock mounting for the RE55. Plus a satin nickel finish that looks great on TV long after most microphones have been scarred and scratched almost beyond recognition.

For convenience we've made the barrel of the RE55 just 3/4″ in diameter. It fits modern 3/4″ accessories. It also fits the hand (and its length makes the RE55 perfect for hand-held interviews). We also provide XLR-3 Cannon-type connectors to help you standardize your audio wiring. Detail refinements that make the RE55 more dependable, easier to use.

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