THE DICTATING MACHINE - See Page 22

AUDIO ENGINEERING

AUGUST 1953
35c

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CONTENTS

AUGUST, 1953 Vol. 37, No. 8

COVER

George E. Sokolsky is one of the most prolific writers on the political life of our time in addition to being a radio commentator and newspaper columnist. Because of the press of work, he would ordinarily need at least ten secretaries to handle his voluminous daily output of words. A devotee of the dictating instrument for this reason, Mr. Sokolsky is shown with his new V. P. Edison Voicewriter.

RADIO MAGAZINES, INC., P. O. BOX 629, MINEOLA, N. Y.

AUDIO ENGINEERING (title registered U. S. Pat. Off.) is published monthly at 10 McGovern Avenue, Lancaster, Pa. by Radio Magazines, Inc., Henry A. Schober, President; C. G. McProud, Secretary. Executive and Editorial Offices: 204 Front St., Mineola, N. Y. Subscription rates—United States, U. S. Possessions and Canada, $3.00 for 1 year, $5.00 for 2 years; elsewhere $4.00 per year. Single copies 35c. Printed in U. S. A. Entered as Second Class Matter February 9, 1950, at the Post Office, Lancaster, Pa. under the Act of March 3, 1879.

AUDIO PATENTS

RICHARD H. DORF*

Tape recording is one of the most important developments of this audio age, but unfortunately it is not (theoretically, at least) as simple in principle as disc recording. For one thing, there are the problems of using recording heads with sufficiently low impedance to permit adequate bias current flow without making the impedance so low at the lower audio frequencies that no conceivable vacuum-tube plate circuit can operate satisfactorily.

The recording head can never be of particularly high impedance as long as the audio winding is also used to carry bias current, as is usually dictated by economy and magnetic circuit design. Given a constant-voltage audio source such as the usual triode, the low-frequency head current is huge compared to that at high frequencies, with corresponding variation in the flux left on the tape. The solution is to have a source which is of a constant-current nature—an essentially resistive head circuit.

There are two usual ways of doing the job. One is to employ triode output tubes and insert a relatively high resistance in series with the tube and the head. This tends to keep the circuit (including head) current constant and works very satisfactorily. It is highly inefficient, however, because of the large audio drop through the series resistor, making only a small portion of the amplified power available.

The second way is to use a pentode tube to feed the recording head, since a pentode is essentially a constant-current generator due to its high internal impedance. Unfortunately, the distortion of a pentode in this service—especially considering the low impedance of the head, which is its load—is far too high for good-quality recording.

* *255 W. 84th St., New York 24, N. Y.*

Myron J. Stolaroff of Redwood City, Calif., discloses a new circuit for using a pentode with very little distortion in his patent No. 2,634,335, assigned to Ampex. One form of it is diagrammed in *Fig.* 1.

As in the usual tape recorder, the microphone is fed to a preamplifier, the output of which goes to the grid of a pentode, a 6AC7 in this illustration. R_1 is the usual grid-leak resistor and R_2 is a combination cathode-bias and negative-feedback resistor.

The load for the tube is a current transformer T with a turns ratio of about 5.5:1. Note that the lower end of the primary goes through choke L_2 to the B-supply and for a.c. is connected to the cathode of the tube through blocking capacitor C_1. The secondary is connected to the head through antiresonant circuit C_4–L_1, tuned to the frequency of the ultrasonic bias to prevent bias current from flowing through the transformer secondary. The secondary circuit is completed through the recording head to ground for a.c. through blocking capacitor C_2. A voltage divider R_3–R_4 across the high-voltage supply adds a small amount of direct current bias to the a.c. bias from the high-frequency oscillator. This does not really constitute part of the invention but the author has found it useful. C_3 is the h.f. oscillator output blocking capacitor.

The lower end of the transformer secondary is connected to the tube cathode rather than to ground. Thus head current passes through R_2 in phase opposition to the current through R_2 due to the tube plate-screen current. This is a negative feedback connection to reduce distortion. Its special value is that it accounts for magnetic-circuit non-linearities as well as distortion produced in the tube.

Now let us see how the circuit performs.

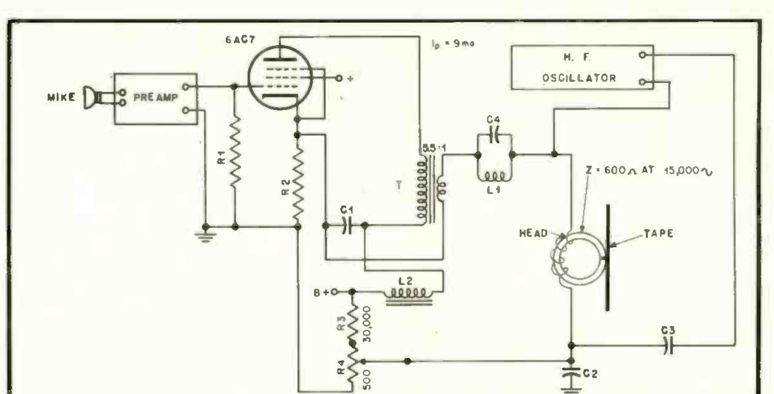


Fig. 1.

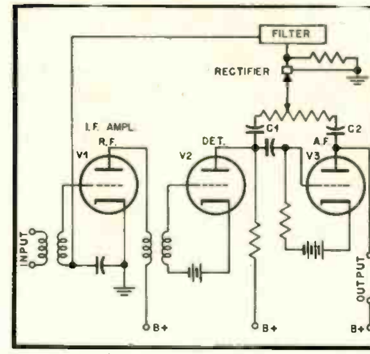


Fig. 2.

The 6AC7 has a plate resistance of about 0.75 megohms and a mutual conductance of 9000 micromhos. The recording head impedance reaches a maximum of 600 ohms at 15,000 cps; this is reflected back to the primary as a little over 18,000 ohms which, added vectorially to the 0.75-meg tube plate resistance, makes a total change too small to notice. Since the net output-circuit impedance therefore remains essentially constant, the head current also remains constant.

The value of R_2 shown gives a plate current of 9 ma, about normal for the 6AC7. The transformer secondary-circuit current flowing through R_2 in phase opposition to the plate current reduces the distortion due to both tube and head by a factor of

$$\frac{1}{1+G_m R_c r}$$

where G_m is the transconductance of the tube in mhos, R_c is R_2, and r is the transformer stepdown ratio. The value of the fraction is 1/12.9 or 7.8 per cent of the initial tube distortion. If, we figure that the initial distortion of the 6AC7 with such a low-impedance load would be about 20 per cent at full output, the total distortion with the feedback circuit is reduced to 7.8 per cent of that or 1.56 per cent, a value easily tolerated, especially since full output is rarely approached.[1]

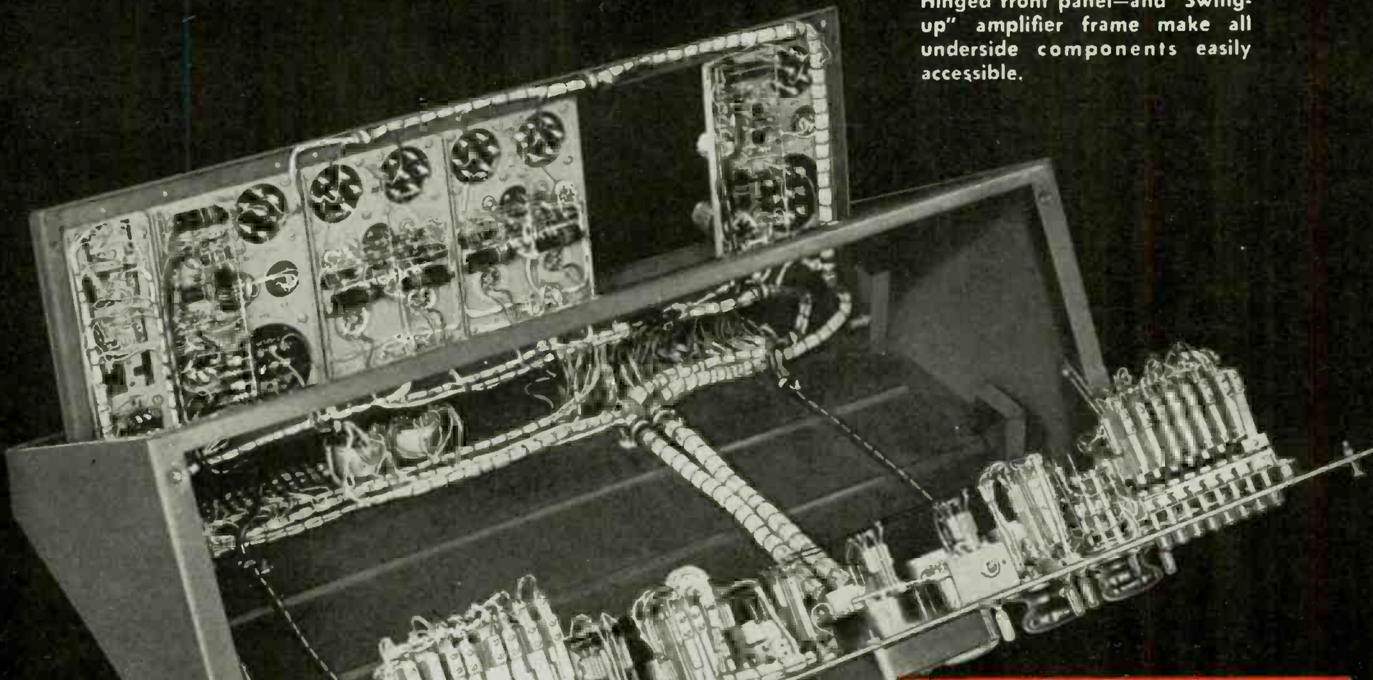
Automatic Distortion Control

George A. Bonadio of Watertown, N. Y., is the inventor of what he calls an automatic distortion control. While there is some question in my mind as to its practical value in very many cases, it is ingenious and simple, making it a good candidate for inclusion in this section of Æ. The patent number is 2,634,339 and one way of using the invention is diagrammed in *Fig. 2*.

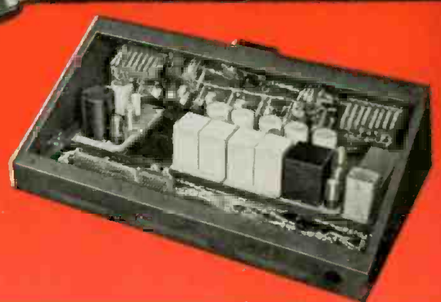
The invention is premised on the fact that even a well designed amplifier will distort if overloaded by too high an input signal. It controls distortion by preventing the signal from getting that high.

In *Fig. 2*, V_1 is a standard i.f. amplifier stage of a receiver, V_2 the detector, and V_3 the audio amplifier. R_1 is connected between the input and output of the audio stage through blocking capacitors C_1 and C_2. Its value is high enough to prevent its having any effect on the stage itself—perhaps 1 to 10 megohms (unless it is also to be used as

[1] The inventor gives values for the fraction, the distortion reduction, and the final distortion of 1/13.9, 7.2 per cent, and 1.44 per cent. One of us is less than a genius, arithmetically speaking. I am taking a chance on my own version.



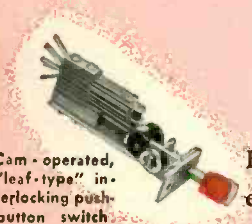
Hinged front panel—and "Swing-up" amplifier frame make all underside components easily accessible.



Easily removed top cover provides quick access to tubes and external connections. The BC-2B will fit snug against the studio window.

NEW DESIGN consolette

Color coded for "error-proof" control



Cam-operated, "leaf-type" interlocking push-button switch

The new consolette BC-2B provides all the essential audio facilities needed by most AM, FM, and TV stations — plus many extra operating advantages not previously available in a standard consolette. It speeds up switching operations substantially over previous designs. It provides for complete control of all studio operations. *The BC-2B gives your station "deluxe" features at a "standard" price.*

Read the list of exclusive "extra" features the new BC-2B offers you. Then ask your RCA Broadcast Sales Representative for complete details. His service is as near as your phone.

11 extra features!

- *"Color-coded" controls quickly identify and tie related functions together.*
- *New, leaf-type cam-operated interlocking, push-button switches.*
- *New hinged front panel for easy access to switches, gain controls, and contacts.*
- *Amplifiers mounted on "swing-up" frame; chassis easy to remove.*
- *New 30° sloping top panel for maximum studio visibility—styling compatible with modern AM and TV practice.*
- *New compact amplifiers use low-noise, long-life, miniature tubes.*
- *Improved, faster-operating speaker relays eliminate key clicks and audio feedback.*
- *Lamp dimmer for VU meter (ideal for TV service).*
- *8 high-level mixing channels, separate gain controls for network and remote.*
- *Turntable mixers with "built-in" cueing switches.*
- *No clearance required at rear—can be installed up against walls and control room windows. Uses less desk space, too.*

RCA

RADIO CORPORATION of AMERICA
ENGINEERING PRODUCTS DEPARTMENT CAMDEN N.J.

a negative feedback resistor, which is extraneous to this story). Since, in the absence of stage distortion, the voltages at its ends with respect to ground have the same waveform in opposite phase, there is some point along the potentiometer (obviously to the left of center—electronically speaking) where the audio voltage with respect to ground will be zero.

At this point the arm of the potentiometer is set and connected to a rectifier and filter. The output of the filter is connected to the grid circuit of V_1.

When distortion is zero—or as low as when the setting was made—everything operates normally. But when the audio stage distorts, the waveforms at the two ends of R_1 are no longer similar and probably the levels are not as before. Therefore a voltage appears at the potentiometer tap. This voltage is rectified, filtered, and applied as negative bias to the i.f. amplifier grid, bringing the volume down. Obviously the level cannot be reduced to the point of the original low distortion since the would then be no control voltage; it must reach an equilibrium point. However, distortion can be kept quite close to the original value, especially if amplification is added somewhere between the tap on R_1 and the i.f. grid.

The same invention can be used on an audio amplifier if the amplifier has a volume expander or compressor. Then R_1 is run between a late stage (preferably the output stage) plate and an earlier stage with the proper phase relationship. The rectified voltage is applied in proper polarity to the expander or compressor to make volume reduce when distortion appears.

The reason I wondered about practicality is that few occasions arise for running sound systems at volumes so close to high-distortion points.

Patent Language

It is a relief to be able to record that the language of patent specifications has been gradually coming closer and closer to reasonable engineering English. While it is necessary to be more legalistic in the language than, for instance, an Æ article, there is little excuse for a phrase that speaks of circuits "in parallelism," for instance. This appeared in a patent of my own and I regret to say I was too weak-kneed to talk my attorney out of it. It is only a mild sample, however, of some of the nonsense that has cluttered up patent specifications and made them hard to understand.

The worst offenders are foreign inventors, who are subjected to translators who sometimes appear to have turned up from the Elizabethan era (the first one, that is) with the aid of a time machine. Not only are constructions problems in cryptology, but the choice of words brings to mind a picture of the translator poring over a *larousse* and selecting the largest unfamiliar word he can find.

There is a patent on my desk at the moment, for example, showing how the relative outputs of four loudspeakers are controlled by a device with objects duplicating in miniature the positions of the speakers. "Under these conditions," it says, "a correspondence which is substantially near to a homology is obtained. . . ." I will not attempt to translate. Words are constantly used in this and many other foreign patents with correct dictionary meanings but entirely erroneous connotations.

One of the essential ingredients of good engineering is maximum possible simplicity. Patents are records of engineering work. Even though attorneys are usually responsible for their form, it seems to me that poor, vague, and intricate construction reflect on the reputation of the patentee.

Here is the first basic advance in tone arm design in many decades!

The GRAY "Viscous-Damped" 108-B Tone Arm

Gives you perfect contact and tracking on all records at lowest stylus pressure—virtually eliminates tone arm resonances—cannot damage record if accidentally dropped.

The entirely new suspension principle of the Gray 108-B makes it *hug the grooves . . .* prevents stylus skidding on worn records—overcomes groove-jumping caused by floor vibrations. Its "viscous-damped" design provides perfect tracking, virtually eliminates tone arm resonances, and prevents any possibility of record damage if the arm is dropped.

The 108-B satisfies every requirement of high fidelity reproduction. A plug-in feature permits instant change from 78-rpm to 33⅓-rpm or 45-rpm, *with automatic adjustment to the correct stylus pressure.* See and try this "viscous-damped" arm soon—solve all your transcription problems with this revolutionary, versatile arm!

GRAY RESEARCH

AND DEVELOPMENT CO., INC., HILLIARD STREET, MANCHESTER, CONN.
Division of The GRAY MANUFACTURING COMPANY—Originators of the Gray Telephone Pay Station and the Gray Audograph and PhonAudograph

Gray Research & Development Co., Inc.
Hilliard Street, Manchester, Conn.

Please send me your Bulletin RF-8 on the new Gray "Viscous-Damped" 108-B Tone Arm.

NAME

ADDRESS

CITY STATE

LETTERS

Mu Versus A

Sir:

In Chapter 12 Part 1 of Mr. Villchur's "Handbook of Sound Reproduction" the statement is made that the "effective source impedance is reduced upon application of negative voltage feedback by the same factor as the gain, namely $1/(1-A\beta)$." Fortunately, this is not the case. I say fortunately, since if this *were* the case, it would be considerably more difficult to make the average feedback beam power tube amplifier look like a triode amplifier. Actually the effective source impedance in a power output stage is reduced by the factor $1/(1-\mu\beta)$. Since the μ of a beam power tube is considerably greater than the gain, the source impedance is reduced much more than the gain is, The typical beam power amplifier to which 12 db of feedback is applied will have the gain reduced by a factor of four, while the source impedance is reduced by the factor of about forty or more.

If the initial effective source impedance is defined as the tube source impedance in parallel with the load, then application of feedback will reduce this effective source inpedance by the factor mentioned in Mr. Villchur's article. However, in power output stages this definition of source impedance is normally of no significance.

ROBERT M. MITCHELL
133-19 Blossom Ave.,
Flushing, N. Y.

Answers to the Q & A Question

Sir:

Your magazine is tops—keep it that way! Do not fill pages with stupid Q&A's, although a few interesting ones might be good.

I would like to see an article on connecting a tape recorder to a TV set. Also a method of changing a Philco 1000 to a sweep audio 'scope. I have found a small 'scope almost indispensable with a frequency sweep record in visualizing equalization. I propose that every complete audio outfit include such an important tool. Hum and high resonant frequencies can be spotted immediately and eliminated without the need for trained ears. Endless bands for tape machines can also be used. Those who are bothered by roll-off trouble should try tweeters which can be turned down in steps. They could mark the switch or pot with the type of record.

THOMAS M. WALTON
4583 G Street
Philadelphia, Pa.

Sir:

. . . Question and answer department would be very welcome. Please don't make it solid high-level engineering dope only, however.

H. R. Medland
Philadelphia, Pa.

Equipment Reports

Sir:

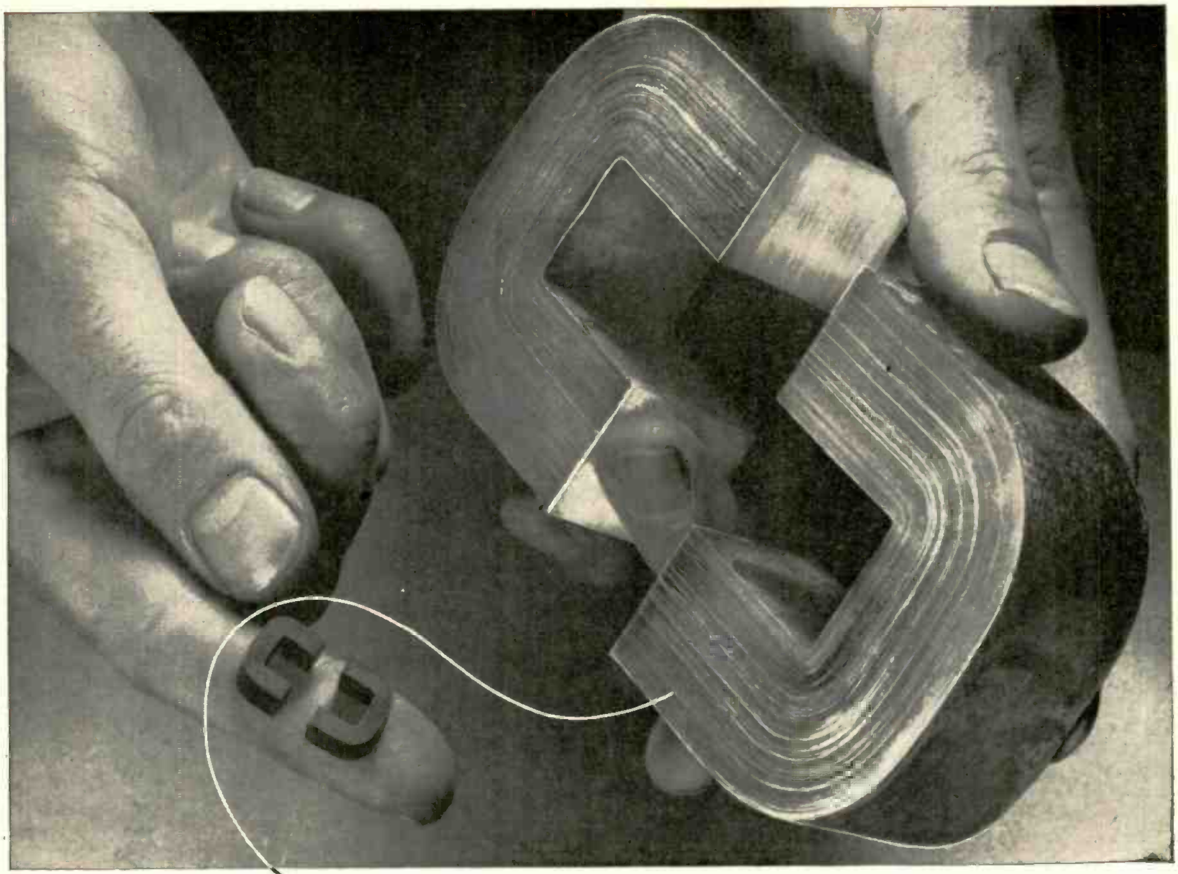
Your Equipment Report section is very interesting, but so far has never reviewed speakers or enclosures. Most audio men will agree that the weakest link in high-quality audio is the speaker/enclosure combination. Why the oversight?

Could you kindly give an opinion of how the .... speakers rate?

Incidentally, I think a Q&A department would be worthwhile.

B. ROTOLO
125 Prospect St.,
East Orange, N. J.

(Equipment Reports on amplifiers, tuners record players, etc. are based on readily measured performance factors such as frequency response, sensitivity, distortion, hum, and so on. Any report on speakers would be largely subjective unless very elaborate measuring equipment and a room especially designed for tests of that type were available. It is well known that two different makes of speakers—of equal quality—will sound different, so the user should actually make his own choice after listening to a number of speakers and enclosures, using the same source material and the same equipment throughout. Since what one person likes may not satisfy another, Æ has consistently recommended this method of selection. Similarly, Æ cannot rate speakers, but simply suggests that all speakers listed by their manufacturers as hi-fi models should be of good reliable construction. Hear them and make your own choice. ED.)



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...*any quantity and any size*

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For users operating on government schedules, Arnold is now producing C-Cores wound from ¼, ½, 1, 2, 4 and 12-mil Silectron strip. The ultra-thin oriented silicon steel strip is rolled to exacting tolerances in our own plant on precision cold-reducing equipment of the most modern type. Winding of cores, processing of butt joints, etc. are carefully controlled, assuring the lowest possible core losses, and freedom from short-circuiting of the laminations.

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NEW LITERATURE

• **National Wooden Box Association,** Barr Building, Washington 6, D. C., has performed a great service to American manufacturers who ship to overseas customers, with the preparation of a 20-page booklet titled "Export Packing." The publication opens with an analysis of goods to be shipped, continues with determination of box or crate specifications, and includes pertinent information on nailing, strapping, interior blocking, bracing, and marking procedures. A glossary of terms and a list of government requirements are included for reference. Printed in color and liberally illustrated, "Export Packing" is available to manufacturers through individual members of the box and crate industry or on written request to the Association.

• **Benjamin Electric Mfg. Co.,** Des Plaines, Ill., covers a variety of subjects interesting to audio engineers in a new data book and catalog titled, "Industrial Signal Guide Book." Devoted essentially to industrial sound signals, the book includes a simplified explanation of fundamentals of sound; a glossary of signal and sound terms; suggested steps in choosing a signal installation, and hints for compiling a coding signal call system. Various industrial signalling devices are illustrated and described. Available without charge.

• **Shure Brothers, Inc.,** 225 W. Huron St., Chicago 10, Ill., covers all of the company's products in newly-issued Catalog No. 44. Products listed include microphones, microphone parts and accessories, phono cartridges and accessories, wire and tape recording heads, as well as replacement data on all of these components.

• **The Indiana Steel Products Company,** Valparaiso, Ind., is now introducing "Applied Magnetics," a new technical publication designed for helpfulness to engineers who have interest in permanent magnets as a component of their company's products. Published bi-monthly, the magazine contains material designed primarily to aid engineers in the development of permanent magnets that are best suited for their particular applications. "Applied Magnetics" will be sent without charge to engineers, scientists, and others interested in the professional use of permanent magnets.

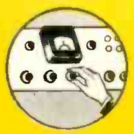
• **Thermosen, Inc.,** 361 W. Main St., Stamford, Conn., in a recently published 4-page folder, gives complete specifications for six temperature-limited diodes having stable emission characteristics and which are available as standard stock items. Three of the tubes incorporate a safety feature by which filament failure closes an external high-impedance circuit. Listing includes both electrical and mechanical data, also basing diagrams and basing designations. Copy available on request.

• **Tube Department of the RCA Victor Division, Radio Corporation of America,** has produced an educational picture booklet of photographs, cutaway drawings, and exploded views showing structural details of vacuum tubes used in both home and industry. Covered are tubes ranging from sub-miniatures to super-power beam triodes, including such diverse types as image orthicons and phototubes. Stated to be the most detailed and complete compilation of its kind, the booklet was prepared especially for use as a visual instruction aid in schools. It is printed on special 70-lb. coated paper to permit direct photographic reproduction of both illustrations and text. Containing 16 pages and priced at 25 cents, the booklet is obtainable from Commercial Engineering, RCA Tube Department, Harrison, N. J.

• **Centralab Division of Globe-Union, Inc.,** 900 E. Keefe Ave., Milwaukee, Wis., is now distributing Printed Electronic Circuit Guide Number 2, a completely revised and enhanced version of the original P. E. C. directory published earlier. Listed are 27 standard stock units furnished by Centralab through its distributors, as well as complete circuits, components, and applications. Requests for free copy of the P. E. C. guide should be directed to Department D-33.

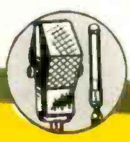
5 Reasons why E-V is STANDARD EQUIPMENT on TV and BC

HIGH SENSITIVITY



Very high flux density is utilized in E-V dynamic microphones for utmost sensitivity. The Model 650, for example, is the most sensitive wide-range microphone that has ever been built commercially.

LIGHT WEIGHT—SMALL SIZE



Comparison shows how much lighter in weight E-V units are than bulky, less sensitive microphones. The Model 655, for example, weighs only 11 ounces. Smaller microphone with matte gray finish is naturally unobtrusive and scarcely noticed if it gets into the picture.

RUGGED—INDESTRUCTIBLE

Use of the blast-proof, corrosion-proof E-V Acoustalloy diaphragm, enables these microphones to withstand the most severe shock without fear of internal damage. Even after immersion in water, operating characteristics are unaffected. This means long, trouble-free, *economical* service under all types of operating conditions.

WIDE-RANGE—STABLE

Because these dynamic microphones utilize the exclusive E-V Acoustalloy diaphragm, they are virtually flat across the entire reproduced range. Furthermore, the frontal cavity in each is used to load acoustically the diaphragm itself for extended high-frequency range.

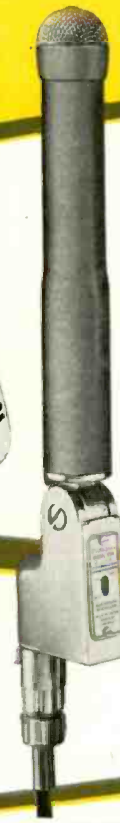
NO BLOOPS AND POPS

E-V Slim-Trim models have integral blast and wind shield. This feature eliminates bloops and pops from explosive labial sounds, insures clean quality throughout the program with the maximum of maneuverability, and solves critical placement. Location work outside with high ambient wind noise is made practical and easy.

MODEL 646 DYNAMIC
Small size. Response 40-14,000 cps, ± 2.5 db in high range, specially compensated for chest operation. Power rating —57 db. Omnidirectional. Integral blast filter. Choice of 50, 150 or 250 ohms. List, $140 Net, $84*

MODEL 655 DYNAMIC
Response 40-15,000 cps, ± 2 db. Power rating —55 db. Omnidirectional. Integral blast filter. 250 ohms imp. Easy change to 50 ohms. List, $200 Net, $120*

MODEL 650 DYNAMIC
Response 40-15,000 cps, ± 2 db. Power rating —48 db. Omnidirectional. 50-250 ohm impedance selector. Dual external shock mount. List, $150 Net, $90*



MODEL 654 DYNAMIC
Response 50-13,000 cps, ± 2.5 db. Power rating —55 db. Omnidirectional. 50-250 ohm impedance selector. Integral breath-blast filter. List, $95 Net, $57*

MODEL 635 DYNAMIC
Compact, rugged for hand, stand or boom. Response 60-13,000 cps, ± 2.5 db. Power rating —55 db. Omnidirectional. 50-250 ohm impedance selector. List, $75 Net $45*

E-V dynamic microphones are in daily use on network and local telecasts and broadcasts—in the studio and on remotes—indoors and outdoors. They give more in-service operating hours. Smooth, peak-free, wide-range response and wide acceptance angle enhance naturalness in the reproduction of voice and music. No closely associated auxiliary amplifier equipment is required. Each E-V microphone is guaranteed to be within its very close tolerance specifications. Standard connectors. TV gray and satin chromium finishes.

Write to E-V Broadcast Division for helpful information.

*Reference Level: 1 mw/10 dynes/cm²

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Export: 13 East 40th St., New York 16, U.S.A. Cables: Arlab

audiology

W. R. AYRES*

Optimum Loading for Power Amplifier

In preliminary amplifier development, graphical methods may be employed for power output prediction, and for the estimation of harmonic distortion. While tube characteristic curves are essential in basic vacuum-tube circuit design, optimum loading of a high-fidelity amplifier is often a practical compromise among varied considerations, some of which may be evaluated readily only by experimental means. But experiments, like calculations, must be carefully controlled as to content and interpretation, to usefully represent planned operation or production practice.

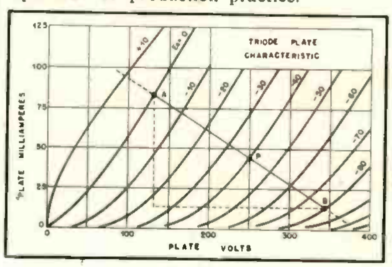

Fig. 1.

The Graphical Method

Amplifier loading sets the path of plate-signal operation as displayed on conventional plate-family characteristic curves. Here the operating point is the no-signal combination of plate-voltage, plate-current, and grid bias, and is shown as P in *Fig. 1*. In simple single-ended operation with resistive loading, the load-line is straight and passes through the operating point with a negative slope equal to the reciprocal of the load resistance. For greatest power output, point P is customarily at the highest plate voltage consistent with tube ratings and power supply capability, with idling plate current corresponding to rated plate dissipation. Occasionally, and possibly with higher distortion, the operating point must be lowered in either voltage or current to keep maximum-signal plate or screen dissipation within satisfactory limits.

Assuming no distortion, maximum power output without grid current is:

$$P = \frac{(E_{max} - E_{min})\ (I_{max} - I_{min})}{8} \text{ watts,}$$

where E_{min} and I_{max} are the instantaneous values of plate voltage and plate current at the intersection of the load line with the curve for $E_c = 0$, and E_{max} and I_{min} are the instantaneous values of plate voltage and plate current at the intersection of the load line with the curve for E_c equal to twice the bias at the operating point.

The load resistance to bring about this operation is:

$$R_b = \frac{(E_{max} - E_{min})}{(I_{max} - I_{min})} \text{ ohms.}$$

Power output is proportional to the area of the right triangle beneath hypotenuse AB. With common triode characteristics, this triangular area, and hence the power output, is greatest for a load-line slope about half that of the bias lines it crosses; i.e., for load resistance about twice the average plate resistance. For pentodes and beam-power types, point A may preferably be positioned near the knee of the zero-bias line.

With either triodes or pentodes, with or without feedback, lower impedance loads than these approximate values yield higher distortion and reduced maximum power output. Greater values of load impedance also give reduced maximum power output, but yield lower distortion at low levels.

In push-pull operation, because of the transformer connection, the output is due to the dynamic difference between the two plate currents. Though the transfer characteristic of each tube may be far from linear, the difference between the two currents, i.e., the composite current characteristic, may be quite straight. Operating bias is ideally chosen to minimize irregularities in the composite characteristic; however, this frequently requires such low bias that allowable plate dissipation would be exceeded. Therefore, the bias is selected as a compromise between distortion and zero-signal plate current.

At the time of peak positive grid signal on one tube of a push-pull stage, the other tube is cut off or nearly so, and a simplified construction may be used in predicting power output with a given load resistance. The method is to draw on the plate family a load-line having a slope of $-1/4R_L$, through the plate voltage axis at plate-cathode supply voltage. Calling this potential E_b, and the intersection of the load-line with the zero-bias line (E_{min}, I_{max}), the maximum signal power output is:

$$P = \frac{(E_b - E_{min})\ (I_{max})}{2} \text{ watts,}$$

under the assumption that the output waveform is negligibly distorted.

The assumption of negligible distortion is particularly valid in feedback amplifiers, in which with original sinusoidal input, the power amplifier grid signal waveform is modified by feedback as required to produce sinusoidal output voltage. Thus in low distortion push-pull feedback amplifiers,

* *RCA Victor, Camden, N. J.*

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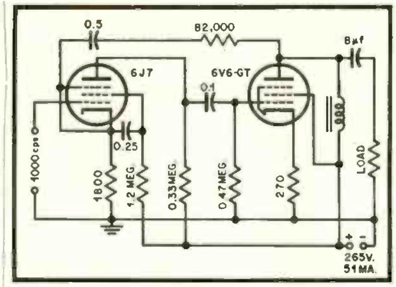

Fig. 2.

power output at the grid-current point is affected only slightly by the exact choice of operating point.

The Experimental Method

If either low values of distortion are involved, or if high accuracy is required, distortion prediction by graphical methods is usually impractical. One reason is that the accuracy with which published curves are drawn and the closeness to which they can be read is necessarily limited. Another important reason is that simply executed graphical solutions assume resistive loads and perfect power-supply regulation. A third difficulty is that preparation of curve families describing distortion *vs.* power output for various load impedances and at various frequencies is a laborious process by graphical means. Wisely conducted experiment solves the problem and checks the result at the same time.

A practical method is to set up the output stage, power supply, preceding stage and any other circuit elements likely to influence the distortion or power output capability. For such tests one employs an output transformer of a type as nearly like the final model as can be estimated and procured; or if an investigation of only the tube is to be made, impedance-coupling may be more satisfactory. Performance measurements are then made to determine optimum loading for the output tube.

A simple circuit and its related harmonic distortion curves are shown in *Figs.* 2 and 3. Gain reduction by negative voltage feedback from output plate to preceding cathode was 22 db at a measurement frequency of 1000 cps. Power output of 2 watts into 7000 ohms was obtained with approximately 5 volts r.m.s. input signal. For 6J7 and 6V6-GT, miniature types 5879 and 6AQ5 could be substituted with little change in performance. Impedance coupling is shown for simplicity; for actual power into the load, one must take output transformer efficiency into account.

The output impedance of this small amplifier is 400 ohms at the output plate in the circuit as shown, and may be further reduced to 160 ohms by bypassing the 6V6-GT cathode resistor. Assuming 90 per cent transformer copper efficiency, the output impedance on an 8-ohm voice-coil winding would be 1.35 ohms, or 1.05 ohms with the 6V6-GT cathode bypassed.

Examination of the distortion curves of *Fig.* 3 shows that if a small amount of maximum power output capability can be spared, considerable distortion reduction at listening levels can be realized by simply operating with higher load impedance.

Since the usual assignment for amplifiers is constant voltage output (for constant input), regardless of load or frequency variations, it follows that higher load impedances at high frequencies due to loudspeaker voice-coil series inductance is more helpful than burdensome as far as nonlinear distortion is concerned. At low frequencies, however, harmonic distortion increases due to shunting of the useful load by finite reactance of the primary winding. The extent to which the load-line can open to elliptical form under these circumstances (due to the alternating component of plate current being large and out of phase with plate voltage) is sufficient to increase nonlinear distortion appreciably. With the circuit of *Fig.* 2, at 2 watts output into 7000 ohms resistive load, distortion increases with shunt reactance of 7000 ohms from 0.2 per cent to 0.7 per cent. With shunt reactance increased to 14,000 ohms, distortion at 2 watts output falls below 0.4 per cent.

The form of *Fig.* 3 is a particularly effective display for finding optimum loading. These curves may be run at a mid-frequency (say 1000 cps), and at the lowest and highest frequencies at which rated output is to be obtained. The resulting curve families permit the designer to select loading conditions which best fit the amplifier to equipment requirements.

The multiple-frequency formality is necessary because optimum loading is a compromise giving proper weights to tube overload, and power supply and output transformer limitations. One might argue that if leakage inductance, iron distortion and reactive loading are severe enough to influence optimum loading, then the output transformer should be of better quality. On the other hand, when the measured distortion at rated power output is no greater at the lowest and highest required frequencies than it is at 1000 cps, the transformer is probably better than is warranted by the particular tube combination and circuit in use.

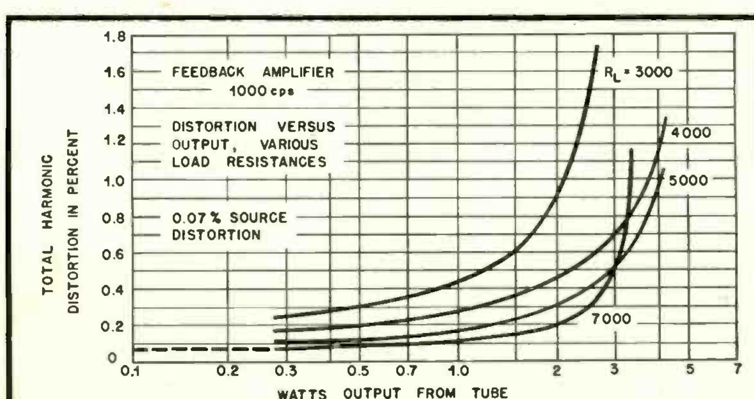

Fig. 3.

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EDITOR'S REPORT

COMING EVENTS

SHOWS, EXPOSITIONS, CONVENTIONS, and the Audio Fair are lined up for the next three months in a fashion that places a staggering drain on the resources of manufacturers and distributors, not to mention those who attend merely as visitors.

First on the list is the Western Electronic Show and Convention—WESCON—which takes place at the Civic Auditorium in San Francisco August 19-21. This gathering, corresponding to the IRE Show in New York in March, is held in conjunction with the IRE West Coast Convention, and offers twenty-five technical sessions at which over 150 papers will be presented. Two of the sessions are devoted to audio—the first being a "Symposium on Measurement of Sound System Performance" under the chairmanship of Vincent Salmon, while the second is titled "Acoustics and Audio Systems." Both sessions were organized with the coöperation of the S. F. Section of the Audio Engineering Society and the IRE Professional Group on Audio, and both are scheduled for Friday, August 21—one in the morning and one in the afternoon.

The midwest's big event is the International Sight and Sound Exposition combined with the 1953 Audio Fair in Chicago, opening for a three-day run at the Palmer House on September 1. The combined show is a new event—since Audio Fairs heretofore have drawn capacity crowds without the addition of TV, it is probable that the Palmer House will resemble a New York subway car at rush hour. The combination of audio equipment with television sets for the home is not such a bad idea—from the audio standpoint. Comparing, in the same exhibit, the sound quality of the average TV set with that usual from high-quality audio installations should advance the interest in the latter to a great degree.

S. I. Neiman, president of the ISSE, recently announced the establishment of the Exposition's gold medals, to be given for outstanding products or services in the industry. These awards constitute the first recognitions to be given out by any of the many exhibit managements. Naturally there is considerable interest in the Exposition's medals.

But to the thousands who will attend the Exposition the greatest attraction will lie in the many exhibits of audio equipment—properly served up so as to be heard as well as seen, which is one of the basic differences between Audio Fairs and most other types of shows. An audio show without the opportunity of hearing would be about as useful as a necktie catalog printed in black and white—the important part would be absent.

The first Audio Fair, held in New York in 1949, drew an attendance of 3022—which was considered adequate by all of the exhibitors. The second year drew 5500, the third 8000, and last year's show in New York registered over 13,000. Chicago's first Audio Fair—held following the Radio Parts Show in May, 1952, recorded an attendance of around 9000, and the Los Angeles Audio Fair last February drew a record 17,000. Since 1953 is audio's biggest year since the industry began to specialize on high-quality reproduction for the home, new records are certain to be established these coming few months.

San Francisco is interested in audio, too, and on September 25-27 the Palace Hotel in the City by the Golden Gate will play host to the Northern California Audio Show, where some sixty exhibits will be presented.

Then comes the New York Audio Fair, October 14-17. Three floors will be occupied this year, providing more room for exhibitors and visitors alike. As usual, the AES will hold its fifth annual convention at the same time. Of these affairs, more next month and the next.

The Third Annual High-Fidelity Conference and Audio Show in the greater Philadelphia area opens November 3 for a two-day run, occupying the fourth floor and the Crystal Room of the Benjamin Franklin Hotel. This show is arranged by the distributors in the Philadelphia area, but many manufacturers will present their own exhibits in addition to those of the distributors.

To anyone planning to attend one or more of these shows, a word of advice: wear a comfortable pair of shoes, get plenty of rest for at least two weeks before, be sure not to have a headache the day you attend, and provide small bits of cotton for those Golden Ears in case the going gets too rough.

A MUCH-ABUSED TERM

Observers at the recent Chicago show of the National Association of Music Merchants comment on the fact that all sorts of merchandise is now being labeled "high fidelity." In bygone years, the RMA limited the use of this term to sets which would reproduce up to 7500 cps. However, the high-fidelity radio set of the mid-thirties is a far cry from some of the superb equipment now available. Even though reproduction from some of the $19.95 "high-fidelity" phonographs may be better than was obtainable from the best equipment of twenty years ago, it is still doubtful if they are entitled to be so labeled. Someone will have to draw the line somewhere, and soon, else the words *high fidelity* may become as obsolete as the word *wireless* (in the U. S. A.).

SPEAKING OF AWARDS

A few months ago, Æ announced its awards for records, hearing aids, and dictating instruments. Descriptions of the award winning instruments will be found in this issue. We are pleased to note that the V. P. Edison Voicewriter won for its designer, Carl Otto, the National Industrial Design Institute award. While this award was for appearance rather than performance, it is pleasant to learn that another body agrees with Æ in the selection of an award winner.

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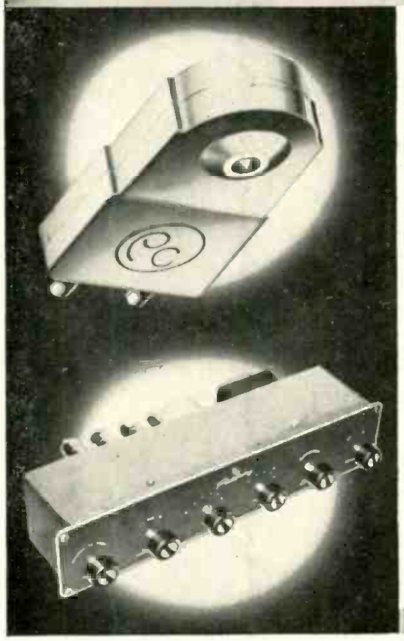
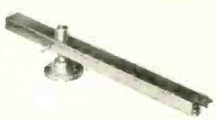


PICKERING CARTRIDGES . . .

are the choice of audio engineers throughout the world. They are universally acclaimed because of their high output, wide range performance and low distortion. They are used wherever a fine cartridge is required in radio stations, recording studios and for purposes of quality control by leading record manufacturers.

MODEL 410 AUDIO INPUT SYSTEM . . .

is designed to provide a complete audio control center. Model 410 may be used in any high quality playback system. Three input channels are provided—one for magnetic cartridges and 2 "flat" channels for other audio circuits. A 3-position equalizer network is built into the magnetic cartridge channel and provides accurate equalization for LP, AES and 78 rpm recording characteristics. Separate bass and treble controls are also provided. These are of the step-type and permit bass and treble adjustments in 2 db increments. The tone control circuits are intended to compensate for record characteristics and for listener-environment acoustical conditions. *They are not intended to compensate for amplifier and/or loudspeaker deficiencies.* Model 410 is intended for use with the highest quality professional type playback equipment. The output of the Model 410 is fed from a cathode-follower circuit and will work into any high quality audio or line amplifier having a high impedance input. It may also be used with a transformer for the purpose of feeding a 500 ohm line. Because of its flexibility, low noise and low distortion level, it is ideally suited for bridging and monitoring purposes and for critical listening applications.



THE MODEL 190 ARM . . .

is designed primarily for use with microgroove records. Its design has been recognized by leading audio engineers as that which incorporates all of the desirable tracking characteristics. Analysis has shown that for maximum performance with LP records the vertical mass of the moving arm element must be held to a minimum and further, that the arm must be counterbalanced about the vertical axis. This permits minimum stylus or tracking force and provides maximum record life. The Model 190 Arm embodies these all important features necessary for proper microgroove record playback.

MODEL 230H EQUALIZER-PREAMPLIFIER . .

is unique in its accuracy of equalization and frequency response. The intermodulation distortion is .2 per cent at normal output level. It is intended for use with high quality amplifiers having gain and tone controls. When used with the Pickering Model 132E Record Compensator the 230H is ideal for radio station and recording studio use and for applications requiring accurate low noise and distortion free playback.

MODEL 132E RECORD COMPENSATOR . . .

is designed to be used in conjunction with a magnetic cartridge preamplifier such as the Pickering 230H or any preamplifier which provides 6 db per octave bass boost. Six playback positions are incorporated:

1—European 78 rpm Records
2—Victor 45 rpm and Decca 78 rpm Records
3—No high frequency roll-off, 500 cycle turnover
4—All Capitol Records, new Victor 33⅓, Audio Engineering Society Curve
5—Columbia, London and most LP Records
6—To remove the hiss from old noisy records

Precision elements are used in its construction to give accurate compensation. The 132E is inherently a low distortion R-C device.

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Unrolled view (one-third size) of capacitor unit wound with "Mylar." The transparent film is only 0.0005" thick yet stands handling without breaking.

AMONG the elements that guard your dial telephone service are electrical capacitors. They help prevent the formation of arcs that pit and may eventually destroy relay contacts. But millions more of these capacitors are needed each year. How could they be made less costly?

Bell Laboratories engineers, on the lookout for new materials, became alert to the possibilities of the new "Mylar" polyester film. A product of the Du Pont Company, "Mylar" is chemically the same as Du Pont's "Dacron" polyester fiber used to make fabrics. Bell engineers discovered that it also had remarkable dielectric properties—of just the right kind to help their capacitor problem.

The film takes the place of impregnated paper formerly used to separate the metal foil electrodes. It is tougher, stands more voltage and needs no impregnation. The new capacitors require no protective housing and are *much* smaller and less costly.

Here is another example of the way America's technology advances through the sharing of knowledge. Just as Bell Telephone Laboratories makes many of its discoveries—the Transistor, for example—available to other companies, so does it adapt the inventiveness of others when it can help your telephone service.

BELL TELEPHONE LABORATORIES

Improving telephone service for America provides careers for creative men in scientific and technical fields

The Sonotone "1010" Transistorized Hearing Aid

HARRY A. PEARSON*

A complete description of the award-winning hearing aid which uses vacuum tubes and a transistor in their most advantageous applications. The author also discusses the design requirements.

There are approximately three million people in this country with hearing impairment who can be helped to lead more normal lives with the aid of the modern hearing aid. Yet many choose to suffer in silence rather than help themselves. Some insight as to the reason for this may be gleaned from a consideration of events leading up to the present status of a person whose hearing is sufficiently impaired to constitute a handicap in carrying on normal conversation. Rarely does a person lose his hearing suddenly, except as a result of some disease or severe acoustic shock. In such cases, the contrast between hearing and suddenly not being able to hear, drives the person to do something about it. First he seeks medical attention and then, if advised, he will seek the services of a qualified consultant to fit a hearing aid best suited to his needs. It is more usual, however, for hearing to be lost gradually over a period of years. At first the loss is imperceptible, affecting only distant sounds. Then, as more and more of the high frequencies that carry the intelligibility of speech are lost, the person has to strain more and more to understand normal conversation. He soon blames his friends for not speaking clearly, and gradually withdraws from social activity. He hesitates to seek medical aid. When he finally does, and is advised to wear a hearing aid, he hesitates. He thinks that wearing a hearing aid will reveal his impairment, not realizing that without a hearing aid he is advertising it. When such a person is finally convinced by family, friends, and physician that a suitably fitted hearing aid may be an answer to his problem, it is little wonder that he would like it to be invisible. It is for this reason that audio engineers concerned with the design of hearing aids strive to achieve maximum performance in minimum space at reasonable operating cost.

* *Director of Research, Sonotone Corporation, Elmsford, New York.*

Fig. 2. External appearance of the Sonotone 1010.

General Requirements

The principal purpose of a hearing aid is to enable a hard of hearing person to hear ordinary conversation. In average conversation sound pressure levels of about 60 decibels are produced. Weak speech may be as low as 45 decibels. The sound pressure level corresponding to average normal threshold of hearing for the frequencies most essential in speech (500 to 2000) is about 15 to 20 decibels. Hearing loss, as measured by an audiometer, is expressed in decibels with reference to average normal threshold, so that the physical sound pressure level required for a hard of hearing person to *just hear* the frequencies essential in speech is about 20 decibels (db) greater than his hearing loss. From these considerations it is clear that a person with a 30-db hearing loss in the better ear requires a sound pressure level of 50 db to just hear, and therefore will have difficulty understanding normal speech. But "just hearing" is not sufficient. To understand speech clearly its level should be about 20 db *above* the threshold. Therefore a person with a 30-db loss would require a sound pressure level of at least 30 + 20 + 20, or 70 db to understand speech well. If he wanted to hear weak speech of 40 db level, he would need a hearing aid with 30 db of acoustic gain, although for normal speech (60 db) he could get by with 10 db of gain. On the other hand a person with 70 db loss would require a 50-db acoustic gain to hear normal conversational speech well. This would require a sound pressure level of 60 + 50 or 110 db. It must be realized that these are only "order of magnitude" figures and are likely to hold for conductive impairments. In nerve impairments recruitment is likely to take place and therefore somewhat lower acoustic gains than those indicated above would apply. The hearing loss of an individual is seldom uniform for all frequencies. So in addition to providing adequate gain, provision must be made to modify the frequency characteristic of the hearing aid to the needs of the individual. In practice this is done by providing receivers having different frequency response curves as well as

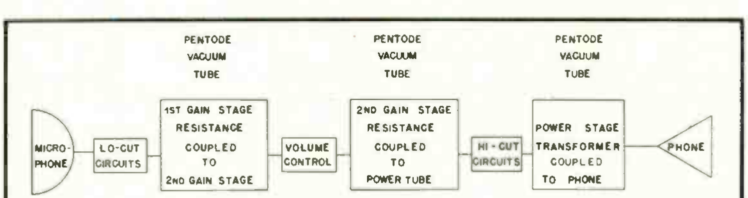

Fig. 1. Block schematic of a typical all-vacuum-tube hearing aid—the Sonotone 988.

Fig. 3. Battery compartment provides space for both A- and B-batteries, and hinged door gives access to fitting controls.

providing for control of the frequency response of the amplifier. The proper choice of ear to be fitted, power rating, type of tone compensation, characteristic of receiver, etc., is the job of the trained consultant.

Design Considerations

The conventional hearing aid circuit comprises a three-stage resistance-coupled pentode vacuum-tube amplifier with provision for a volume control and frequency response adjustment, such as that of *Fig*. 1. The input is usually a miniature crystal microphone and the output is a magnetic receiver.

With the advent of the junction transistor a new tool has become available to hearing aid designers for the realization of the objectives previously discussed. The result of the effective integration of a junction transistor into the design of a miniature hearing aid may be seen in *Fig*. 2, which shows the Sonotone 1010. This was the first product generally available to the public to use the transistor. One of the outstanding advantages of the junction transistor is that it requires virtually no power for electron emission whereas a subminiature gain tube used in a hearing aid requires about 5 milliwatts of power from an A-battery. Generally two gain tubes and one power tube are used in a hearing aid. About one-half of the A-battery power and virtually all of the B-battery power is used by the output tube. A good junction transistor will deliver about twice as much output power as a subminiature vacuum tube for equal power supplied. In addition, the cost of a given amount of B-battery power is several times as great as the cost of the same amount of power abstracted from the single cell required by the transistor. This is primarily due to the fact that cells do not become less expensive in proportion as they are made smaller. From this alone it is apparent that there is a great deal to be gained by replacing the power tube with the transistor. However, in December, 1952, when the 1010 was introduced, the available junction transistors also had some undesirable properties that precluded their indiscriminate use as replacements for the subminiature vacuum tubes in hearing aids. Foremost among these were noise and susceptibility to variation of certain parameters with temperature and time. The noise power generated by a transistor with a 20-db noise figure in the frequency range from 300 to 3000 cps is about 10^{-3} micro-microwatts whereas the sensitivity of a typical microphone is about 1 micro-microwatt for 74-db sound pressure level. This means that at a 60-db sound pressure level the signal to noise ratio is only 16 db, and for weak sounds at 45 db, signal and noise are equal. Temperature stability can be gained at the expense of battery power. Since a microminiature gain tube uses only 5 mw, the advantage of the transistor using current stabilization rapidly decreases. Variations of collector resistance and operating current with time are both serious problems in high-gain low-power stages which operate with currents of less than 1 ma and load resistances in excess of ten thousand ohms. By contrast, noise is no problem at all when the transistor is used in the power stage and the variation of collector resistance has virtually no effect since the load resistance is about 1/100 of that in a gain stage. Also, because the power stage is biased to operate at a relatively high current, (3 ma), it is relatively insensitive to changes in temperature encountered in the use of a hearing aid. These considerations led to the adoption of the junction transistor for the output stage only. With the rapid advances in transistor technology taking place it is to be expected that many of the problems discussed above will yield to solution. In the meantime this choice has resulted in providing a hearing aid having more than twice the power output and operating at half the battery cost of a well designed all-vacuum-tube hearing aid of the same physical size.

In a space $\frac{5}{8} \times 1\frac{3}{4} \times 3$ inches—only about one-half the size of a pack of cigarettes—a three-stage amplifier complete with microphone and battery supply is housed. Since a hearing aid is worn on the body, the design of the 1010 was aimed at ease of wearing. To achieve this, the thickness was reduced to ⅝ in. The case is formed of .015-in. stainless steel polished to a smooth satin finish which reduces noise due to clothes rubbing. To take full advantage of its thinness, a firm, readily removable two-fingered spring clip, see *Fig*. 2, is used. It may be mounted either on front or back, has been provided for supporting the 1010 on shirt or underclothing or in a pocket. The volume control, located in the upper right corner is provided with a smooth deeply serrated knob so that it may be operated readily through clothing from the top or side by the touch of a finger. The operating lever of the ON-OFF switch is located in the upper left hand corner. A third position on this switch reduces low-frequency response, which is useful for hearing in noisy locations. The microphone is flexibly supported behind the grill at the

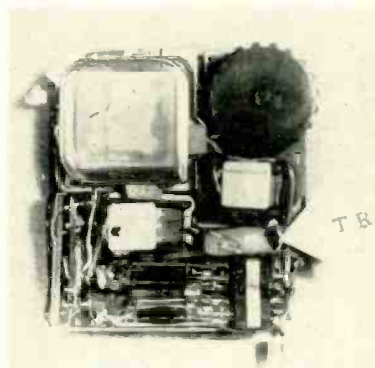

Fig. 4. The amplifier chassis, showing the locations of microphone and volume control. TR indicates the transistor, and the small white squares are transformers.

top of the instrument. *Figure* 3 shows the back with the hinged cover open to reveal the battery compartment. Silver contact buttons mounted on stainless steel springs are used for contacts in the battery compartment. Immediately above the battery compartment two oval-shaped openings are visible through which the low- and high-frequency control switches are adjusted. At the top of *Fig*. 3 a small insert receiver is shown connected to the output socket of the set. *Figure* 4 is a front view of the chassis showing the volume control and microphone at the top, two transformers below them, and the transistor (TR) and two tubes below these. On the left side below the microphone is a female receptacle for connecting accessories such as telephone pickup or external microphone.

Figure 5 is a block diagram of the 1010. Since the amplifier is used to couple the transducers, the requirements of the amplifier will become clear after examining the properties of the transducers.

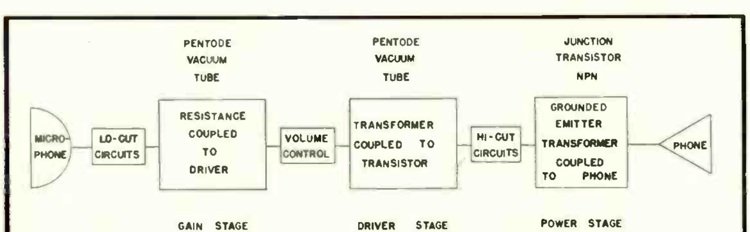

Fig. 5. Block schematic of the 1010 hearing aid.

The microphone shown in the upper left corner of *Fig*. 4 is about ¾ in. square and 3/16 in. thick. It has a series-connected, square twister bimorph Rochelle salt crystal sealed in a chamber to protect it from the effects of sustained high humidity. A ribbed conical aluminum diaphragm 1.5 mils thick provides the driving force for the crystal. Grille cloth, of specified acoustic resistance, seals the protective grille and provides controlled damping. The free-field frequency-response curve of this microphone is illustrated in *Fig*. 6. It has a smoothly rising frequency response from 200 to 3000 cps and carries out well to 7000 cps. This rising frequency response is necessary to compensate for the increase in sound pressure at high frequencies that occurs at the ear drum of a person when listening under constant sound pressure free-field conditions. It also serves to compensate for the additional attenuation of clothing at high frequencies. The microphone has a capacitance of 300 μμf and a sensitivity of about 50 db below 1 volt per microbar. This means that for ordinary speech the output of the microphone is of the order of a millivolt.

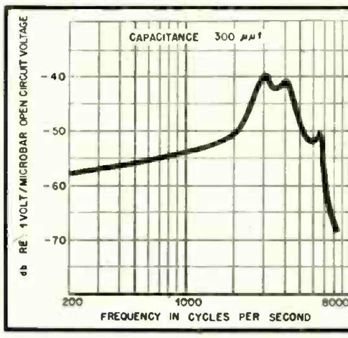

Fig. 6. Frequency response of the crystal microphone used in the 1010.

A variety of receivers with different frequency response is available from which the qualified consultant may fit the one best suited to the needs of the hard of hearing individual. One of these receivers, type 8080, is shown in the upper part of *Fig*. 3. It is only ⅝ in. in diameter and 5/16 in. thick, small enough to be concealed within the ear. Yet it has a frequency response which extends close to 4000 cps and can produce a sound pressure level of about 115 db in the ear when operating from a power source of one milliwatt capacity. Curve (A), *Fig*. 7, shows the frequency response of 8080 receiver, while curves (B) and (C) show the response curves of two other receivers, types U-1 and U-2, respectively. The U-1 is about four to five db more sensitive than the 8080 and its frequency response extends to 3000 cps. The U-2 is of lower sensitivity and has relatively greater high-frequency emphasis. The frequency-response curves were made using the standard 2-cc artificial ear as defined in ASA Z-24.9. All of these receivers are of the magnetic type. A concentric magnetic system embodying a cylindrical Alnico Magnet and a coil on a permalloy core constitutes the driving system of the receiver. The diaphragm is a thin round sheet of permalloy about 4 mils thick to the center of which a round slug is welded for carrying the flux. The diaphragm is held accurately spaced about 4 mils from the magnetic system on a machined shoulder in the brass shell. A housing and a cap complete the assembly. The frequency response of the receiver is controlled by proportioning the mass and stiffness of the diaphragm and the associated cavities and apertures in the cap and shell. The impedance of the receivers is about 90 ohms at 1000 cps.

From the previous discussion of the receiver characteristics it is clear that to develop a sound pressure level of about 120 db the amplifier should have an output of about 1 mw. Similarly in the discussion of the sensitivity of the microphone it was shown that the output of the microphone is of the order of 1 mv for conversational speech. To provide 1 mw for the receiver from 1 mv at the microphone, the voltage gain of the amplifier should be about 50 db. *Figure* 8 shows the electrical frequency response of the amplifier, and *Fig*. 9 is the circuit schematic. Reference to the block diagram, *Fig*. 5, will facilitate following the schematic. Starting at the output end, the transistor is connected with emitter grounded and biased to draw 3 ma of emitter current with 1.25 volts on the collector. This provides a power input of 3.75 mw to the transistor from which it is possible to get 1.5 mw into the load. The load impedance required is theoretically

$$R = \frac{E}{I} = \frac{1.25}{.003}$$

or about 400 ohms. While it is possible to wind 400-ohm receivers, this is not as efficient as providing transformer coupling. A 2-to-1 stepdown transformer of microminiature size was designed to match the receiver to the load impedance required for maximum power output. This transformer weighs only 5.6 grams, or less than 1/5 oz. Since the transistor requires driving power, a Sonotone microminiature power tube V_2 was designed to provide it. To assure adequate output to drive the transistor even when the A and B batteries have dropped to their end-point voltage, an initial operating current of 50 μa at 15 volts was chosen. The load required by the tube was about 0.7 meg., while the input resistance to the transistor was about 270 ohms. A microminiature transformer of high-nickel permalloy was designed to match the input resistance of the transistor to the required load. The high-impedance winding required 10,000 turns of #48 wire, a wire so fine that a bundle of ten such wires has the cross section of a human hair.

V_2 is fed from the output of V_1 through a volume control having a 40-db range. This volume control is isolated from the grid circuit of V_2 by C_6 and R_7 to prevent noisy action which would result if grid current were to flow through it.

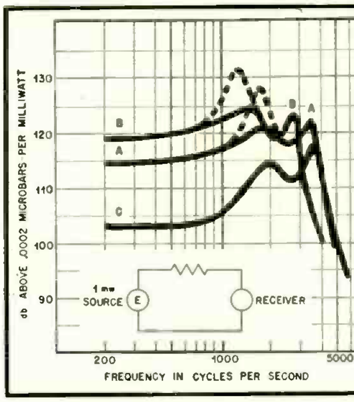

Fig. 7. Response of three types of receivers used with the 1010. (A), type 8080; (B), type U-1; and (C), type U-2. All three are plotted to show relative acoustic output when coupled to a 2-cc artificial ear. Dotted curves indicate undamped response.

The input of tube V_1 is derived from the microphone through a low-frequency control circuit. The operation of the low-frequency controls may be understood more easily by reference to *Fig*. 10. Here *e* represents the open circuit voltage developed by the microphone, C_M represents the capacitance of the microphone (300 μμf). The other symbols correspond to those on *Fig*. 9. *Figure* 10 shows that with the switches in the position indicated, the capacitance of the microphone. R_1, C_1, and R_2 constitute two stages of high-pass or low-cut networks. This provides the greatest amount of low-frequency attenuation. When the upper switch is moved from 1 to 2, one of the cascaded networks is eliminated. This provides less low-frequency attenuation. If now the switch at the lower end of R_1 is moved to the right, the full low-frequency response is available. Reference to *Fig*. 8 shows the effect of these controls on the low-frequency response of the amplifier. The high-frequency response modification shown in this figure is controlled by switching C_8 in or out of the circuit. It is quite remarkable that despite the inclusion of two microminiature trans-

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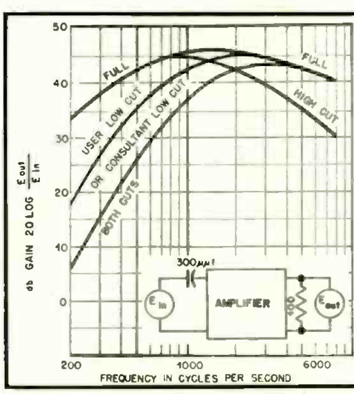


Fig. 8. Frequency response of the amplifier for various positions of the fitting controls.

The Dictating Machine—
A Specialized Recording System

In two parts—Part I

RICHARD M. SOMERS*

In 1877 Thomas A. Edison initiated the art of sound recording by his invention of the Business Phonograph. Far down on his list of possible uses was home entertainment. We have recently celebrated the 75th anniversary of Mr. Edison's invention and the company he founded has recently placed upon the market a completely new concept of the Business Phonograph.

The dictating machine has held a rather unique place in the recording business. Representing the largest sales (over 40 million dollars per year) of recording equipment in daily operation by "consumers," it has never been recognized by the technical fraternity because of its highly specialized nature and application. Technical manuals, text books and journals usually devote a paragraph or two to "another use of recording is in the business dictation field . . . ."

Reviewing the history of the dictating machine business, it is seen that for many years its product consisted of the simple, robust, voice-powered wax cylinder; with vertical "hill and dale" recording gouged by a 1/10-oz. recorder operating a sapphire cutting stylus and played back by a 1/3-oz. reproducer using a sapphire ball. In 1930–31 electrical amplifiers were added to some of the cylinder instruments and in 1941 electronics began to play an important part in the business. In 1945 the electronically powered wax cylinder system reached its peak of performance and popularity. Other media then began to throw off their swaddling clothes and by 1948 plastic was well on its way to replace the wax cylinder. From just two manufacturers during the first 65 years or so, the industry has expanded to over 50 today—altho only a few per cent of the sales are effected by other than four companies.

*Manager of Engineering, Ediphone Division, Thomas A. Edison, Inc., West Orange, N. J.

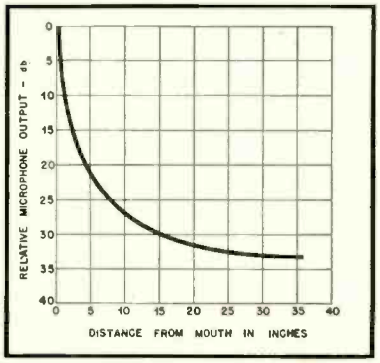


Fig. 1. Curve showing relative microphone output for various distances from dictator's mouth.

Just what is a dictation machine? It is a strange object into which one talks and, in some mysterious way, a neatly typed letter suddenly appears on the desk. Nobody really wants one—a charming girl would be, shall we say, more acceptable. It is a machine tool, usually purchased by one man for another's use. It must pay for itself in one to two years. Dictating machines, called instruments by the field, are one of the few equipment items used by top executives and department heads, rather than only by skilled or semi-skilled labor. Yet in spite of all this, it is a part of a man's business life and a failure to produce the finished letter is inexcusable.

Let us consider some other strange aspects of this equipment. It is used by totally inexperienced people as an adjunct to their thoughts. The microphone is waggled in accordance with the dictator's position, varying from zero to a foot or more from his mouth with the resultant wide variation in output shown in *Fig*. 1. The recording is done in the middle of a factory or overlooking a busy street or in a quiet, sound proofed modern office. The old voice-powered models did their best in the noisy places—they did not record the background but made fine recordings of the shouting dictator. The user often forgets how to operate even the relatively simple controls and, to cap the climax, rarely listens to his recording.

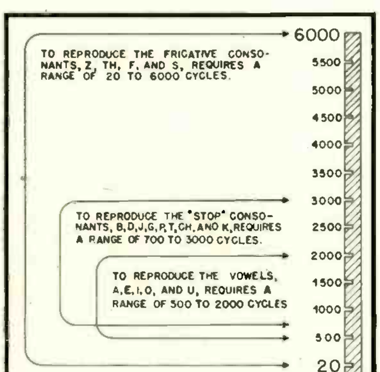


Fig. 2. Frequency range required for intelligible transmission of various speech sounds.

Intelligibility Requirements

The office dictating machine has only one function—to record the voice in a manner and with sufficient secondary information that a typist can transcribe it. This brings us to the question of recording quality, which most technical people not in this business usually call "sufficiently good for voice recording." In general, dictating machines will not record music satisfactorily. But on the other hand, good musical recorders would be unsatisfactory for office dictation. Let us see why.

Three basic intelligibility requirements are recognized in the voice field. That requiring the narrowest frequency spectrum is called Sentence Intelligibility, wherein only the general context of a whole sentence can be understood. The second, requiring a somewhat wider band is called Word Intelligibility. The modern telephone is the best example of this range. The third is called Syllable Intelligibility wherein each syllable of each word is readily recognized, and this is the type required for dictation. A study of the chart (*Fig*. 2) and of references [1,2,3,4] shows that, to obtain a syllable intelligibility of some 98 per cent, the frequency range of the equipment would have to be from 20 to 6000 cps.

Good low-frequency response is undesirable because slamming doors, machine tool vibration of buildings, footsteps, and closing desk drawers fall in this range. Since these low frequencies do not add much to syllable intelligibility, we omit them from consideration. and we end up with a sharp low cut at 100 to 150 cps and a high cut at between 5500 and 6000 cps. The low cut

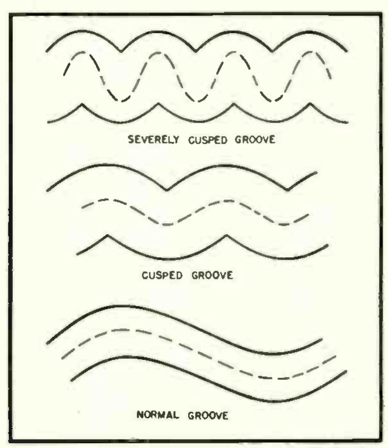


Fig. 3. Normal recorded groove in comparison with slightly cusped groove, and one which is severely cusped.

is usually obtained by RC filters and interstage coupling. Recorder resonance generally determines the upper frequency limit, aided by some RC filtering to prevent fuzz from unwanted harmonics.

Power Handling Curve

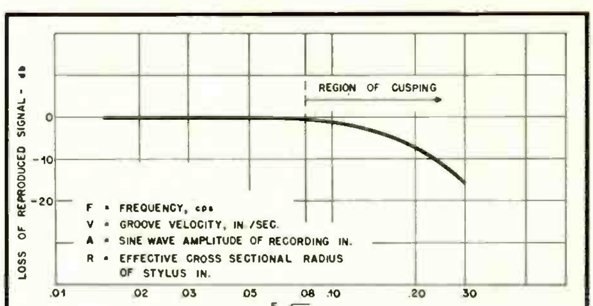
Theoretically, a stylus will record frequencies up to the point where the wavelength is equal to the diameter of the stylus tip. Above this value "cusping" occurs. This is caused by the stylus tip wiping out part of the groove. Actually, it is extremely difficult to reach this frequency experimentally in embossed recordings because of the springback of the record material which, at times, may be as high as 50 per cent of its original displacement. The springback is never uniform—i.e., the original groove shape is changed in a non-linear manner depending on original amplitude, history of the material, temperature, type of material, and so on. Thus, although cusping would seem to determine the high-frequency cutoff point, actually it does not do so for several reasons.

First, as the cusping point (*Fig.* 3) is a function of amplitude and frequency, among other factors, it will be seen that the cutoff frequency will depend on the desired permissible maximum amplitude. Secondly, the distortion which arises from cusping merely results in the generation of harmonics, the lowest one of which is twice the fundamental frequency. By using a rather sharp high-frequency cutoff slope, these harmonics are not reproduced through the transcriber's hearing device.

Figure 4 shows a form of curve useful in predicting embossed plastic performance over an extended high-frequency range. By appropriate choice of variables, a single curve depicts all conditions and allows the designer to select relative values of recorded amplitude, stylus radius, groove velocity, and cutoff frequency suitable to his conditions.

The maximum recorded amplitude in the middle register is usually matched to the output power available from the recording amplifier. The low-frequency maximum amplitude is limited by the distance between grooves. Thus, a curve can be drawn of maximum permissible recording amplitude (power) *vs.* frequency, as in *Fig.* 5. If this curve is compared to the one of peak speech energy *vs.* frequency, (*Fig. 6*), it can be seen that they are of the same general shape and order of magnitude. Thus, a flat recording is actually obtained over the voice-frequency range because the system can and does handle the maximum power of the voice at all frequencies. This phenomenon also explains one of the important reasons why the dictating machine cannot be used for music—the average maximum energy curve of musical instruments is quite different. For example, *Fig. 7* shows the relative energy output for the drum and cymbal.

Fig. 4. Effect of frequency response of varying degrees of cusping. This curve may be used for any embossed recording by substituting the correct values for F, V, A, and R.



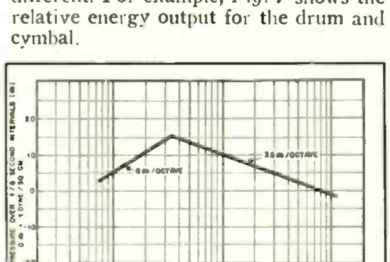


Fig. 6. Curve showing peak speech energy vs. frequency.

It is well to keep in mind the fact that these curves have nothing whatever to do with frequency response (fidelity) except that they do point up the importance of the amplitude at which a frequency response curve is measured. Obviously, a true picture will be obtained when using the standard constant input signal method only if the measurement is made at a very low level. Alternatively, the frequency response can be measured by changing the input level in accordance with the voice energy curves and correcting the output accordingly.

Signal-to-Noise Ratio

With the level of playback so low, as in earphone reception, it is possible to obtain an apparent signal-to-noise ratio of nearly infinity. This means, simply, that the playback volume is adjusted so that the noise is below audibility. A "normally" measured ratio of only some 15 to 25 db is entirely sufficient to obtain this result. It becomes clear, then, that when loudspeaker playback is attempted, as would be required for music, the noise would be unacceptable. But practice in the dictation field requires that the controls be set for the maximum permissible recording level. Several automatic methods of obtaining this result will be described later.

Plastic Medium

It might prove interesting to consider why the four principal dictating equipment manufacturers all use embossed plastic media. Two forms are in use today—the flat disc, usually about 0.010 in. thick, and the belt, some 0.005 in. thick. Each has its advantages and disadvantages. The fact that both forms are in successful competition would tend to indicate that the end results of both are acceptable. The main reason for the use of these shapes, as against "filar" types, such as tape or wire, is the simplicity and directness of locating any portion of the recording. This is obviously necessary during transcription but is also important during dictation.

Many methods of recording could be employed on these two forms of meduim. The most important criterion is, of course, the number of words that can be recorded per square inch. This reduces to the factors of linear speed and lines per inch. Strangely enough, all the known practical methods of recording seem to require about the same linear speed for equivalent recorded quality. The mechanical methods—cutting and embossing—are limited in minimum speed of the medium by their tip radii and angle for reasons of strength and life. The photochemical methods are limited by their slots, and the magnetic by their gaps. As this equality is undoubtedly one of chance, it is readily conceivable that one of these methods might suddenly outstrip the others as the result of a new discovery.

There are many other factors contributing to the selection of embossed plastic. Cleanliness is a nice one. Those

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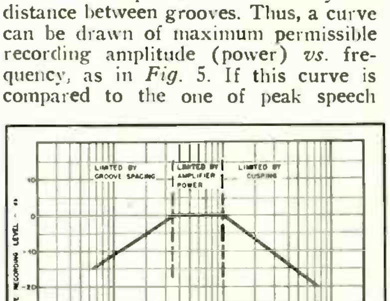


Fig. 5. Curve showing maximum recorded amplitude vs. frequency for embossing on plastic.

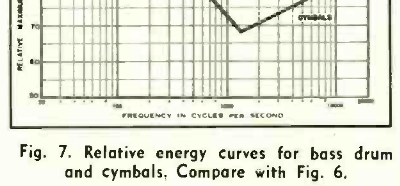
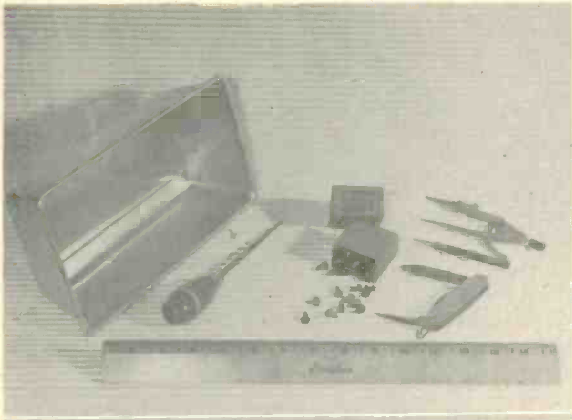
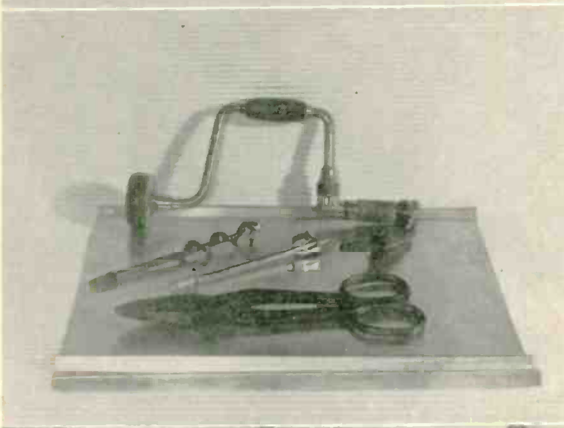
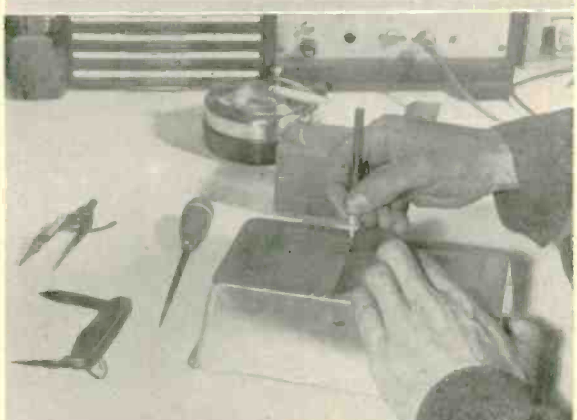


Fig. 7. Relative energy curves for bass drum and cymbals. Compare with Fig. 6.







The Bread-Pan Layout

L. B. HEDGE*

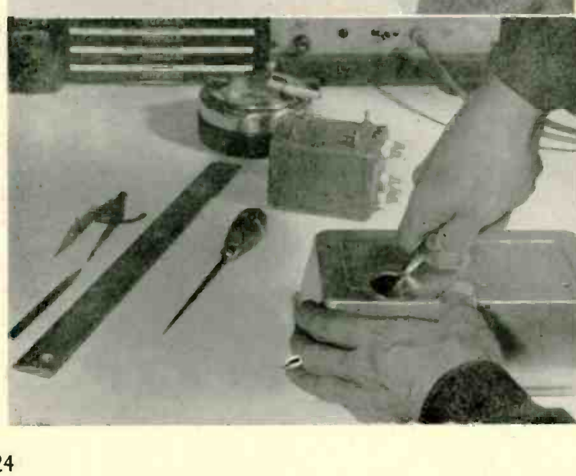
THERE ARE FEW serious workers in the audio field who have not sweated out an original or newly published circuit on an old bread board—then built it into a "professional looking" chassis and discovered that the fight had only begun. It seems a part of the innate perversity of inanimate objects of the electro-acoustic *genre* that they will not display their most exasperating peculiarities and sensitivities until they are set up in a final model—the most carefully contrived bread-board layout will not fool them.

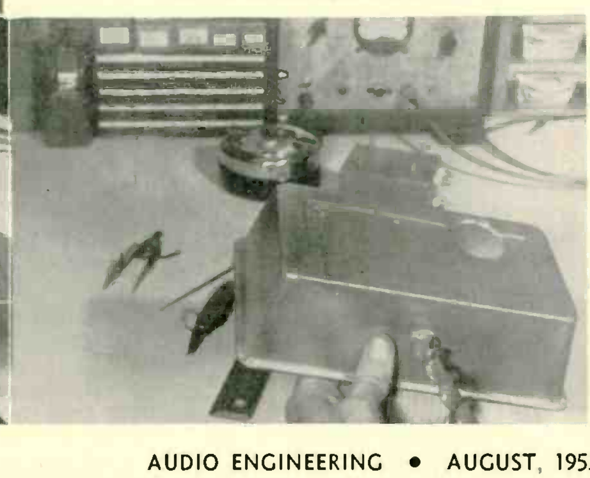
One practical answer to this ever present hair whitener (Audio engineers do not get that distinguished look just from over-exposure to good music!) is shown in the accompanying photos—*The Bread Pan Layout.* I first used it when a weekend emergency required installation of a booster amplifier in a theatre sound system which, in a moment of weakness, I had agreed to care for in sickness and in health. That amplifier, built in a bread pan commandeered from the home kitchen, was still serving nicely in the theatre system four years later when I took off for the wars. The layout plan it suggested has been enlarged upon since then (I now buy my own bread pans, for example!) but it is still basically the same idea.

An ice pick, a ruler, a Boy Scout knife, a pencil, and the basic aluminum bread pan will put you in business for this simple layout job—a compass and some self-tapping sheet metal screws will help, but you can do without them (*Fig.* 1). A sheet of light aluminum (roofing material, or another pan to cut up) will provide easily formed internal shields and a cover to be slipped over the bottom if desired. A pair of metal shears, a brace, a burr reamer and some wood augers (an expansion bit, if available) will complete the tools and material for a neat, fast, and business like job (*Fig.* 2).

* *6219 Massachusetts Ave. N. W., Washington 16, D.C.*

Left, top to bottom: Fig. 1. A bread pan and a few simple tools are the essentials in this scheme. Fig. 2. A few additional tools and some light aluminum sheet will help. Fig. 3. No template is required —mark the holes and cut-outs in pencil on the pan itself. Fig. 4. A knife cuts large or irregular holes easily. Below: Fig. 5. A wood bit will cut the medium-sized holes.





The author proposes a simple developmental circuit layout method which is an excellent substitute for the well known "bread board" form of construction. The advantages of the old method are retained to a large extent, yet the new method permits permanent use if desired, or at least offers an opportunity to approximate final construction while "de-bugging."

• • •

Parts layouts, mounting-hole and lead-through-hole locations can be marked directly on the aluminum pan with a pencil (*Fig.* 3). Small holes are punched with the ice pick (just started for metal-screw holes), and larger ones can be cut out on a compass drawn line with the punch blade of the knife (*Fig.* 4) or with a wood auger (*Fig.* 5) or the expansion bit (*Fig.* 6). The burr reamer will open up holes too small for the expansion bit, or in between auger sizes (*Fig.* 7).

Bolts can, of course, be used as with any other chassis for parts mounting, or self-tapping screws will secure socket flanges, receptacle plug wells (*Fig.* 8), and similar parts. Layout, cutting, and assembly of the whole chassis is the work of but a few minutes.

The advantages of this layout system are numerous and not all of them are completely obvious. Parts can be arranged in the same manner they would be on a more conventional chassis—under and over placement can be used to shorten leads, balance circuitry, etc. Secondary shields are easily formed and attached. The metal in these pans and roofing sheets is thin (0.020 in. approximately) but the shape, use of stiffening contours, and rolled edges makes them surprisingly sturdy. Since these pans come in a variety of shapes and sizes, unit construction in this style is easily planned and executed. The ease of working the light metal, and the simplicity of making "patches" where they are required make the successive revisions incident to a "cut and try" shakedown of a new job easy and effective. By no means the least of the advantages of the bread-pan scheme lies in the fact that a system, once beaten into working shape, can be put into permanent service "as is." A magnetic recording system thus put together has been in use by the author for over three years. The recording-playback amplifier with master switching controls was set be-

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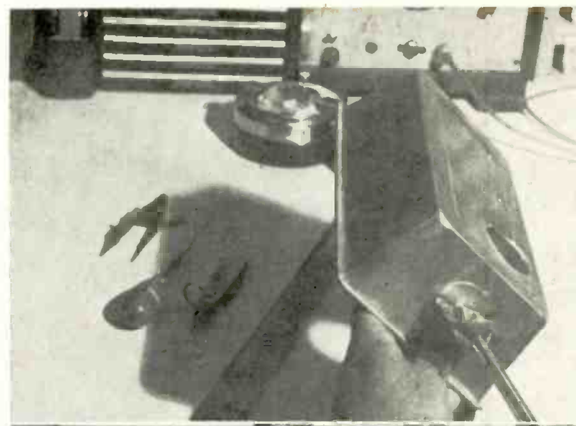
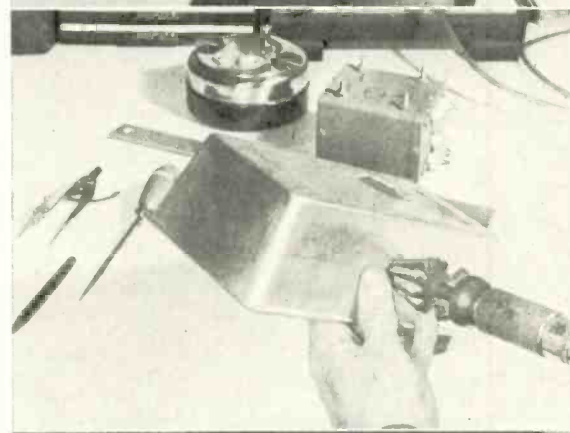
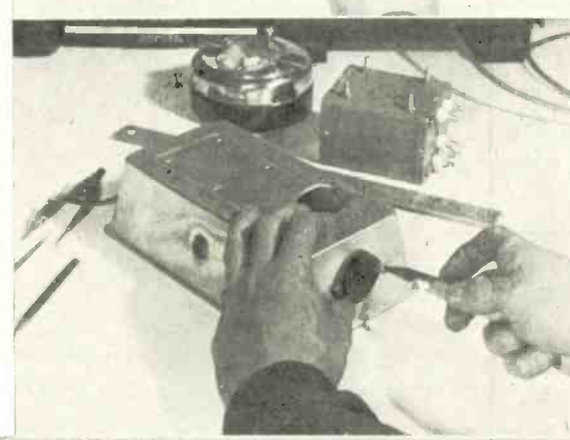
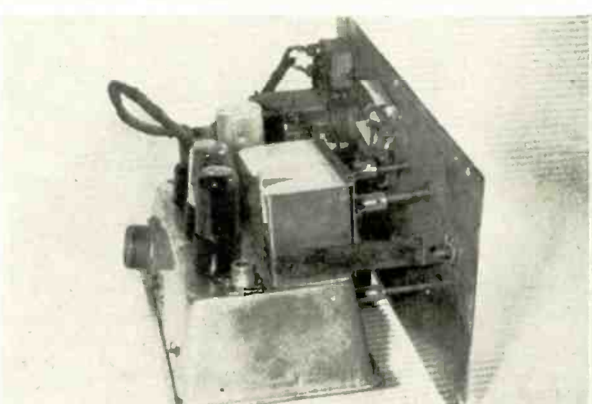


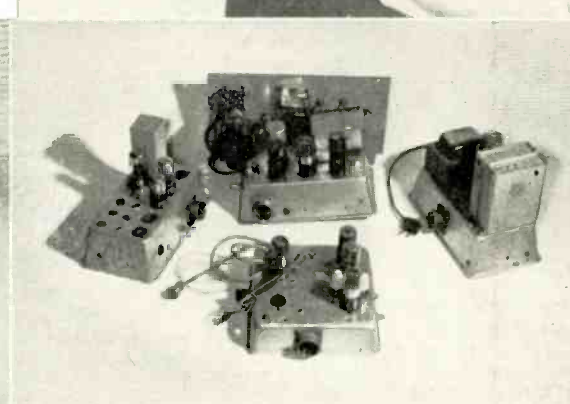




Right, top to bottom: Fig. 6. An expansion bit cuts the large round holes. Fig. 7. An ordinary reamer can be used to enlarge holes to odd sizes. Fig. 8. Self-tapping screws are used to secure sockets, plug-wells, and similar parts. Fig. 9. Typical unit construction—a magnetic recording unit with oscillator in foreground, power amplifier, record-playback amplifier, and power supply left to right in the rear.



Below: Fig. 10. Typical panel-chassis unit ready for installation.



Handbook of Sound Reproduction

EDGAR M. VILLCHUR

Chapter 12. The Power Amplifier, Part II

Continuing the discussion of the output stage with a consideration of the effect of source impedance upon the performance of loudspeakers and an analysis of methods used to reduce source impedance to a minimum.

Source Impedance of the Output Stage

All sources of electrical power have an internal impedance of their own which appears, electrically, in series with the load. The full output voltage of a battery or generator (see *Fig.* 12—10) is not applied directly across the load, but to a voltage divider whose two arms consist of the internal impedance of the power source and the load impedance. *Figure* 12—10 shows that as the load impedance is decreased, the distribution of total voltage will be shifted in favor of the voltage drop across the source impedance, and the voltage appearing across the load will be correspondingly less.

When the source impedance is much higher than the range of values of the load impedance, as it is in *Fig.* 12—11, the total impedance presented to the electrical source will be affected very little by changes in the value of the load. The small fraction of total voltage taken up by the load will vary almost in step with the load impedance variation itself, but the total current drawn from the source will remain practically constant. Such a relationship between a source of power and its load is therefore spoken of as providing *constant-current* operation.

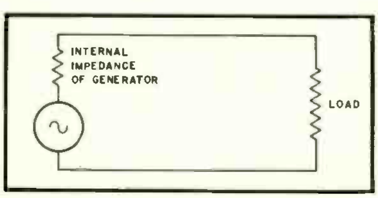


Fig. 12—10. The internal impedance of an electrical generator in relation to the load.

Constant-current output is not suitable for the dynamic loudspeaker. As the speaker impedance increases many-fold at resonance the voltage across the speaker will, with constant current, rise correspondingly while the current remains the same, and the increased power delivered to the speaker in this frequency region will result in heavy accentuation.

Where the source impedance is much smaller than the range of values of load impedance, as in *Fig.* 12—12, the instantaneous value of the total load presented to the source is determined almost exclusively by the load itself. Current drawn will therefore vary according to the changing values of load impedance, but the voltage across the load comprises almost all of the total generator voltage under any conditions, and will change very little. This provides *constant-voltage* operation.

Constant-voltage or near-constant-voltage operation is normally chosen for the loudspeaker load. As the impedance of the loudspeaker increases in the region of bass resonance it is the current, not the voltage that changes. Since the current is decreased for substantially the same voltage the electrical power delivered to and absorbed by the speaker is also decreased, but this is in partial compensation for the increased electro-mechanical efficiency of the speaker motor in the resonance region. The decrease of power in the treble may not be so compensated. However, the speaker's rated frequency response characteristics, whatever they are, have usually been measured under conditions of near-constant-voltage drive. Speakers driven by a source having a high internal impedance tend to be shrill.

The actual source impedance of a push-pull output stage consists of the plate resistance of the tubes in series, as illustrated in *Fig.* 12—13. This is is the source impedance seen by the primary of the output transformer; the value of source impedance seen by the speaker itself is stepped down by the

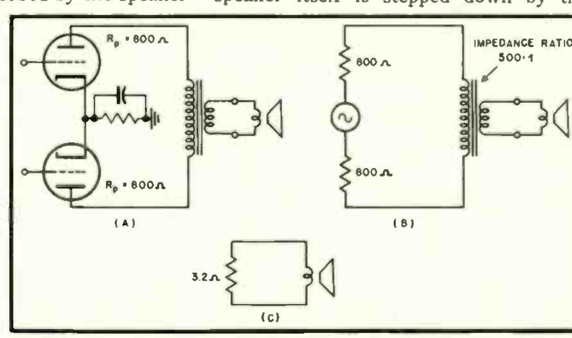


Fig. 12—13. Method of calculating source impedance seen by speaker.

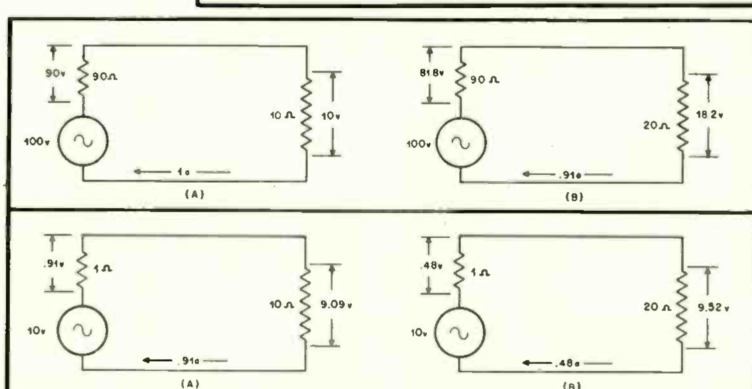


Fig. 12—11 (above). Constant-current operation due to high internal impedance. Doubling the value of the load almost doubles the load voltage, but has little effect on the current. Fig. 12—12 (below). Constant-voltage operation due to low internal impedance. Doubling the value of the load almost halves the current, but has little effect on the voltage across the load. This is analogous to the operation of the output stage and loudspeaker.

transformer impedance ratio. The *damping factor* is the relationship between load impedance and source impedance (the former divided by the latter), and may be calculated from either the primary or secondary circuits.

The combined plate resistances of the two push-pull tubes in *Fig.* 12—13 is 1600 ohms. The transformer has been chosen, according to the tube manual recommendations, to present a 5000-ohm plate-to-plate impedance to the tubes when it is properly terminated, and the secondary in this case is matched to a speaker with a nominal impedance of 10 ohms. The source impedance seen by the speaker is thus $1{,}600 \times 10/5000$, or 3.2 ohms at the 10-ohm tap, and the damping factor may be calculated from either the primary or secondary circuit as 5000/1600 or 10/3.2, both terms yielding a value of 3.12. Six db of negative voltage feedback will approximately double the damping factor.

The use of a low-internal-impedance output stage to drive a loudspeaker has another important effect on operation, which is really part of the same picture. The loudspeaker, like an electrical motor, becomes an electrical generator when the voice coil is subjected to motion. The output of this generator—looking back into the amplifier this time—is applied to a voltage divider consisting of the amplifier source impedance at the transformer secondary (at this instant taking the place of a load), and the internal speaker impedance.[7] The speaker is thus "loaded down" electrically against mechanical hangover.

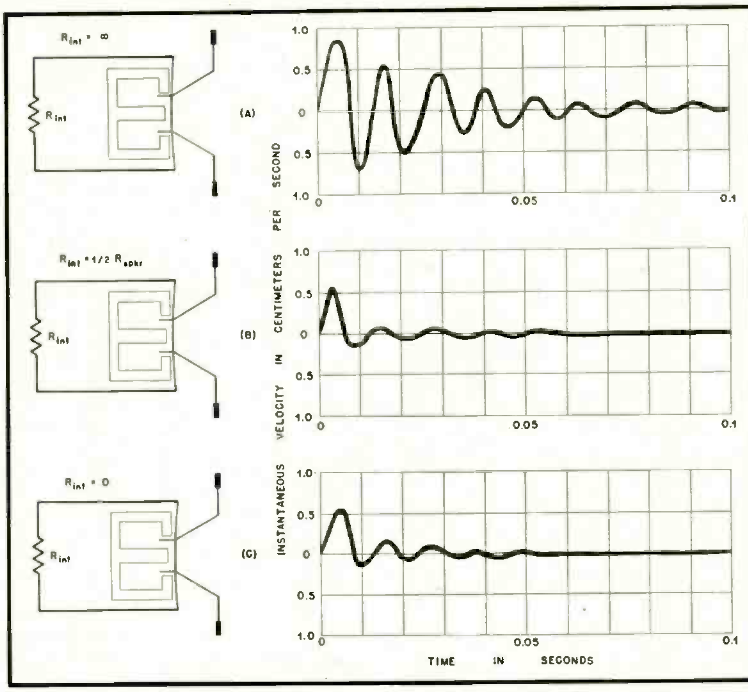
Low amplifier source impedance, low voice-coil resistance, and high mechanico-electric efficiency of the speaker generator all increase the effectiveness of the electrical braking action of the amplifier for any ringing oscillation unauthorized by the signal. When the voice coil faces a low-impedance load it cannot move, in response to inertial or elastic restoring forces rather than to the signal, without sending appreciable energy into this load, so that whatever extra mechanical energy it may possess is quickly dissipated. The low source impedance, however, appears in the output signal circuit as a series element, and so damping is achieved without the usual energy waste and efficiency loss. With a high-source-impedance stage driving the loudspeaker it takes longer for mechanical hangover energy to be dissipated electrically.

Since triode power amplifiers use a load impedance greater than their plate resistance they have inherently good damping factors. Pentode and beam-power tetrodes, on the other hand, are connected to a load whose impedance value is much smaller than the plate resistance, and therefore have poor damping factors. The use of inverse feedback on such stages can increase the damping factor beyond that of triodes without feedback. In all cases the d.c. resistance of the voice coil must still be contended with. *Figure* 12—14 illustrates the very small improvement in damping that is to be had by increasing the damping factor beyond a value of two. A further reduction of source impedance, however, can bring the output stage closer to constant-voltage operation.

Effect of Feedback on Amplifier Source Impedance

From the above it may be inferred that a relative independence of generator output voltage from changes in load impedance is an indication of low internal impedance, or at least of something just as good that can be counted as low internal impedance. Negative voltage feedback, as we have seen, produces precisely that independence between output voltage and circuit conditions that we are seeking—any voltage changes not proportional to the signal are opposed—and therefore effectively lowers the output source impedance of the amplifier. An output stage with high source impedance will, after sufficient feedback has been applied, exhibit the same relatively constant voltage output at speaker resonance and in the higher treble ranges as a stage with initially low source impedance.

The damping effect of negative feedback can be illustrated in a less abstract way, in *Fig.* 12—15. Thinking of the speaker as a generator, back e.m.f. applied to the transformer secondary has two effects; it directly stimulates current changes in the plate circuits of the push-pull tubes, and it introduces a signal into the feedback loop. The feedback signal will be amplified, and must reach the output plates in such phase as to reduce the voltage applied to them from the direction of the speaker. Plate



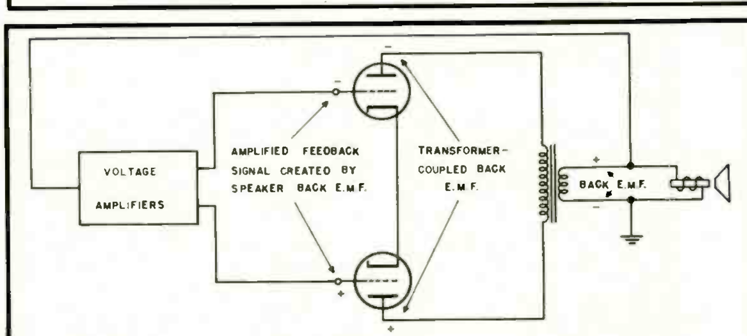


Fig. 12—14 (above). Damping effect of amplifier source impedance on speaker. (*After Olson*) The lack of significant improvement after the source impedance has been reduced to one half the value of speaker impedance is due to the series resistance of the voice-coil.
Fig. 12—15 (below). Effect of negative voltage feedback on amplifier source impedance.

[7] This internal speaker impedance consists of the d.c. resistance of the voice-coil—usually about ⅝ to ¾ the value of the nominal impedance—and the self-inductance of the voice-coil, which is only significant at the higher frequencies. Motional impedance is not seen as an internal impedance by the speaker generator, since this motional impedance is itself the effect of the back e.m.f. generated by the speaker.

and grid signal voltages are always 180 deg. out of phase, so that the instantaneous polarity of the negative feedback voltage reaching the grid will be the same as that of the speaker back e.m.f. reaching the plate. Thus, when the speaker e.m.f. drives the plate more positive and increases tube current flow, the grid is also driven in the positive direction, decreasing the instantaneous resistance to current flow between cathode and plate. When the speaker e.m.f. drives the plate in the negative direction and decreases tube current flow, the grid is also driven more negative; this increases the instantaneous forward resistance between cathode and plate, but decreases tube resistance to the current change created by the back e.m.f. The changes in plate voltage caused by free speaker oscillations are thus applied to a resistance whose instantaneous value is always decreased by the feedback signal, and the change in current flow, and energy dissipation, is increased.

The effect of negative voltage feedback on internal impedance may seem a little unreal, like a magician's vanishing act. One may well ask—have the internal plate resistances of the output tubes really disappeared? Has a circuit been created in which the signal no longer has to dissipate energy in the tube resistance to current flow? The answer to both questions is negative. If the voltage across the load does not go up with an increase of load impedance, it is because the total signal voltage applied to the load-internal impedance voltage divider has gone down. We may secure constant-voltage operation and high damping—both indices of low source impedance—but the optimum load impedance must still be calculated on the basis of the original, actual plate resistance. This resistance remains in the circuit where it always was, and takes up its normal share of output voltage.

In the years prior to the general use of negative feedback in power amplifiers, a resistance-capacitance network was usually strapped across the output transformer primary of pentode stages to compensate for the variation of load impedance with frequency.

Negative current-controlled feedback increases rather than decreases the source impedance of the stage over which it is applied. It does this in two ways: (1) by providing a resistor in series with the load, which the load sees as a direct addition to the value of the tube's plate resistance; and (2) by influencing the input signal towards

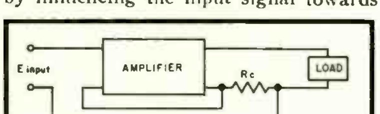


Fig. 12—16. Negative current feedback. The feedback voltage across R_C varies with the current through the load rather than with the voltage across it. Note that an un-bypassed cathode resistor is in exactly the same position in the circuit as R_C above.

providing constant-current operation. Any change in *current* caused by a change in load impedance sends back a feedback signal whose effect is to oppose the change.

The source impedance of an amplifier with current feedback (see *Fig.* 12—16) may be expressed as:

$$R_{int} = R_P + R_o - A'R_C = R_P + (1 - A')R_o$$

where R_{int} = Source impedance of current-feedback amplifier

R_P = Plate resistance of output tube or tubes

A' = No-load voltage gain between point of re-entry of feedback signal and point from which it is derived (taken as negative when feedback is purely negative)

R_C = Resistance in series with the load (often the cathode resistor) from which the feedback IR drop is derived.

With positive current feedback the sign of A' must be reversed. Since the expression $(1 - A')R_o$ may become larger in absolute value than R_P, the source impedance may be reduced to a "negative" value, which is a fictional way of expressing the fact that the speaker sees a generator, of such polarity that more current is drawn than would be by an external short circuit.

When current feedback is used with a reactive load an additional complication appears, as will be seen in the following paragraphs, in the form of varying phase shift of the feedback signal from the signal voltage.

The Output Stage and the Loudspeaker—Variation in Load Reactance Characteristics

We have spoken of the speaker impedance in mechanical terms: we have seen that above the resonant frequency of the speaker's mass-elasticity system the effect of inertia is predominant over that of the restoring forces, and the mechanical system is mass-controlled. We have also seen that below the speaker's resonant frequency the effect of the suspension forces becomes predominant, and the system becomes compliance-controlled. The speaker therefore constitutes a load of varying mechanically reactive characteristics, and the velocity of motion of the voice coil will usually not be in phase with the force applied to it. Voice-coil velocity and force will instead maintain a phase relationship analogous to that between voltage and current in the equivalent electrically reactive circuit.

It is obvious that the amplifier cannot "see" a mechanical impedance directly—the mechanical load (including the acoustical influences) must be translated into electrical terms by the electro-mechanical transducing system of the speaker. The translation takes place through the medium of the motion-induced back e.m.f., but it plays tricks. What appeared at resonance as a minimum mechanical impedance (a minimum of force was required to produce a given voice-coil velocity) is transformed into a maximum electrical impedance. This is a reasonable expectation; since the required force is reduced, the load does not have to draw as much

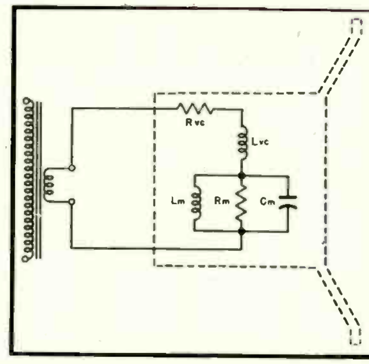


Fig. 12—17. Equivalent circuit to electrical load presented by the speaker to the amplifier. R_{VC} = Voice-coil resistance; L_{VC} = Voice-coil inductance; L_M, R_M, C_M = Components of motional impedance reflected by the compliance, resistance, and mass respectively of the speaker mechanical-acoustical system, forming a damped parallel-resonant circuit.

current and presents a higher electrical impedance to the source. The decreased mechanical impedance of the speaker is communicated to the amplifier through an increased back e.m.f. which varies directly with voice-coil velocity and which introduces an increased motional electrical impedance.

Not only does the conversion from mechanical to electrical impedance reverse the minima and maxima of impedance values over the frequency spectrum; the nature of the reflected reactance characteristic is also reversed. Force on the voice-coil must be in phase with the magnetizing speaker current, but the force will usually be out of phase with voice-coil velocity. The velocity-dependent back e.m.f. is thus shifted in phase, and the final current that flows in the speaker circuit suffers a phase shift from the signal voltage. Graphical or vector analysis will show that a mecahnical mass-reactance causes the current to lead the signal voltage, which makes the speaker load capacitive, electrically speaking, and that a mechanical compliance-reactance causes the current to lag, which means that the latter is reflected into the electrical system as an inductive reactance.

The electrical speaker load seen by the amplifier is thus a complex one, and may be represented (with certain simplifying assumptions) in the equivalent electrical circuit of *Fig.* 12—17. This is not a mechanical-electrical analogy, but it does include the electrical characteristics reflected into the amplifier output circuit from the mechanical-acoustical system of the speaker. *Figure* 12—18 illustrates the variation of single-unit speaker reactance characteristics with frequency. The inductive effect at the higher frequencies is due to the self-inductance of the voice coil rather than to motional impedance.

With a constant-voltage source of power the vagaries of speaker reactance or resistance have no significant effect upon the voltage across the speaker terminals. Thus, although negative voltage feedback may be derived from

(*Continued on page 44*)

Matched Line of Hi-Fi Equipment

HARRY F. OLSON*

An evaluation of the various parameters which must be considered in the design and assembly of components for a high-quality home music system.

THE TERM "high-fidelity sound reproduction" means sound reproduction in which the characteristics of the original sound are reproduced with realism and naturalness. The science of sound reproduction has advanced to the stage where it is now possible to provide the listener in the home with high-fidelity sound reproduction. However, great care must be exercised in the development and design of the sound reproducing equipment in order to achieve truly high-fidelity performance. Consideration must be given to the characteristics of speech and music, from the standpoint of handling these sounds without any attendant distortions. Furthermore, not only must each element of the system be of high quality, but the elements must be matched one to the other so that no losses or distortions occur in coupling the units together to form the complete system.

This paper will outline the characteristics of speech and music affecting the design of high-fidelity sound reproducing equipment and will establish the characteristics of such a system which will reproduce the sounds of speech and music. In addition, it will describe an integrated and matched set of components which has been developed and designed in accordance with these principles.

Characteristics of Speech and Music

From the standpoint of speech and music, sound waves may be described and defined in terms of six physical variables,[1] namely, frequency, intensity, waveform, duration, growth and decay, and vibrato.

Frequency is the number of cycles occurring per unit of time. To produce the significant and characteristic sounds of speech and music with frequency fidelity, the frequency range of the sound reproducing system should cover the frequency band from 40 to 15,000 cps without appreciable frequency discrimination. Such a system will reproduce well below 40 cps and above 15,000 cps because the cutoff in any practical system will not be sharp or abrupt.

Intensity of a sound field in a specified direction is the sound energy transmitted per unit in a specified direction through a unit area normal to this direction at the point. The reference level[2] of 0 db corresponds to 10^{-16} watts of sound power flow per square centimeter. The sound reproducing system should be capable of producing an intensity level in the average room, of at least 100 db. Under these conditions it will be possible to match the intensity level of a symphony orchestra. Scattered tests have shown that the sound levels found in home reproduction offer a figure of 65 to 75 db. Ordinary conversation has an average level of 70 db. Thus it will be seen that a top level of 100 db is quite adequate. The lower level of sound reproduction is limited by the ambient noise in the room, and the noise in the sound reproducing system. System noise should fall below the ambient noise in the average room so that the room and not the system becomes the limiting factor with respect to noise.

The *waveform* of a complex sound is determined by the fundamental frequency and the overtones. The overtone structure is expressed in the number, intensity, distribution, and phase relations of the components. The waveform or timbre is the most important fundamental attribute of all speech and music. It is the timbre that plays the most important part in determining the character of a voice or a musical instrument. If the response of a sound reproducing system is not uniform with respect to frequency, the original overtone structure will not be reproduced because there will be deviations from the original amplitude of the components. As a result, the character of the voice or the musical instrument will be impaired or destroyed. If the sound reproducing system introduces nonlinear distortion, then spurious harmonics and overtones are produced which, of course, alter the timbre and the resultant character of the voice or musical instrument. Therefore, the nonlinear distortion should be less than a perceptible value.

Duration is the time that a tone persists or lasts. There does not appear to be any limitation in almost any system as far as reproducing a tone of any duration.

The *growth* of a tone involves the time required for the tone to build up to some arbitrary fraction of its ultimate value of intensity. The *decay* of a tone involves the time required for a tone to fall in intensity to some arbitrary fraction of the reference intensity. The growth and decay of a sound emitted by a musical instrument depends upon the particular instrument. In some, such as the piano, the build-up time is short while the decay time is long. In the pipe organ, the growth and decay time are both relatively long. In a sound reproducing system, the transient response of the system determines whether it can follow the growth or decay characteristics of musical instruments. A reproducing system with good transient response will reproduce faithfully all sudden changes in the amplitude envelope. Thus it will be seen that it is important that the sound reproducing system exhibit good transient-response characteristics.

Vibrato means primarily a frequency modulation of a musical tone. The vibrato is also accompanied by an amplitude modulation at the modulating frequency as well as a pulsating change in tibre. The vibrato is used as an artistic embellishment by singers. Under these conditions, the average rate of the frequency modulation is seven cps. In sound reproducing systems using moving parts in the reproduction, such as the turntables in recording and reproducing phonograph records, any variation in speed will produce a frequency modulation. This is a form of vibrato usually called a "wow" in technical language. The wows due to variation in speed are most noticeable in the reproduction of music. The mechanical elements in a system should be free of nonuniform motion which produces any noticeable wow.

Characteristics of a Hi-Fi System

The preceding section has outlined, in terms of the characteristics of speech and music, the requirements for a high-fidelity sound reproducing system. It is the purpose of this section to correlate these characteristics with the subjective characteristics of hearing, and to give a specific set of data for the development and design of the system.

Frequency Response

The determination of the frequency-response range needed in high-fidelity sound reproduction involves the response of the human ear and the frequency ranges encountered in speech and music. The frequency range of the average normal ear of a young person is 20 to 20,000 cps. However, it has been shown[3] that no appreciable loss in quality of sound reproduction occurs when a sound reproducing system covers a frequency range of 40 to 15,000 cps. This range, therefore, may be treated as ideal. Any attempt to evaluate the effect of restricting the frequency range between these limits involves a personal judgment of quality. In general, the choice between ideal and restricted frequency ranges becomes a matter of economic considerations. That is to say, the user must make his own decision as to a balance between cost and the frequency range.

Nonlinear Distortion

Nonlinearity in the elements of a sound reproducing system produces distortion in the form of spurious harmonic components which were not present on the original sound. Since all the elements of a system are nonlinear in some degree, means must be devised for measuring the effects of the nonlinear distortion which is produced. This is done by subjective tests. Tests[4] have been performed to determine the perceptible, tolerable, or objectionable levels of nonlinear distortion as a function of the frequency range. In a high-fidelity system the perceptible level of nonlinear distortion is the logical one because one can hardly ask for more than being unable to hear the distortion. For an upper frequency limit of 15,000 cps and a listening level of 75 db, it has been shown that a total r.m.s. distortion of 0.75 per cent is perceptible to critical listeners. The value is selected as the ideal for this level of reproduction and this fre-

* *RCA Laboratories, Princeton, N. J.*

[1] H. F. Olson, *"Musical Engineering,"* McGraw Hill Book Company, New York, N. Y., 1952, p. 246.

[2] Acoustical Terminology, American Standards Association, Z 24.1—1951, p. 23.

[3] W. B. Snow, *J. Acous. Soc. Am.*, Vol. 3, No. 1, Part 1, p. 155, 1931.

[4] H. F. Olson, *"Elements of Acoustical Engineering, 2nd Edition,"* D. Van Nostrand Company, New York, N. Y., 1943, p. 488.

Figs. 1 to 5 (top to bottom). Type SRC-51 record player, ST-1 and SVT-1 tuners, the latter with a built-in preamplifier/amplifier combination, the SVP-10 amplifier/preamplifier, and the SV-1 preamplifier.

quency range. The ear is not as critical to distortions for higher levels of sound reproducton, and this value may be relaxed at the higher levels. For example, at a sound level of 90 db and an upper frequency limit of 15,000 cps, total r.m.s. distortions of about 2 per cent are hardly perceptible. Any compromise with the ideal nonlinear distortion becomes involved in complex subjective problems which cannot be easily resolved.

Transient Response[6]

Since all speech and music is in a continuous state of change, it is, of course, of a transient character. Therefore, a high-fidelity system must be capable of reproducing these rapid changes without any appreciable deviation. Poor transient response leads to fuzzy reproduction in the high-frequency ranges and hangover of the tones in the low-frequency ranges. As a result, much of the character of struck-string and percussion-type instruments is lost in a system with poor transient response.

One of the common methods for testing the transient response of a loudspeaker is by observing the response to a tone burst. A tone burst is a single-frequency sine-wave signal of short duration, having a rectangular envelope. The duration of the signal may vary from 50 to 500 milliseconds, depending upon the frequency. It may be said that a tone burst consists of a signal of practically instantaneous rise and decay periods. A deviation from the rectangular envelope depicts the transient response of the loudspeaker. It will be seen that a rectangular envelope, that is, the almost instantaneous rise and decay of the tone burst, is a very severe test. Therefore, any loudspeaker passing this test will handle any transients encountered in speech and music without distortion. From the shape of the envelope of the tone burst emitted by the loudspeaker, it is possible to correlate the shape with the frequency-response characteristic. For example, if there is a peak in the frequency-response characteristic, some time will be required for the output to build up to the steady-state value after the electrical input has been applied, and some time will also be required for the output to decay after the electrical input has been stopped. If there is a dip in the frequency-response characteristic, there will be a rapid build up in the output after the electrical input has been applied: then this is followed by a very low, steady-state value and then another rapid rise in the output when the electrical input has been stopped. From the foregoing, it will be seen that the growth and decay characteristics of speech and music will not be reproduced unless the system exhibits good response to transients. Faithful transient response requires a smooth frequency-response characteristic.

Directivity[6]

The directional characteristic of a loudspeaker is the response as a function of the angle with respect to some reference axis of the system. The directivity patterns are usually depicted in polar coördinates. The directivity of a piston-type radiator, such as a cone, becomes sharper as the ratio of the diameter to the wavelength increases. However, the directivity pattern can be controlled by the shape and material of the cone. If the directivity varies with frequency, then frequency discrimination will result for points removed from the axis. In most conventional loudspeakers, the high-frequency response is confined to a narrow beam, and is thus materially reduced when the observation point is a few degrees off the axis. This makes it important that the directivity pattern of a high-fidelity loudspeaker should be broad and independent of the frequency.

Power Requirements[7]

The efficiency of a loudspeaker is the ratio of the acoustical power output to the electrical power input. Therefore, the acoustical output power requirements involve the efficiency of the loudspeaker, and the electrical input power to the loudspeaker. Sound level requirements will vary over wide limits due to individual preferences. In the home a pattern of listening to radio sound reproduction has been established by the average listener. It would seem that the volume control of the radio receiver is adjusted so that speech is reproduced at conversational level and no change is made as the program changes to music. Ordinary conversation has an average level of approximately 70 db. This pattern of listening has been substantiated because tests have shown that the sound levels found in home reproduction yield a value somewhere between 65 and 75 db.

During the past seven years, hundreds of demonstrations of high-fidelity sound reproduction have been made to visitors to the RCA Laboratories. They were made in the Living Room Laboratory,[8] designed to simulate the acoustics of the living room in the average home. The average power to the Type LC-1A loudspeaker was 0.050 watt. This electrical input gave a sound level of 80 db. Over one half of the listeners have commented that the level of reproduction was too high. There have been very few that have felt it was too low. An average sound level of 75 db would thus be adequate for home sound reproduction. The electrical power input corresponding to this sound level is approximately 0.016 watt.

The above considerations involve average power. Obviously the sound reproducing system must be capable of handling the peak power. The ratio of peak[9, 10] to average power in speech and music is approximately 10 db. This means that the peak power level corresponding to the average power level of 0.016 watt is 0.16 watt. Following this, if an electrical input of 0.050 watts will deliver a level of 80 db, then an electrical input of 5 watts will deliver a sound level of 100 db. It has been shown that the peak sound level[11] in a desirable seat in a concert hall with a large symphony orchestra playing "full orchestra" is not likely to exceed 100 db. The listener can therefore simulate in his living room the sound level of a symphony orchestra with only five watts electrical input to the loudspeaker.

The question may arise as to the reason for 10 and 20 watt power amplifiers if 5 watts is sufficient. The fundamental reason is that a ten-watt amplifier will deliver 5 watts of output with less distortion. Furthermore, a larger amplifier allows a reserve so that, even with changes in the tubes with age, etc., which may reduce the top output somewhat, it will still deliver 5 watts with low distortion.

[5,6] H. F. Olson, Audio Engineering, Vol. 34, No. 10, p. 15, 1950.

[7] Olson and Morgan, *Radio and Television News*, Vol. 44, No. 5, p. 50, 1950.

[8] H. F. Olson, *J. Acous. Soc. Am.*, Vol. 15, No. 2, p. 96, 1943.

[9] Wolff and Sette, *J. Acous. Soc. Am.*, Vol. 2, No. 3, p. 384, 1931.

[10] Sivian, Dunn and White, *J. Acous. Soc. Am.*, Vol. 2, No. 3, p. 330, 1931.

[11] *Bell Laboratories Record*, Vol. 12, No. 10, p. 314, 1934.

High-Fidelity Sound Reproducing Equipment

The components shown in *Figs.* 1 to 13 have been developed and designed in accordance with the principles outlined in the preceding sections. They are part of a new line of "Matched High Fidelity" Equipment which will be marketed by the Engineering Products Department of RCA.

Players and Tuners

The Type SRC-51 record player, shown in *Fig.* 1, is a three-speed automatic record changer. It is provided with two interchangeable spindles for standard center hole and "45" records. Center hole records are handled by a gentle pusher-type platform. There is an easily inserted spindle for 45-rpm records. The spindle is designed to eliminate wear of the center hole of the record. Reproducer heads are of the plug-in type, designed for the users' choice of cartridges.

The AM-FM tuners, Types ST-1 and SVT-1, shown in *Figs.* 2 and 3, include separate tuned r.f. stages and triode converters on both AM and FM to insure low-noise reception. Wide bandwidth in the i.f. stages insures broad response, and sharp skirts on the band-pass characteristics of the i.f. sections provide good selectivity. A 10,000-cps narrow band-elimination filter cuts out the beat note produced by interchannel interference in AM reception. A completely shielded chassis minimizes oscillator radiation and insures electrical isolation of the tuner. Double-shadow tuning eye and "no-drift" a.f.c. on FM simplifies tuning.

The Type ST-1 tuner provides adequate output to drive an amplifier such as the Type SVP-10 which contains a preamplifier, but requires a preamplifier such as the Type SV-1 to drive straight power amplifiers such as Types SP-10 or SP-20.

The SVT-1 tuner of *Fig.* 3 has a built-in preamplifier which provides sufficient output to drive the Types SP-10 or SP-20 amplifiers. This amplifier may also be coupled between the record changer Type SRC-51 and the SP-10 and SP-20 amplifiers. It is provided with the following controls: input selector switch for phonograph (Ortho, LP, AES, 78 compensations), radio, television, auxiliary, and tape; volume; tuning; bass; and treble.

Amplifiers

The SVP-10 amplifier, shown in *Fig.* 4, is a combined preamplifier and power amplifier. It is provided with the following controls: selector for phonograph (78, LP, AES, Ortho compensations), radio, television, auxiliary and tape equipment; volume; bass; and treble with an OFF-ON switch.

The SV-1 preamplifier, shown in *Fig.* 5, is designed to operate in the high-quality system with the SP-10 and SP-20 power amplifiers. There are five input jacks with a control for selecting the following inputs: input selector switch for phonograph (Ortho, LP, AES, 78 compensations), radio, television, and auxiliary; one volume control for all inputs; a bass control; and a treble control with an OFF-ON switch.

The SP-10 power amplifier, shown in *Fig.* 6, is a high-quality unit with 10-watt output. Full output is obtained for an input of 0.5 volts. The amplifier supplies 6.3 volts a.c. at 1.2 amperes and 250 volts d.c. at 8.0 milliamperes for operation of the SV-1 preamplifier.

The SP-20 power amplifier, shown in *Fig.* 7, is a high-quality unit with 20-watt output, which is obtained for an input of 0.5 volts.

Loudspeakers[12]

The SL-8 and SL-12 loudspeakers, the former shown in *Fig.* 8, are extended range 8- and 12-inch single-cone loudspeakers for high-fidelity application. These loudspeakers were especially designed for a smooth frequency-response characteristic. A loudspeaker with a ragged and non-uniform response frequency will introduce frequency discrimination, will not exhibit good transient response, and will tend to accentuate noise.

The smooth response of these loudspeakers was obtained by employing a particular shape for the curvilinear cone, a special pulp for the material of the cone, and a damping ring in the outer suspension which provides a matched terminating acoustical impedance. The shape and material of the cone play important roles in determining the directivity pattern of the loudspeaker. In wide-range loudspeakers the pattern should be broad in order to reduce frequency discrimination for listening points removed from the axis. The material and shape of the cone were selected through extensive research and development so that a distribution angle of more than 40 deg. was obtained in these two loudspeakers.

The LC-1A loudspeaker,[13] shown in *Fig.* 9, is a duo-cone loudspeaker. It was first demonstrated in a test at Tanglewood, Massachusetts, on July 29, 1947 in comparison with a full symphony orchestra. The Shed at Tanglewood in which the demonstration was conducted is an immense structure—239 feet long, 200 feet wide at the rear, and 40 feet high, and seats 6000. A record was made of a selection played by the Boston Symphony Orchestra, and twelve LC-1A loudspeakers suitably housed for this type of auditorium were placed at the front of the stage to reproduce it. In the demonstration, the orchestra played the first portion of the selection; and then suddenly the orchestra stopped playing, and the remainder of the selection was reproduced from the record. Dozens of music critics from several large cities stated that it was scarcely apparent where the live orchestra left off and the reproduced sound began.

One of the outstanding features[14] of the original LC-1A loudspeaker was the broad directivity pattern. This has been broadened even further so that now uniform response is obtained over an angle of 140 deg. and the characteristic is even more uniform than in the preceding model.

These desirable characteristics are due to the following new features: a series of conical domes placed on the surface of the large cone, a damping ring in the suspension system, and a multiple vane deflector in front of the high-frequency cone. The series of domes placed on the low-frequency cone breaks up the symmetry of the unit and thereby eliminates interference normally characteristic of a symmetrical shape. The high-frequency efficiency is improved because the solid angle into which the high-

(*Continued on page 53*)

[12] It is beyond the scope of this paper to provide a complete technical presentation of the theory, development, and design of these loudspeakers. This complete treatment will be reserved for another paper in AUDIO ENGINEERING.

[13] Olson nd Preston, *RCA Review,* Vol. 7, p. 155, 1946. This paper describes the first model of this loudspeaker which was introduced in 1947.

[14] Olson and Preston, *RCA Review,* Vol. 7, No. 2, p. 155, 1946.

Figs. 6 to 9 (top to bottom). Type SP-10 power amplifier—10 watts, Type SP-20 power amplifier—20 watts, Type SL-8 loudspeaker and the newest model of the LC-1A loudspeaker.

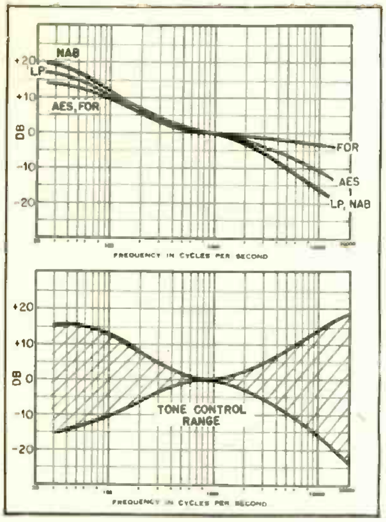


Fig. 1. Performance data for the Pilotuner, model AF-824. The upper section shows the four phono equalization curves, and the lower shows the tone-control limit curves.

New Equipment

NEW MODEL PILOTUNER—AF-824

IMPROVED DESIGN in the a.f. portion of the Pilotuner AF-821A has resulted in a new model—AF-824—which has several advantages over the earlier unit. The AF-821A was reported in this column in March. While the tuner section in the new model appears to be electrically and mechanically identical with the earlier set, the audio section is radically changed, providing better operating conditions and greater flexibility in matching recording curves.

The phono preamplifier has been redesigned to provide four different curves—LP, NAB, AES, and FOR (signifying foreign)—with curves shown in the upper section of *Fig.* 1. To accommodate the extra control, the power switch has been coupled with the volume control. In addition to the circuit change, three fixed terminating resistances have been provided, so it is now only necessary to plug the pickup lead into one of three input jacks, labeled 15,000, 27,000, and 47,000. These three values will match the most popular pickups.

Changes in the tone control circuit provide smoother operation of the controls, and a sufficiently wide "flat" section at the center makes it unnecessary to use an extreme of accuracy in setting the controls to obtain a uniform response.

A study of the schematic, *Fig.* 3, will show the new output circuits, both of which are cathode followers. Two circuits are employed so that it is possible for the user to use the tuner portion to feed another amplifier and control unit with its own controls, if desired, which may possibly have different characteristics. Another use which, to this reporter, is more important is that the detector output may be fed to a tape recorder, even though the normal output of the tuner is feeding an amplifier and speaker. Since neither volume nor tone controls affect the detector output, any change in the control setting to provide the desired sound level or response will have no effect on the recording being made. Furthermore, both of the outputs are from cathode followers, and any shielded cable connected to the detector output will have no effect on the frequency response of the regular output. And, of course, the low impedance of the cathode-follower output eliminates the possibility of affecting the frequency response adversely by long leads to the amplifier.

For crystal cartridges, the preamp may be switched out of the circuit, and a spare input may be used for TV, tape playback, or other source. A two-section volume control reduces hum and crosstalk below audibility, maintaining an excellent signal-to-noise ratio as the volume is lowered. With both L-C and R-C filtering, which uses a total of 180 µf of capacitors, and with a hum-balancing adjustment, the residual hum level is about 80 db below 1 volt at full setting of the volume control, with the input shorted.

Fig. 2. The new model AF-824 Pilotuner.

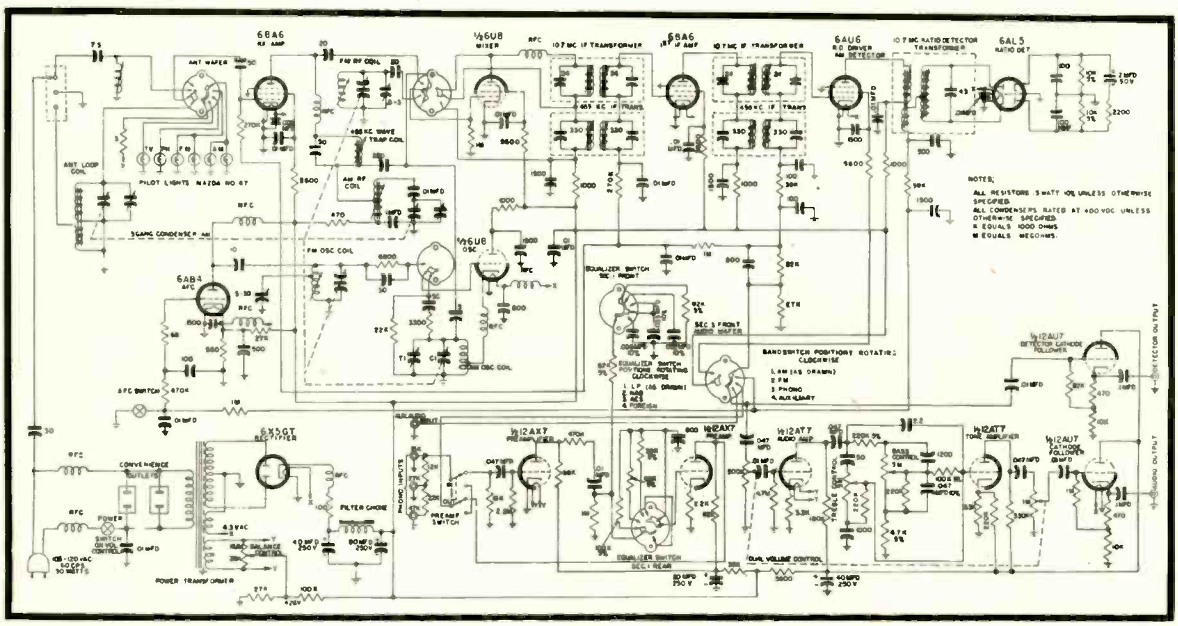
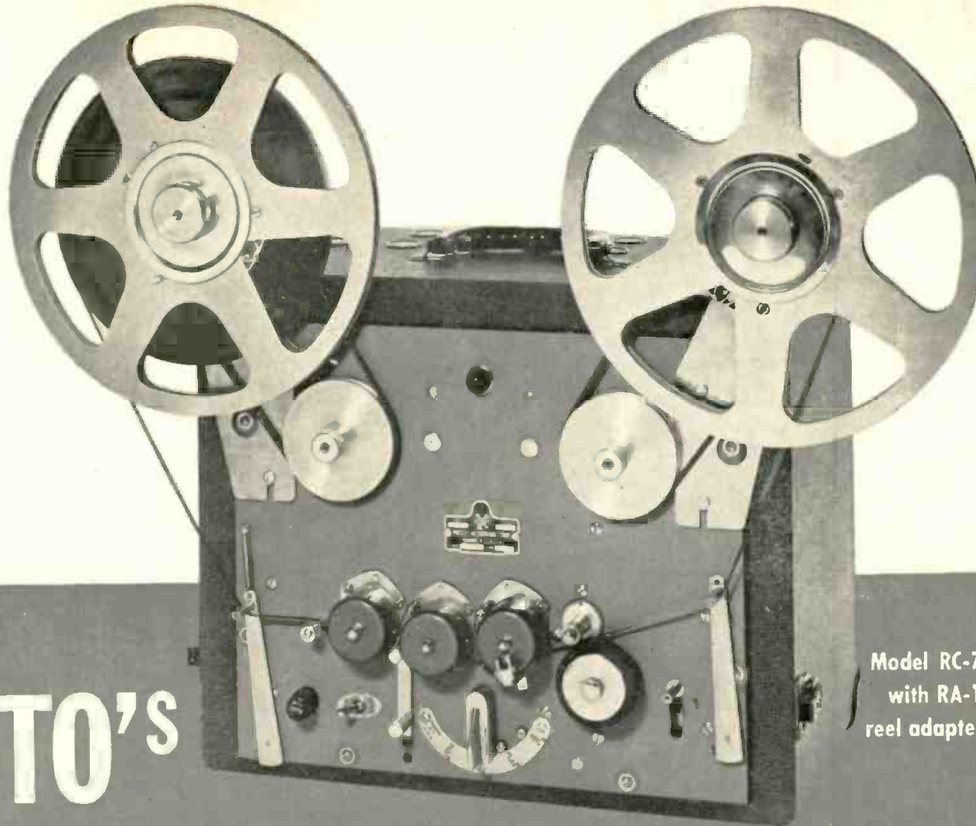


Fig. 3. Schematic of the complete tuner, including phono-preamp and tone control circuits.



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Multiple-Track Magnetic Heads

KURT SINGER* and MICHAEL RETTINGER*

A description of an economical construction method for multiple-track magnetic heads used in recording on 35-mm perforated magnetic film, particularly when such heads are used in a theater for the stereophonic presentation of motion pictures.

With the advent of "Cinerama," increased attention is again being directed to the recording and reproducing of stereophonic sound in motion pictures. It may be remembered that in 1940 the Walt Disney production "Fantasia" was reproduced stereophonically with the use of three photographic sound tracks and one photographic control track with three frequencies. Since magnetic recording has, to a large extent, replaced photographic recording in motion picture studios, it stands to reason that future stereophonic recordings will be made with magnetic film.

Two six-track heads were constructed for the purpose of obtaining data for multichannel magnetic recordings on 35-mm magnetic film. Subsequently, such a unit will be referred to as a cluster, in distinction to any of the six heads of which a cluster is composed. Two methods of construction were employed: one cluster was built by assembling six individual heads in their proper shield cans; the other unit, by combining two half clusters, which will be described in greater detail later.

The width of each track was .080 in. and the space between the tracks was .084 in. Each outside track was .050 in. from the closest sprocket hole edge. The heads were numbered 1, 2, 3, 4, 5 and 6, number 6 being closest to the recorder panel.

Assembling six individual heads in their proper shield cans presented numerous difficulties. not only in respect to the construction of the individual heads, which had to be made small enough to allow some space between the heads for individual azimuth adjustment, but also with respect to assembly. Each head had to be adjusted for track placement, head height, bearing, azimuth, and rotation (to bring the six gaps into one line). All adjustments were made under a microscope, utilizing special tools and assembly fixtures. The cluster was then plasticized in a rectangular aluminum receptacle.

* *RCA Victor Division, Radio Corporation of America, Hollywood, Calif.*

Presented on May 1, 1953 at the SMPTE Convention in Los Angeles, California. This article will also appear in the SMPTE Journal.

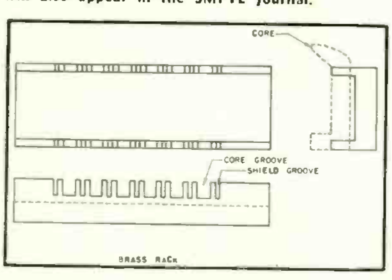


Fig. 1. The brass rack used in the No. 2 head cluster.

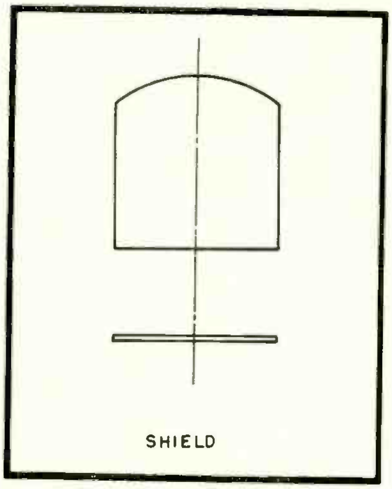


Fig. 2. Mu-metal partition used in the No. 2 cluster.

Assembling the cluster composed of two half clusters was easier and quicker and could be performed with greater accuracy in every respect. Six half cores, wound with a pre-determined number of turns of wire, were put in a brass rack, into which grooves had been milled for the placement of the cores. Additional grooves had been provided in the rack between the core grooves for the location of mu-metal partitions required for shielding purposes. After adjusting core positions under a microscope for accurate track placement with a special assembly fixture, the mu-metal partitions were inserted in their grooves, the cluster was filled with casting resin and placed in an oven for curing. Before the half cluster was completely polymerized, the greased, .020-in. thick mu-metal partitions, which were twice the size of the cores (see *Figs.* 1 and 2), were pulled out of the rack, and the half cluster was allowed to cure for the required period. This was done to permit the lapping of each half cluster on a flat plate; otherwise, with the partitions protruding, this lapping would not have been possible. What was desired above all was that all the pole faces should be in the same plane. The partitions had to be inserted to produce grooves in the plastic, and they could be removed only while the plastic was hot and incompletely polymerized.

After soldering the coil ends to a small terminal board fixed to the brass rack, equipping the cores with the required front and back gap spacers, and re-inserting the mu-metal partitions, the two half clusters were screwed together, after which the cluster was plasticized in the same type of receptacle used for the six-individual-head cluster. In this discussion, the cluster made by assembling six individual heads is referred to as cluster No. 1, and that built by combining two half clusters, as cluster No. 2.

It may be noted that the azimuth of each head of cluster No. 2 can be set, under a microscope, eleven times more accurately than the azimuth of any of the six heads of cluster No. 1. The reason for this is that the combined gap length of the heads of cluster No. 2, including the interstices between the tracks, is eleven times the gap length of an individual head. Thus, the combined length of the heads of cluster No. 2 can be utilized for the azimuth setting of any one head, because all the pole faces are in the same plane as noted before.

The inductances of each of the heads varied slightly about a nominal value of 5 mh.

Performance

The operational tests for both clusters consisted of the following:

(1) *Determination of maximum sensitivity bias.* Approximately 10 ma bias per head was required for each cluster. This test was made at a frequency of 400 cps at a level 20 db below the nominal 100 per cent modulation level.

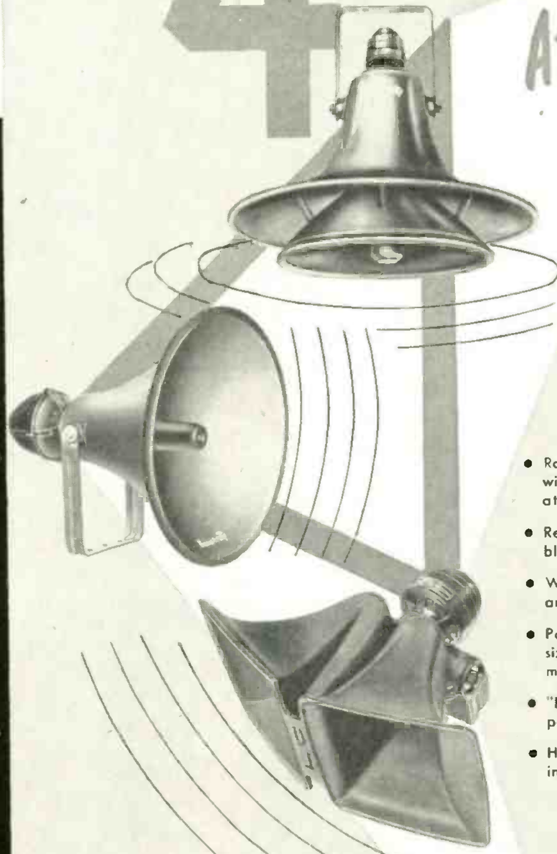
(2) *Determination of 100 per cent*

(*Continued on page 51*)

Fig. 3 (left). One of the heads used in the No. 1 cluster. Fig. 4 (right). Receptacle used for both clusters.

there are 4 points to a triangle...

THE POINT OF APPLICATION



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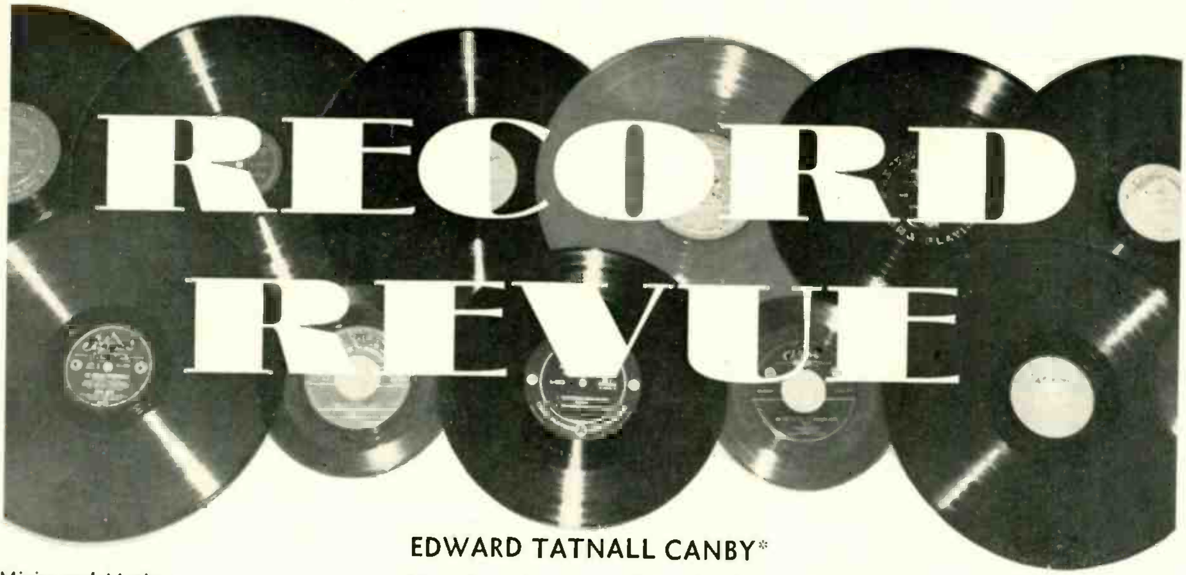

EDWARD TATNALL CANBY*

Keeping the Score III. Rhythm

Minim and Maxima

Actually the proportional system which we now use was first developed along about the 13th century, when the beginnings of counterpoint—more than one melodic line at a time—made some kind of time co-ordination absolutely necessary for musical notation. It boasted such fancy terms as *maxima* (or *duplex longa*, the double-long), *longa, brevis* (short), *semi-brevis, minim* (the minimum—except that it wasn't), the *seminimim* and the *fusa* (perhaps indicating that the notes were so fast they melted together. . . .) Note the general avoidance of a single "unit," plus the acceptance of the fractional theory, in part.

The British, in characteristic fashion, still retain a good part of this terminology and so do other musical nations. They don't even know about our theoretical "unit," the whole-note, which to them is a semibreve, a half-breve! The English call our half-note a minim and our quarter a crotchet, our eighth a quaver (it quavers but it doesn't melt)—thereby setting up a system which is completely non-informative as to the indicated relationships. Compounding logic with illogic, they proceed to the semiquaver, which is, as it says, a half-quaver (sixteenth-note); then onward via the hemidemi system; the imperturbable British musician calls our 64th note—which is always very fast—a hemidemisemiquaver. They don't have to use them very often, luckily, and if the British can pay taxes in guineas and pounds and shillings they can cope with any phlegmi-hemidemisemi that comes their way.

Hemi or otherwise, the mathematical proportion is the foundation of rhythm writing and we need add but one more joker, of the expected variety: given a theoretically exact set of fractional relations and even a good, measurable time speed for the unit-note, we must still leave room for added flexibility of remarkable amplitude, in the actual playing. The mathematical musical time-ruler is made of stretchy rubber, as you know if you've ever checked up on a Strauss waltz—where the unit value is a quarter-note, three to a measure, and the speed is . . . well, waltz speed. It would take a rubber metronome with fluid drive to keep track of the quarters and the eighths and sixteenths in that music!

* *780 Greenwich St., New York 14, N. Y.*

(Did you ever try to "keep time" to a metronome? It's uncanny how "wrong" the metronome always is—you are ready to swear that *it* plays unevenly, not you. Moreover, you are perfectly right, if you accept music in its own terms. Except for gross errors, good playing is invariably inexact as to time values.)

* * * * *

Maybe it's just as well I'm not a teacher at this point. A teacher is supposed to make everything very, very simple, so the little tots—or the dopy adults—will have a good time and enjoy their education. It's the indigestion that comes later that I'm thinking about. Sometimes you just have to get sick and start all over again. Sure enough, a quarter-note is half of a half-note and if you count "ONE, two, three," ONE, two three, you'll have three-four time. Ask any metronome. But that, my friends, is not music, even if a couple of million music students are doing it right now at this very moment. (Almost a couple of million of them will fall by the wayside before they ever get near enough to real music to hear it.)

Bar Lines

To end, then, let's assume that all and sundry reservations above-mentioned have been absorbed concerning musical flexibility. If so, we can speak of the system of notation as most teachers do, as though it were indeed a fixed and rigid system, when we know it is not. Convenience, no more.

In most recent music, between roughly 1650 and 1850, the rhythmic flow is marked off into regularly recurring pulses, felt throughout the whole musical texture even though individual elements in the music often ignore it or thrust against it (syncopation). In notation, this pulse is regularly preceded by a *bar-line*, a vertical line across the musical staff. (It has been added in modern reprints of older music where no bar lines were originally used—often merely as an eye convenience, to help keep one's place and to count off "measures" for locating a given spot. In much contemporary music the pulses are irregularly spaced, though the bar lines continue to be used.)

The time signature in recent music of this sort consists of two figures, plus sometimes a word or two that indicates the tempo or speed in general terms. The top figure of the fraction you will see thus written out indicates how many "units" you will find between the recurring pulses in the music, within the bars or measures. Some people call this the "count"—since you can count it out or beat it out. In such counting, you say "one" for the strong pulse, and add succeeding numbers for the intervening units.

The lower number of the written fraction indicates the arbitrary fractional value of the "unit"—which is more often than not the quarter-note. Thus "three-four" time, written $\frac{3}{4}$ at the head of the music, has a pulse that recurs every three units of time-beating, as in ONE two three, ONE two three, and each of those units is represented by a quarter-note's worth of abstract time value. (Or two eighths or four sixteenths.) Two-two time, or 2/2, is counted ONE two, ONE two, and the unit for this count in the music is a half-note.

Confoosin'? A wee bit—but you have no idea how much more subtle it all is than I have here indicated! I'll leave you with one last questioning, to stir up more of that mental indigestion. Think hard—and then try to pin down to a simple definition the nature of this "count" or "beat" or "pulse" I've been talking about. Is it a point in time, or an area in time? (It's both.) Is it a percussion, or merely an inner nervous explosion, an unheard but strongly felt "ummph?" (It is more often that than it is a measurable peak of volume.) Indeed, you'll find that the rhythmic pulse like almost every other part of music is one of those elusive things that begins nicely and logically, in the analysis, as a neatly scientific mathematical element and ends up, after a good long close study, as basically a phenomenon of human psychology—a part of the human animal and a small piece of Life itself. The basic pulse of life, after all, is the heart beat, which has its tempos, fast and slow, that by a strange sort of coincidence are immediately related to the pulse of music. The heart beat is a throb, a pound, and a flow—so is musical rhythm; the lung-pulse is a long drawn out and smooth rhythm—such as we find everywhere in music. Other functions of life have rhythms too—even to the rhythms of the seasons and of the ages of men (see Haydn, above); all of them find some sort

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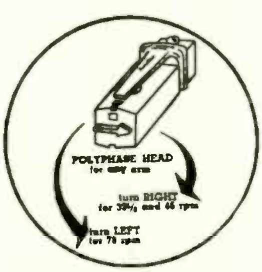

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A CAPELLA

A Festival of Choral Music. Choral Chamber Group of Pamplona, Morondo.
Westminster WL 5195

A virtuoso amateur group of fifteen Spaniards who have won assorted prizes in Europe. They make a conventional bow to older choral music—a piece by Victoria (16th century) and a few madrigal-style works, not well done to my ear; but their forte is what might be called "color choral." semi-folk arrangements, with solos, much bumm-bumming, imitation guitars, strings, etc., done with fine subtlety, good feeling. and fine musicianship. Nice Spanish feeling, moody, romantic, dark-colored, though not particularly "authentic" from the strictly ethnic viewpoint. A stunning soprano solo voice.

Recorded at a performance, this shows almost no indication of it—except an ill-advised and startling burst of applause at the very beginning of the first side. Was this Westminster's conscience, apologizing for the on-the-spot job? No need for it at all!

The Triumphes of Oriana. Various Elizabethan composers. Randolph Singers.
Westminster WAL 212

This is a triumph of common sense alertness and good musicianship. The famous series of several dozen madrigals published in 1601, each ending with the refrain "Long live fair Oriana," referring to Queen Elizabeth I, is mentioned in every music history, but as is so often the case of the entire set just one is commonly sung, over and over again, as a historical "example," the madrigal by Weelkes entitled "As Vesta was from Latmos Hill Descending." The other madrigals in the set are never heard, because they aren't generally available in print, which is merely because nobody has had the gumption to reprint them.

Mr. Randolph had the brilliant idea of recording the whole series, in honor of Elizabeth II, of course, and he had the gumption as well to go out and dig for the treasure. It wasn't an easy job at all but few of us can do less than admire a man with such a pleasing and sure-fire bit of library research on his hands! A bit like setting out to uncover the collected plays of Shakespeare, for the very first time.

There are 32 works here, not to be played as a lump—they are individual entities, unrelated to each other beyond a similarity of style and refrain. Several were written after the Queen's death, which necessitated a slight adjustment in the refrain to take care of the "long live" section. Recording is Wesminster's rather typical domestic variety, ultra-clear, ultra-close. Fortunately, the music was written for solo voices sitting about a table and the sound is reasonably faithful to that conception, in spite of a touch of tonsil scraping here and there. The Randolph group doesn't blend as faultlessly as the old English Singer and other such famed madrigalists—the St. George Singers, for instance—but intelligent musicianship makes up for it. Text provided, and you should invariably follow same while listening. Music of this sort exists to enhance the text, word by word, phrase by phrase.

Flemish Choral Music. Ghent Oratorio Society, De Pauw. **Esoteric ES 514**

Another semi-amateur group, this one numbering a hundred or so singers, well recorded via Telefunken-Ampex-Scully in an ideally live acoustic, warm resonant but sharp and clear. This group specializes in Flemish "popular" music—over some four or five hundred years—and their older music is as well done as the newer. A wide variety of material here, all most attractive in the listening. Those who know the Trapp family repertory, ranging from sacred music through Christmas music, folk settings and popular tyrolese yodels, will find a similar mixture here, in the Flemish tradition. As in the case of the Trapps, some of it gets pretty bullfroggy, some of it is almost too sentimental. A piece for every taste. Unexpected novelty: a 16th c. lute piece played on a guitar, very nicely.

Gregorian Chants, vol. 2. Monks of the Benedictine Abbey En Calcat, Boys' Choir from L'Alumnat. **Period SPL 570**

"An attempt to recapture the actual atmosphere of monastic life," this recording, played at length, certainly gets over the special mystery of this kind of Catholic dedication, splendidly aided by peals of distant bells, cathedral acoustics and an organ. The a capella (unaccompanied) portions are superb, with the special French-Latin way of nasal singing that may represent the nearest approach to older styles of vocal art that we now have. But the works with organ—alternating with the others—are anachronistic, sullying the pure and easily understood ancient choral modes with alien modern harmonies, suggesting perhaps César Franck or Gabriel Fauré, of the late 19th century. The theory is that this makes the old music more palatable; my conviction is that the music in the original, without accompaniment, is able much more easily to make its own genuine sense. to the listening ear that has patience to give it a fair trial. It is solid stuff and nourishing. Why sugar-coat roast beef? Stunningly realistic recording; crackly surfaces on my copy.

OPTIONS

Beethoven: Piano Concerto #5, "Emperor."
A. Gieseking; Philharmonia, von Karajan. **Columbia ML 4623**
B. Badura-Skoda; Vienna State Opera Orch., Scherchen. **Westminster WL 5114**

If you can put out of mind the eleven other available LP versions (by hasty count), these two make a significant and interesting comparison. Badura-Skoda is young, Austrian, one of the finest pianists for Mozart and Schubert we have, a sensitive musician. Gieseking is an old master, long established as one of the world's wonder pianists, a specialist, strangely, in Debussy, but superb in German music too.

The "Emperor" Concerto takes heroic playing; on direct comparison (I did a radio program that blended the two performances via tape trickery), Badura-Skoda plays well, but Gieseking has the strength, the long experience, the sense of huge drama, to pit his piano against the mighty orchestra on more than equal terms—next to him, B.-Skoda's playing is sincere but almost timid. The old master truly knows his stuff as few others do! Moreover, Scherchen's rather fussy, heavily dramatic Beethoven is no good foil for Badura-Skoda, whereas van Karajan's more straightforward style supplements Gieseking to perfection. Both are fine performances, in the upper brackets, but each playing will bring a greater appreciation to you of Gieseking's mastery.

Technical we have another story. English Columbia's Gieseking recording is one of those so-so affairs, perhaps from disc originals, put out by U.S. Columbia with what amounts to a flat high end, no pre-emphasis; it will sound muddy and muffled with normal LP settings. Loud passages are not always clear. Westminster's recording for Skoda is up-to-date tape, the disc cut with the sharp pre-emphasis Westminster favors (and I don't). In order to balance these I had to play the Columbia disc flat in the highs, the Westminster with the full NAB roll-off—a startling difference. With this equalization, the Gieseking disc is plenty good enough for fine listening; the Westminster is rather close, the piano a bit hard in tone.

Mozart: Divertimento in E flat, K. 563, for strings. (Violin, viola, cello.)
A. Bel Arte Trio. **Decca DL 9659**
B. Pognet-Riddle-Pini. **Westminster WL 5191**

One of those misleading Mozart works, written for a handful of instruments and yet big—in every musical sense. Those who know the famed Sinfonie Concertante for violin and viola, a prime Mozart favorite, will find this work, in the same key, startlingly similar, though the Sinfonia boasts a full orchestra and two soloists. These two performances are both sensitive, intelligent Mozart playing. Of the two I'd rate the Decca version as very slightly ahead of the Westminster; it is the more intense and Romantic, where Westminster's is cleanly classical.

There's a nice technical difference. Westminster's trio plays close-to in a good liveness, sounds like a trio. Decca's, on the other hand, is miked so that the three instruments sound as rich and big as a string orchestra—but the effect is not overdone. Which is "right"? Who can say—for the plain fact is that the music is very big music. For many listeners, then, the big Decca sound will make full appreciation of the musical sense more easy. For those who like trios to be trios, the other version will do beautifully. Again, it's a pleasure to own both.

Schumann: Die Davidsbündlertänze (Dances of the League of Little Davids).
A. Joseph Battista. **M-G-M E3011**
B. Adrian Aeschbacher. **Decca DL 7531 (10")**

This is one of those sizable collections of little piano pieces, to be played continuously as one large work, that were a Schumann specialty (the best known being "Carnaval," basis of the ballet music for orchestra). This work is as usual full of fanciful keyboard tone painting; Schumann's two make-believe characters, Eusebius the dreamy one and Florestan the fiery, have signed their initials to the eighteen pieces according to their opposite temperaments—they were the chief imaginary members of Schumann's League of Little Davids who, needless to say, went about fighting musical Philistines wherever they appeared in the then musical world. Schumann edited his own musical journal and filled it with his brilliantly playful serious comment on the then musical scene, the 1830s.

M-G-M continues its outstanding recent piano recording here; Decca improves upon its recent run of rather dry, toneless piano recordings. On quick comparison most of us would choose the M-G-M for the bigger piano sound, but the Decca, aside from a rather hissy surface, is almost as good. Both are very steady—as are most new LP piano discs from reputable companies. (That is, if your own turntable is good; piano shows up turntable faults with merciless accuracy. If you collect piano, get a good manual heavyweight table of the Presto or Rek-O-Kut variety. You'll notice the difference at once on piano.)

Joseph Battista's Little Davids are the more tempestuous here; Herr Aeschbacher has a drier, more staccato technique, but his is the more accurately phrased and executed. I'd suggest his (Decca) for pianists familiar with Schumann, the Aeschbacher (M-G-M) for non-pianists.

Haydn: Symphonies #45 ("Farewell"), #7 ("Midi"). Phila. Orch. Ormandy.
Columbia ML 4673

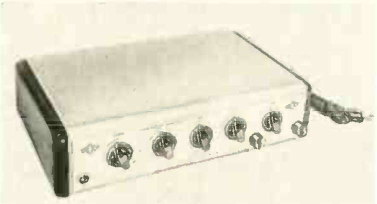
It is gratifying to see that the early symphonies of Haydn, largely resurrected in the last few years thanks to LP (The "Farewell" was the only pre-LP work of the sort that was generally known) are now at last reaching the major symphony orchestra repertory. Who would have believed, ten years ago, that the Philly would get to play number 7—in the complete list of 104 symphonies!

Unfortunately, the music is victim in this recording of a familiar misconception—that which divides all music into "symphonic" and "chamber." That is a strictly 19th century idea which has become rooted in our very performing organizations themselves and in our concert life, which generally splits cleanly into the one or the other. Only a few of the newer "little symphonies," chamber orchestras and the like have been able to provide the proper in-between musical conditions for the playing of a vast amount of earlier music that is certainly not our "chamber" music and even more definitely not symphonic, in our inflated sense.

These two symphonies, written for a small group or "orchestra" (by today's standards), playing intimately and close-to, are here grossly over-inflated to fit the arbitrary classification of symphonic. Perhaps the effect is in part due to Columbia's well-meaning engineers—but the music, for those who know the extraordinarily sensitive beauty and delicacy of this sort of expression, sounds muddy, loud, coarse; the performance relatively crude and superficial. I detect a bit of that old attitude towards Haydn so rife among our elder conductors, that his work is merely pretty in a formal, bewigged manner without true depth. Nothing could be farther from the truth, especially concerning the "Farewell" which, under its surface pleasantry, is to me one of the most profound musical expressions we have.

NEW PRODUCTS

● **Improved Brook Preamplifier.** Although similar to the earlier Model 4B in appearance, the new Brook Model 7 preamp-control unit has the additional feature of built-in power supply, as well as a number of other notable improvements in circuitry. Hum level of the unit is well below thermal noise of the tubes. Frequency response with controls set in flat



position is stated to be within 1 db from 15 to 30,000 cps. Separate 10-position stepped bass and treble controls provide a wide range of attenuation and boost. A 9-position playback-characteristic control permits matching the playback requirements of virtually all makes of records. In view of being self-powered, the new Brook Model 7 may be used with any standard basic amplifier. Complete technical specifications will be mailed free upon request to Brook Electronics, Inc., 34 DeHart Place, Elizabeth, N. J.

● **High-Power P. A. Amplifier.** Rated at 125 watts with less than 5 per cent distortion, the Masco Model MA-125 amplifier is sufficiently powerful for covering football stadiums, large indoor arenas, factories, and other area with similar sound requirements. Despite its high power, the MA-125 is a portable unit complete with preamplification and four input

channels, three for high-impedance microphone and one for phono. Separate controls are provided for bass and treble. Circuit features include stabilized inverse feedback, constant voltage output, oil-filled filters, and automatic safety interlock switch. Complete technical information can be obtained from Mark Simpson Mfg. Co., 32-28 49th St., Long Island 3, N. Y.

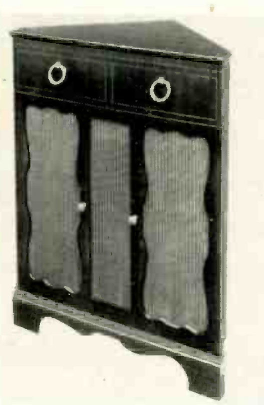
● **Heater Rectifier.** Developed especially for reducing hum by supplying d. c. heater voltage to the first stage of high-gain amplifiers, the Vector rectifier-filter unit is provided with an octal plug and occupies a space above the chassis less

than 2 ins. square by 4 ins. in height. Since its rated input is 6.3 to 7.5 volts a.c., it can be connected directly to the filament winding of the power transformer. It provides up to 0.3 amperes d.c., the d.c. output voltage being 85 per cent of the a.c. input at 0.3 amperes load, and 95 per cent at 0.15 amperes load. Filtering is adequate to reduce the a.c. component in the load over 20 db. Vector Electronic Co., 3352 San Fernando Road, Los Angeles 55, Calif.

● **Ampex Model 350 Tape Recorder.** Designed for both professional and semi-professional tape recording, this new Ampex model represents a number of departures from previous studio recorders. The tape transport mechanism is sloped at a 30-deg. angle, thus affording easy operation from either sitting or standing position. The control panel, directly in front, is also slanted. All tape motion controls are push button operated. To avoid tape stretch and breakage when small

plastic reels are used, a switch is included which automatically compensates for the increased tape tensions encountered with such reels. For servicing, the top plate is pivoted at the balance point. For routine checks and adjustments, the tape transport may be secured in a vertical position with its underside exposed, even while the machine is operating. Likewise, both the top and bottom of the amplifier-oscillator unit may be serviced without interrupting operation. A two-speed recorder, the 350 may be obtained to operate at 3¾ and 7½ ips, or at 7½ and 15 ips. Full details may be obtained from Ampex Electric Corporation, 934 Charter St., Redwood City, Calif.

● **Corner Speaker Enclosure.** Authentic furniture design is combined with quality sound reproduction in the new Musicorner line of speaker cabinets recently announced by University Loudspeakers, Inc., 80 S. Kensico Ave., White Plains, N. Y.



Though specially designed to house the University Model 6201 coaxial speaker, the Model 6200 extended range unit, or the wide-range Diffusicone-12, the Musicorner can serve equally well as the housing for most standard 12-in. speakers. Available in a wide range of finishes, Musicorner dimensions are 37" high x 28" wide x 15" deep.

● **Acetate 45-rpm Recording Blanks.** Recently announced as an addition to the line of Audiodiscs is a new 7-in. lacquer-coated aluminum-base recording blank for immediate playback on any 45-rpm phonograph. Identified by a red label, the new Audiodisc has the standard 1½-in.-diameter center hole and is designed to use a brass center-hole adapter when record-

ing. The adapter is placed over the center pin on the recorder turntable, fitting snugly within the center hole of the disc. The turntable drive pin engages the drive-pin hole of the disc in the usual manner. Paper labels on both sides of the disc provide ample clearance between adjacent surfaces of stacked records. Two re-usable brass center-hole adapters are included with each package of 25 discs. Audio Devices, Inc., 444 Madison Ave., New York 22, N. Y.

● **Microphone Mounting.** A long felt need by public speakers is satisfied by the new Atlas Model TB-1 duplex microphone mount with desk attachment. It is designed to fit any standard floor or desk microphone stand, and permits the use of

two microphones, thus reducing the possibility of a speaker getting "off mike" when turning his head. When desired the unit may be used to mount a single microphone at the rear of the desk attachment. Atlas Sound Corp., 1451 39th St., Brooklyn 18, N. Y.

● **High Fidelity Record Changer.** Constructional features of the new V-M Model 935HF changer include a laminated turntable with precision-formed concentricity, a die-cast aluminum tone arm which is virtually free of resonance ef-

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25th ANNIVERSARY YEAR

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Model C104
PHONO PREAMPLIFIER and EQUALIZER

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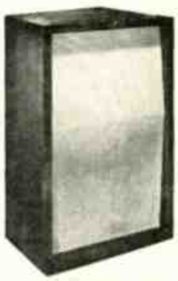
Incorporates the famous patented McIntosh output circuit for extremely high quality performance. Frequency response at 30 watts is ± .5 db from 10 to 50,000 cycles. Harmonic distortion is less than .5% at 30 watts from 20 to 20,000 cycles. Intermodulation distortion, with peak below 60 watts, is less than 1%. Has two high impedance inputs for .25 and 2.5 volt signals, and four outputs: 4, 8, 16, and 600 ohms. Power supply is built in, with provision for furnishing power to McIntosh and other preamplifiers. Uses 6BG6 output tubes.

Complete with Tubes **$139.50**



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Height 39½"; Depth 23"
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Model D-34001M (mahogany) **$331.50**
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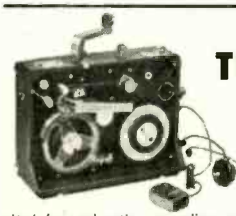

The NEW
TRAVIS TAPAK
Portable
TAPE RECORDER

A dry-cell battery-operated tape recorder which closes up to small suitcase size: 10x14x5½ inches. The Tapak is ideally suited for on-location recording anywhere: at sea, in the air, aboard trains, in cars, or on foot. It erases, records, monitors, and has built-in editing device. Provides for fast power-rewind and instantaneous playback through single earphone headset. Drive mechanism is spring-powered. Runs 5-6 minutes on one winding. Indicates when spring has run down, and permits rewinding without interrupting recording operation. Records at 7½"/sec.

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A high quality front end for receiving both FM and AM broadcast signals. **FM section** employs Armstrong System with 1½ microvolt sensitivity for 20db of quieting on 72 ohm antenna input (3 microvolts for 300 ohm input). Frequency response is uniform from 20 to 20,000 cycles, ± 1db. AFC can be switched in or out of operation from front panel. **AM section** employs superheterodyne circuit with front panel knob for selecting wide band-width for high fidelity reception or narrow band-width for greater selectivity. Sensitivity is less than 1 microvolt for 1 volt output. Has built-in 10kc filter.

Four knobs furnish the following controls: Power Switch, Volume, Tuning and Selector Switch for AM-broad, AM-sharp, FM, FM-AFC, phono, TV. Cathode follower output permits long line without high-frequency losses. Complete with tubes, AM loop, FM dipole, and built-in power supply **$159.50**

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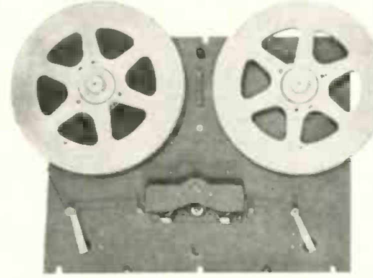
fects, and a 4-pole 4-coil motor which affords freedom from induced hum when the changer is used with magnetic cartridges. The changer is furnished minus cartridges, but with interchangeable heads into which most top quality cartridges can be mounted. The unit plays all three sizes and speeds of records automatically, with intermix of 10- and 12-in. records of the same speed. A muting switch affords noise-free changing cycle. Descriptive brochure is available from the V-M Corporation, Benton Harbor, Mich.

• **Presto 8-Hour Tape Reproducer.** Operating at a tape speed of 3¾ ips and accommodating up to 14-in. reels, the new Presto Model PB-17 playback unit will automatically play up to eight hours of recorded material without interruption. At the end of each reel, the machine will either stop or repeat, as desired. Frequency response is 50 to 8000 cps. A dual-

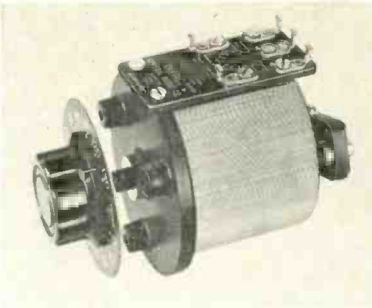


track unit, the PB-17 may be used with pre-recorded program material available from an increasing number of sources. For individual users of industrial background music, as well as for operators of wired music systems, it combines low initial cost with minimum operating overhead. Presto Recording Corporation, Paramus, N. J.

• **Console Home Tape Recorder.** Known as the "Recordio Grand", the new Wilcox-Gay Model 3F40 is a 2-speed dual-track tape recorder with all functions controlled by push buttons. Excellent audio quality is afforded by a 6-watt amplifier with push-pull output, which feeds into a 12-in. p.m. speaker mounted in a bass reflex enclosure. In addition to micro-

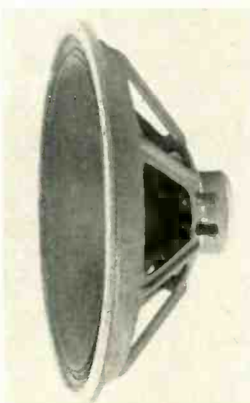
phone input, the unit is equipped with a retractable cord which is terminated with an input jack for radio-phono-TV recording. A visual index counter facilitates locating a desired recording on a given spool of tape. Fast forward and fast rewind operate at 10 times normal recording speed. A simple panel control permits choice of 7½ or 3¾ ips. The entire recording unit is mounted on a sliding drawer which can be closed while the unit is in operation. Cabinet dimensions are 30 ins. high by 22 ins. wide. Wilcox-Gay Corporation, 79 Washington St., Brooklyn 1, N. Y.

• **Improved Variacs.** Surface oxidation of the brush track on Variacs is virtually eliminated, even under continuous operation at fixed settings and high overload, by Duratrak, a silver alloy which is now being applied to all Variac models. Surge and inrush currents at least twice as great as could be handled by earlier models may be applied to the new units with complete



safety. With this new development, the limitation of the brush and track has been removed from the adjustable transformer, and ratings may now be determined by the same factors that apply to any fixed-ratio transformer, namely the temperature rating of its electrical insulation. General Radio Company, 275 Massachusetts Ave., Cambridge 39, Mass.

• **Eighteen-Inch Woofer.** Frequency range of 30 to 4000 cps is covered by a new p. m. driver recently introduced by C.-S. Manufacturing Company, 4089 Lincoln Blvd., Venice, Calif. Cone resonance ranges from 27 to 31 cps and maximum



power input is 25 watts. Outer edge of the cone is treated with Pyralin to improve damping and reduce distortion. Voice-coil diameter is two inches and impedance is 12 ohms. The unit is constructed on a heavy cast-aluminum frame and contains a 2½-lb. Alnico V magnet.

• **Three-Way Crossover Network.** Crossover frequencies of 800 and 5000 cps are provided by the New Wharfedale Model HS/CR/3 crossover network designed for use with high-quality three-way speaker systems. The unit is of constant-resistance half-section construction, and includes a built-in variable attenuator across the high-end output terminals for adjusting upper frequency response. When desired the network can be used as a two-way crossover by ignoring the treble terminal. Maximum input is 30 watts. Attenuation is at the rate of 12 db per octave. For descriptive literature, write Dept. W, British Industries Corporation, 164 Duane St., New York 13, N. Y.

SOUND HANDBOOK

(*from page 28*)

points directly across the speaker, the feedback signal does not include the speaker's back e.m.f. The signal voltage across the speaker may be thought of as the vector sum of fictionally abstracted IZ drops across various circuit elements of *Fig.* 12—17; this sum must always, according to Kirchoff's law, be equal to the applied signal voltage in both phase and magnitude. What happens physically is that the voltage remains steady while the phase and magnitude of the current change. It is the speaker current rather than voltage that reflects instantaneous voice-coil velocity in a constant-voltage system.

It would therefore be most desirable if variations in current amplitude could be made to influence the feedback voltage. Such a system has been proposed.[8] The phase-shifts between signal voltage and current, and between signal voltage and voice-coil velocity, however, present difficulties in using current-controlled feedback with the reactive speaker load.

Putting the question of phase-shift aside for the moment, it may be seen that the relationship between the magnitude of the voice-coil velocity and of speaker current is inverse, so that current-controlled feedback for correcting inaccuracies of voice-coil motion relative to the signal must be positive rather than negative. (To illustrate: if the speaker overshoots, the instantaneous voice-coil velocity is greater than called for by the signal, but current is decreased from the normal by the exaggerated back e.m.f. The current-controlled feedback voltage must be positive if a decrease in feedback is to create the required compensatory cut in output signal. The same relationships hold for speaker velocity held back from the normal by undue restraint from the suspension system.)

This positive feedback, if it could be applied in proper phase at all frequencies, would improve speaker performance but degrade amplifier performance. The increase in amplifier distortion can be overcome by using negative voltage-controlled feedback in addition to the current feedback, so that the instantaneous magnitude of the negative feedback voltage is made to vary inversely with the magnitude of the current.[9] The system begins to look as though it uses feedback which is at least partly motion-controlled, and as though it dispenses with the need for a separate feedback coil or similar element in the moving system. But the assumption made above, that a current-controlled signal can be applied in proper phase at all frequencies, is not correct. The large, varying, and reversing phase shifts, at critical frequencies, between signal voltage, current, and voice-coil velocity remain to plague the feedback circuit. *Figure* 12—19 illustrates phase relationships, at a particular frequency, between the amplifier signal voltage, the speaker voice-coil velocity, and speaker circuit current.

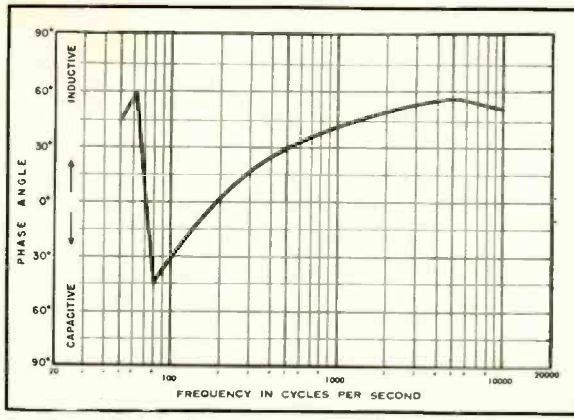


Fig. 12—18. Reactive characteristics of typical single-unit loudspeaker. (*After F. Langford Smith*)

[8] H. F. Mayer, "Control of the effective internal impedance of amplifiers by means of feedback," *Proc. I.R.E.*, March 1939, p. 213.

[9] It is interesting to note that the effect of both negative voltage feedback and positive current feedback is to reduce amplifier source impedance, up to the point of zero impedance. When positive current feedback begins to drive the amplifier source impedance negative, however, the voltage feedback, which is innocent of knowledge about internal impedances and knows only that it must oppose voltage changes across the load, has the opposite effect from current feedback: where negative source impedance dictates that the output-voltage *decrease* with an increase of load, negative voltage feedback tries to keep the output voltage constant. Thus, if positive current feedback reduces the amplifier source impedance to −10 ohms, 20 db of negative voltage feedback will increase this impedance to approximately − 1 ohm.

These relationships may be checked by an examination of the equations describing the effects of current and voltage feedback, respectively, on source impedance. Current feedback causes a new quantity R_C—$A'R_o$ to be added to or subtracted from the source impedance, so that the final sum can pass the zero mark and reverse its sign. Negative voltage feedback, on the other hand, *multiplies* the source impedance by the fraction $\frac{1}{1-A'\beta}$; in this case the source impedance, whether it is positive or negative, will always be reduced in absolute value, and driven towards but never to zero from either direction.

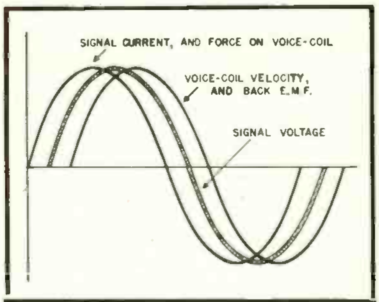


Fig. 12—19. Phase relationships, in frequency region just above speaker resonance, between signal voltage, current and voice-coil velocity.

Another proposal to make feedback dependent on back e.m.f. has been made,[10] in which the speaker is connected in a bridge circuit, and the instantaneous voltage changes due to motional impedance detected. Here, too, the varying reactive phase shifts between the output and feedback signals present a problem.

The two systems referred to are illustrated in *Fig.* 12—20. The phase-shift problem in these circuits is similar to that which exists in direct electro-mechanical feedback systems, where the transducer creates a shift between force and velocity. In all cases the minimum effect of the phase-shift is to limit the reduction of non-linear effects that can be achieved. The feedback factor is a complex quantity—that is, it has both magnitude and phase—and the extent to which the feedback signal is other than 180 deg. from the signal with which it is mixed is the extent to which its benefits are lost. Beyond a critical point, at which feedback has no effect on either gain or distortion, further phase-shift bares the second edge of the feedback sword, and distortion is increased.

Part III of The Power Amplifier will follow next month.

[10] J. P. Wentworth, "Loudspeaker damping by the use of inverse feedback," Audio Engineering, Dec., 1951, p. 21.

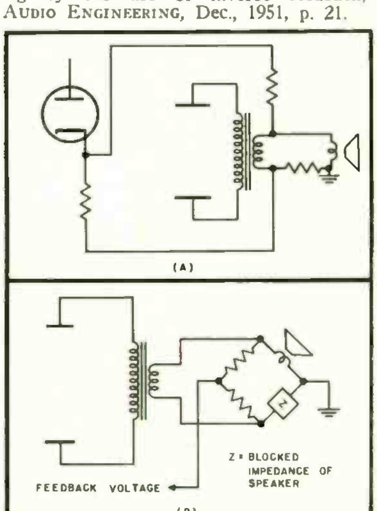


Fig. 12—20. Proposed motion-controlled feedback systems: (A) Positive current feedback added to negative voltage feedback (*after Clements*). (B) Bridge circuit.

DICTATING MACHINE

(from page 23)

of us who struggled through the years with the "chip brush" of the wax cylinder or the "thread" from the commercial master needed little prodding toward embossing. The bluntness of the embossing stylus permits relatively wide changes in material without causing comparable changes in depth of groove. Embossing in plastic has the advantage (at least when the flat disc shape is used) of permitting reuse of the record. It is merely necessary to heat the plastic beyond its "memory" point (about 180° F. for Vinyl) and the previous recording amplitude is reduced some 40–50 db—a complete wipeout for all practical purposes.

The present state of the art is fluid, but it still appears that the embossed disc is the best. One of the problems in embossing is the choice of shape and form of the groove. *Figure* 8 shows a cross section of the grooved record to show that the material is not disintegrated but merely "pushed around." Thus the "horns" contain the material displaced by the grooving. The playback stylus must distinguish between the land and the groove. Many ideas have been tried to rid the record of one or both of the steep horns without much success and with rather bad effects on high-frequency response. Specially formed and/or cocked (tilted sideways) styli are old methods. The present art has licked the problem, for all practical purposes, by recording at amplitudes high enough to wipe out the land many times in each revolution of the record.

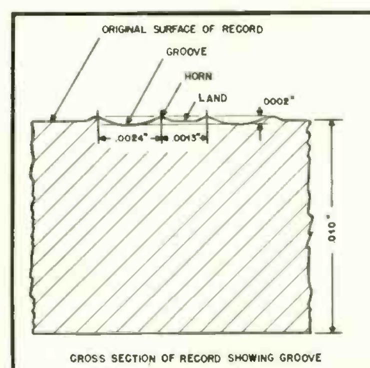


Fig. 8. Cross section of recorded disc, using embossing process. Note relative shallowness of grooves, and their relation to the "lands."

The Ediphone instruments employ a ball-tip recording stylus of 1.7-mil radius, and a playback tip of 3-mil radius which just fits the groove made by the smaller recorder stylus. Pressures range around 55 grams for recording and 15 grams for reproducing. Maximum permissible reproducer side thrust for typical groove depths of 0.2 mils (bottom of groove to top of horn) is about 1.0 gram—quite different from the values in typical phonograph pickups. Yet all of the dictating machines will record and play back with the record tilted 30 deg. or more.

Two other characteristics not peculiar to embossed plastic alone are tearing and static. Tearing of the groove tends to occur in the cusping range. Static attracts dirt and raises surface noise. Both have been brought under control by our company by multiple spraying of the disc with a liquid whose formula is a trade secret. The result is about a 2-to-1 gain in effective frequency range and a reduction of static to a negligible value. Five-year aging seems to have little, if any, effect on its efficacy.

System Requirements

Dictating machines are interesting devices, even forgetting their specialized recording and playback requirements. Take, for example, the problem of cueing or indexing. Although the dictator creates the index, it is the transcriber who must use it. Letter length must be reasonably well defined, and corrections or changes must be clearly indicated so the girl may locate these points in advance, since they may refer back some distance into the body of the letter. As the recorded intelligence per inch of recorder travel is rather high, the corresponding index motion is magnified on the scale.

One important requirement for dictation is extremely fast starting and stopping of the record. At first thought, it might be assumed that starting and stopping times may be several tenths of a second. This is based on the fact that human reaction time between the original thought impulse and the start of an action is about a minimum of 0.1 second. Of course this does not apply to dictation, for the user usually decides what he wants to say *before* he presses the start switch. Most of the machines on the market today reach recording speed in 10–40 milliseconds and stop in the same or shorter time. It is obvious then that heavy turntables, mandrels, or other flywheel-type devices are out of the picture—at least in that part of the drive chain following the clutch. Theoretically, the flywheel could be placed before the clutch. This has never proven practical, even in the largest model machines, as space and size are very important factors. The lack of any material flywheel effect is in itself a large distinction between the dictating machine and other recording and playback equipment. Yet keep in mind the fact that these machines both record and reproduce and that flutter and wow effects enter as a product rather than a sum.

The transcribing machine is quite different from a phonograph and even from the dictator's executive instrument. Fast start and stop is, if anything, even more important to the typist for she usually listens only to short portions while typing. If the machine stopped slowly each time or coasted past the desired stop point she would either miss words or have to backspace and repeat the last few words every time she restarted. Although she may do this automatically, it is an unsatisfactory expedient because of the time loss and the mental upset it occasions.

Most typists require a rather wide range of speed control on their equipment. Strange as it may seem, they usually set the playback speed quite a bit off from the recording speed. Often shrill voices are slowed down and gutterals increased in speed. Some girls like to type more or less in step with the recording, while others stop typing and listen to a short sentence. Further, most girls do not remember what they type nor do they think in terms of sentences while typing.

Speed ranges are controlled by a knob on the front of the machine and vary from 1.5 : 1 to 3 : 1. Usually the typist desires to be able to slow down and speed up the record over a greater range than she believes she will ever wish to use. For all practical purposes, a 2 : 1 range would satisfactorily take into account differences in dictators and recording machines.

The backspacing problem is a nice one to solve. The ideal would probably be a system connected to the girl's mind so that when she pressed a button she would hear a repeat of the exact word or words she wishes. This being highly improbable of attainment, the next best solution might be a dial to select the amount of repeat obtained. However, the words dictated per inch of groove vary greatly from dictator to dictator or even with one individual as he changes from grasping for the right word to reeling off a standard sentence or reading a section of prepared material. It is not probable that men's speaking and dictating habits can be changed, so the backspacers employed are of the types that give a fixed physical backward movement to the reproducer stylus. With around 250 lines per inch and "average" dictating speed of 100 words per minute, and an average repeat of 5 to 15 words, the stylus should move about 0.008–0.012 inches per repeat stroke, and it must move rapidly. Tone arms tend to have quite a bit of mass, even in the smallest machines, so it takes considerable power to move them rapidly. One company uses the drive motor, resulting in a repetitive stepping action; the others employ magnetically operated ratchet type drives. It might be noted in passing that the constant angular velocity disc, cylinder and belts maintain relatively constant word pickup per backspacing operation. The constant linear velocity disc with its varying r.p.m., as used by one manufacturer, will vary its word pickup per groove of backspacing between the inside and outside of the disc.

Because of the use to which dictation instruments are placed, simplicity, directness and flexibility of operational controls are important. At first thought, forgetting costs, the designer might lean in the direction of complete automaticity. The difficulty of such a system is that it is limited in use to the conditions for which it was designed, and attempts to use the equipment under other conditions often result in mechanical failure.

There are, however, two types of semiautomatic approaches that may be and are used. The first is a manual preventive; for example, interlocks prevent operating the lever that will lower the recorder stylus unless a record is in place. This type requires a sturdy mechanism. The other type is permissive preventive. In this case the lever could be operated normally, but an internal interlock would prevent the stylus from being lowered. A signal of some sort would indicate that the equipment is not ready for use. In both types the system must be completely restored by a normal, everyday act on the part of the user. In the examples cited, the normal act would be to put the recording disc in place.

This concludes the general discussion. Part 2 will describe the application of these principles to modern dictating equipment.

References

1. Harvey Fletcher, *Speech and Hearing*, D. Van Nostrand Co., 3rd printing, Jan. 1936.
2. H. K. Dunn and S. D. White, "Statistical measurements on conversational speech," *J. Acous. Soc. Am.*, 11 (Jan. 1940) 278.
3. D. K. Gannett and Iden Kerney, "The discernibility of changes in program band width," *Bell Sys. Tech. J.*, 23 Jan. 1944) 1–10.
4. Jensen Technical Monograph #4—*The Effective Reproduction of Speech.*—Jensen Radio Mfg. Co., 1944.

HEARING AID

(*from page 21*)

formers and a host of RC coupling networks, the "full" electrical response curve of the amplifier (*Fig.* 8) does not vary by more than 2.5 db from the mean over the frequency range of 400 to 7000 cps. In order to get this large number of components into the small space occupied by the amplifier, all of the capacitors are high-K ceramics except for the tantalum capacitor used to by-pass the bias resistor in the transistor stage. *Figure* 11 shows the over-all acoustic gain of the Sonotone 1010 when fitted with a U-1 receiver. The mid-frequency acoustic amplification is about 55 db and the maximum sound pressure level in the ear may be 125 db. This is sufficient to provide satisfactory hearing for individuals with a hearing loss as great as 75 db. This response may be modified by setting the fitting controls. Many individuals with small low-frequency loss and excessive high-frequency loss require more low-frequency attenuation than can be provided with the fitting controls. Additional frequency-response modifications are based on the selection of a receiver, as indicated by the curves of *Fig. 7*.

The battery complement required to operate the 1010 consists of a 1.25-volt mercury A battery and a 15-volt B battery. The A battery furnishes the current required to heat the filaments of

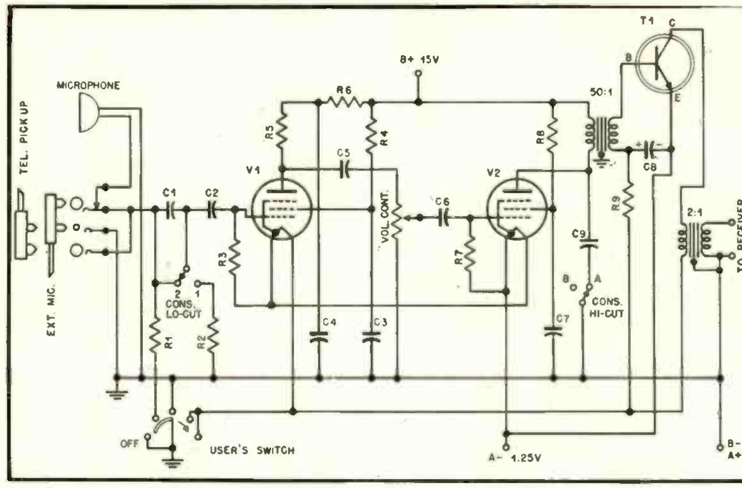


Fig. 9. Complete schematic of the 1010 hearing aid. Switches marked "Cons Lo-Cut" and "Cons Hi-Cut" are intended for adjustment by the hearing consultant. The wearer can change low-frequency response to some extent by means of the "User's Switch."

the gain and driver tubes, as well as the current for the transistor, a total of 13 ma. The B battery furnishes the current for the screens and plates of the gain and driver tubes—a total of 60 μa at full B voltage. Laboratory tests indicate a useful life of 65 hours for the A battery and 2000 hours for the B battery. With respective costs being 30 and 95 cents, this results in a total cost of ½ cent per hour. Even for fifteen hours use per day, the operating cost is only two dollars and a quarter per month—little enough for the advantage of good hearing.

Accessories

Figure 9 indicates additional items which may be plugged into the accessory receptacle on the side of the case. The external microphone is designed to be worn on the outside of the clothing and is attachable by means of a pin plug which pierces the clothing from the inside. The pins engage jacks on the back of the microphone, thus permitting the connection to be made to the instrument without visible wires. When the cable plug is inserted into the instrument jack, the internal microphone is disconnected from the circuit.

The telephone pickup is an inductor which is simply held near the receiver of a conventional handset. It is of relatively low impedance, and is equipped with a switch to open or close the circuit. The cord plug is inserted into the instrument jack, thus paralleling the pickup coil with the high-impedance crystal microphone. To use the telephone, the wearer closes the switch on the pickup coil—shorting the microphone—and the input signal is derived solely from the inductor. At the conclusion of the phone conversation, the switch is opened and the microphone is again operative. This has the advantage of completely eliminating outside noises during telephone conversations.

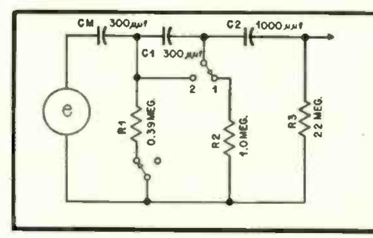


Fig. 10. Low-frequency control circuit.

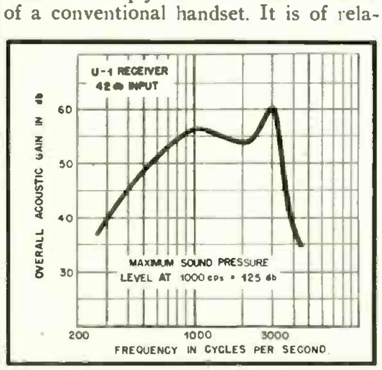


Fig. 11. Over-all acoustic gain curve from microphone to receiver output.

Conclusion

This discussion indicates how the requirements for an effective hearing aid are met by the Sonotone 1010, which was designed to integrate the advantages of the battery economy of the junction transistor with the inherent quietness of microminiature vacuum tubes. The result is a hearing aid about half the size of a pack of cigarettes, providing 55 db of acoustic amplification and capable of producing a sound pressure level of 125 db in the ear. In addition, a wide range of over-all frequency responses is made available for fitting a wide variety of hearing impairments.

Many members of the Technical Operations Division of Sonotone Corporation contributed to the successful development of this hearing aid. I am indebted to all of them for making it possible to prepare this report.

MULTIPLE-TRACK MAGNETIC HEADS

(*from page 34*)

TABLE 1

	Magnetic Head Number					
	1	2	3	4	5	6
10,000-cps response improvement in db due to perfect azimuth	+ 0.9	+ 1.0	+ 0.5	+ 0.1	+ 0.8	+ 1.6

modulation level at bias current established in (1). A 400-cps signal of a level of +1 dbm can be applied to the record head to obtain playback head output with a distortion content of 2.5 per cent.

(3) *Frequency characteristic.* The frequency characteristic was determined at a recording level of 10 db below 100 per cent modulation. We made use of our standard pre- and post-equalization frequency characteristics as currently employed in our PM-63 racks. Measurements were made by recording on cluster No. 1 and reproducing first on cluster No. 2 and then on cluster No. 1 itself. The difference in the frequency characteristics thus obtained can be ascribed to difference in slit azimuth between cluster No. 1 and cluster No. 2. In general the frequency characteristics of both cluster No. 1 and No. 2 were well within the limits set up for our standard magnetic heads used for single and triple track recording.

(4) *Determination of slit azimuth differences between the six heads in cluster No. 1.* This measurement was made by making frequency recordings with cluster No. 1 as recording head. The recordings were then reproduced first on cluster No. 2 and then on cluster No. 1. Since cluster No. 2 was constructed in such a manner that the azimuth deviation of all its component heads was the same as explained in detail in the first part of this paper, the difference in high-frequency output experienced between using cluster No. 2 and cluster No. 1 as reproducing heads, represents the azimuth difference of the individual heads of cluster No. 1. Table I indicates the improvements in 10,000-cps response that were obtained when playing back the recordings made on cluster No. 1 on itself, as compared to playback on cluster No. 2.

(5) *Cross-talk of adjacent heads.* The curves of Fig. 6 depict the signal-to-crosstalk ratio between adjacent heads. This measurement was made by recording frequencies at a level 10 db below 100 per cent modulation on head No. 3 of cluster No. 1 and No. 2. While recording on head No. 3, normal recording bias without signal was applied to the adjacent heads No. 2 and No. 4. Subsequently, heads No's 2, 3 and 4 were used as reproducing heads and the output levels obtained from the three adjacent tracks were noted. In *Fig.* 6, the line marked SIGNAL indicates the output from the track on which the various frequencies had been recorded. Below the signal reference line, two curves are shown which indicate the outputs that had been obtained when playing back the tracks adjacent to the originally recorded track. It can be seen that the crosstalk from cluster No. 2 is somewhat higher than the crosstalk from cluster No. 1, particularly at the high and low frequencies.

Shown also on the figure is the curve obtained when the 40-db equal loudness contour characteristic is superimposed on the crosstalk curve for cluster No. 2.

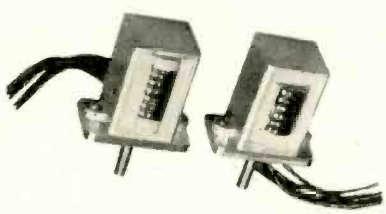


Fig. 5. The two clusters in completed form: left, No. 1, and right, No. 2.

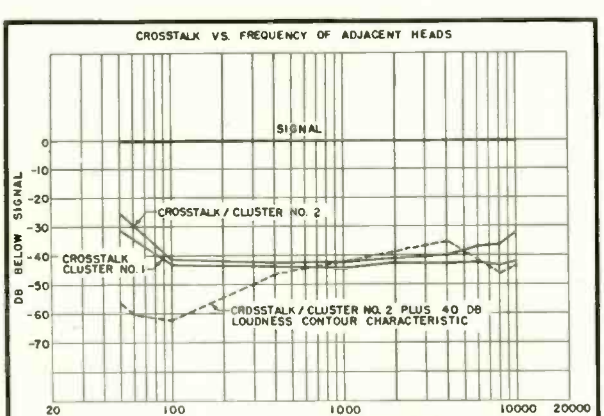


Fig. 6. Measured performance of clusters with respect to crosstalk between adjacent heads.

This curve shows that both the extreme lows and extreme highs are not audible when the crosstalk from cluster No. 2 is reproduced at such a level that its 1000-cps component is 40 db above threshold.

(6) *Determination of signal-to-noise ratio.* A signal-to-noise ratio of 57 db, referred to 100 per cent modulation output, was measured. For comparison purposes it should be mentioned that the signal-to-noise ratio on the PM-63 channel with MI-10795-1 magnetic heads lies between 55 and 60 db.

The crosstalk values shown on *Fig.* 6 appear more than adequate for motion picture presentation in auditorium-type theaters. This is based on the findings by Mueller,[1] which show that the useful volume range for dialog in a theater is of the order of 25 db, and that even for music this range never exceeds 35 db. In the case of review rooms, in which the audience noise is from 30 to 35 db above hearing threshold, compared to 40 to 45 db in auditorium-type theaters, the use of magnetic decouplers may prove desirable.

[1] W. A. Mueller, "Audience noise as a limitation to the permissible volume range of dialog in sound motion pictures," *J. Soc. Mot. Pict. Engrs.*, July, 1940.

A MONTHLY SUMMARY of product developments and price changes of radio electronic-television parts and equipment, supplied by United Catalog Publishers, Inc., 110 Lafayette Street, New York City, publishers of Radio's Master.
These REPORTS will keep you up-to-date in this ever-changing industry. They will also help you to buy and specify to best advantage. A complete description of most products will be found in the Official Buying Guide, Radio's Master—available through local radio parts wholesalers.

CORRECTION: Fisher Radio

In the July column, Model 50-C Control Chassis was erroneously reported as discontinued. Actually, new model numbers were assigned to the 50-C to designate new cabinet styles as follows: 50-CB (Blond wood finish); 50-CM (Mahogany wood finish); 50-CH (Chassis only). No changes were made in specifications.

Books and Manuals

SAMS & CO.—Addel No. AA-4 "Audio Amplifiers and Associated Equipment," Volume 4 at $3.95 net.

Recording Equipment, Speakers, Amplifiers, Needles, Tape

AMPEX ELECTRIC—Added Audio Magnetic Tape Duplicator; Series 350 Audio Magnetic Tape Recorder; Series 450 Audio Magnetic Tape Reproducer.

BERLANT ASSOCIATES—Discontinued Model 503 Console Tray for mounting Basic Recorder; Model 702 Console Cabinet.

FAIRCHILD RECORDING—Discontinued Model 201-B Three-Way Turret Head.

ORRADIO INDUSTRIES—Discontinued Paper Base "Irish" Tape No. 205RKA.

PERMOFLUX—Added Headset Adaptor Model BMA-1 at $2.25 net.

RECOTON CORP.—Added Model 850 Replacement Phoneedle Kit at $39.98 net.

ROCKBAR CORP.—Added Model 3/531 Three-Speed Non-Intermix Record Changer with 2 plug-in shells and mounting hardware, less cartridges at $40.87 net; Model 3/532 Three-Speed Intermix Record Changer with 2 plug-in shells and mounting hardware, less cartridges at $48.75 net; Model 3/534 Three-Speed Record Player with 2 plug-in shells and mounting hardware, less cartridges at $25.20 net.

TALK-A-PHONE—Increased price on Model LM-5 Master Station for 5 Sub-Stations to $34.80 net.

UNIVERSITY LOUDSPEAKERS—Added several new High-Fidelity Corner Enclosures for 12-in. Speakers. Increased price on Model PH Reflex Loudspeaker Projector to $18.60 net.

MOVING?
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MATCHED HI-FI LINE

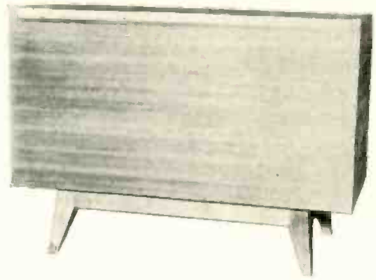
(*from page 31*)

frequency cone operates is reduced, thereby increasing the acoustical load imposed upon the small cone. At the same time, the conical diffusers increase the distribution angle due to a reduction in propagation velocity in the low-frequency cone and to diffraction of the sound emitted by the high-frequency unit. The damping ring in the outer suspension provides a terminating acoustical impedance which eliminates standing waves in the suspension and cone. As a result, the response frequency characteristic is very smooth and the response to transient sounds is very faithful.

Cabinets[15, 16]

The cabinets designed for these speakers are of the phase-inverter type. The walls are of heavy construction combined with bracing which minimizes vibration. The

internal damping is provided by diagonally placed sound absorbing material which is located away from the cabinet walls, placing it where the sound volume current is high and giving high absorbing efficiency. In this way effective damping of standing waves is obtained with a minimum of absorbing material. Excessive low-frequency absorption which would reduce the effi-



ciency on the low-frequency range is thereby obviated. The design of the cabinets incorporates a configuration which is functional from an acoustic diffraction standpoint and is esthetic from a styling viewpoint. While the cabinets are contemporary in spirit, they can be combined with virtually any decor. They may be used either horizontally or vertically.

An equipment cabinet, type SE-1, which matches the loudspeaker cabinets, is available in the same finishes. This cabinet will house a record changer, tuner, and preamplifier. These units may be mounted either horizontally or vertically depending upon mounting of the cabinet. The two cabinets are shown in *Figs*. 10 and 11.

Matched High-Fidelity System

The first two sections of this paper have outlined the characteristics of speech and music, and the characterists of a high-fidelity sound reproducing system which must be correlated with the characteristics of speech and music to obtain a specific set of data for the developement and design of the system. The third section of this paper has described the components of the RCA High-Fidelity System which have been developed and designed following the fundamental principles given in the first two main sections of this paper. There is one remaining consideration, namely, satisfactory operation of the various possible assemblages of components to form a variety of high-fidelity systems. In other words, it is not enough that all the elements each exhibit high-fidelity performance; the elements must be matched so that no loss in fidelity results when the units are used together to form a complete system. Therefore these units have been designed to operate together as a system. The units are provided with input and output impedances such that when connected together, according to simple instructions, all units are "matched." Volume controls provide correct inter-unit voltages at normal setting and with minimum noise levels. All grounds are carried through the system in a scientific manner so that "ground loops" which would cause hum and induce oscillation are eliminated.

[15] H. F. Olson, AUDIO ENGINEERING, Vol. 35, No. 11, p. 34.

[16] H. F. Olson, *Radio and Television News*, Vol. 45, No. 5, p. 53, 1951.

COMING EVENTS

August 19–21—1953 WESCON (WESTERN Electronic Show and Convention), Civic Auditorium, San Francisco, California.

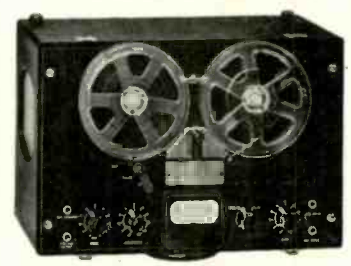
September 1–3—INTERNATIONAL SIGHT AND SOUND EXPOSITION, combined with the CHICAGO AUDIO FAIR. Palmer House, Chicago, Ill.

September 13–16—THE ELECTROCHEMICAL SOCIETY, INC., Sessions on corrosion, electrodeposition, and batteries. Ocean Terrace Hotel, Wrightsville Beach, N. C.

September 25–27—NORTHERN CALIFORNIA AUDIO SHOW. Palace Hotel, San Francisco.

October 14–17—Fifth Annual Convention of the AUDIO ENGINEERING SOCIETY, and THE AUDIO FAIR. Hotel New Yorker, New York City.

November 3–4—THIRD ANNUAL HIGH-FIDELITY CONFERENCE and AUDIO SHOW. Benjamin Franklin Hotel, Philadelphia.



BREAD-PAN LAYOUT

(from page 25)

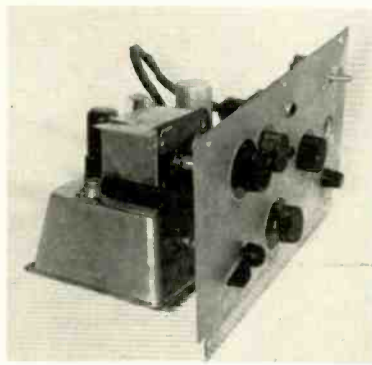


Fig. 11. Another view of bread-pan amplifier arranged as a panel-chassis unit.

hind a panel, with shaft extensions used to bring out the controls mounted in the pan-unit (*Figs.* 10 and 11). A power switch and pilot lights were added on the panel, and the electron-ray tube connections carried thru a short tube-base, tube-socket extension cable. Evidence of changes during construction is seen in the unused holes and empty sockets in the power amplifier and oscillator chassis. The complete unit gives no superficial suggestion of its unorthodox construction and in performance it equals that of the development layout—it *is* that layout. The application of the bread-pan layout to a new job will be found easy and effective, as well as economical of time, energy, and money—and the layout will end up in a form usably permanent and easily translated into a more conventional setup for production or other application if required.

Fig. 12. Complete recording unit is conventional in appearance.

PROFESSIONAL DIRECTORY

Industry Notes . . .

Purchase of **Measurements Corporation**, Boonton, N. J. by **Thomas A. Edison, Inc.**, to be operated as an **Edison** subsidiary, caught entire industry by surprise. Management personnel will remain unchanged with Harry W. Houck, former **Measurements** president, becoming vice-president and general manager of the subsidiary, with Jerry Minter, vice-president, and John M. van Beuren, former chairman of the board, serving as vice-presidents. Harry G. Riter, 3rd, takes over the presidency of **Measurements** in addition to being president of **Edison**.

Correct name of the new transformer manufacturing company resulting from the merger of **Tetrad** and **Triad** is **Triad**—not **Tetrad** as delineated in this space a couple of issues back . . . **Raytheon** is first major manufacturer to announce price reduction on stock model transistors—made possible by large volume production . . . **RCA Victor Division** announces construction of new component plant at Findlay, Ohio, located 50 miles southwest of Toledo—the new plant will be RCA's fourth manufacturing center in the Buckeye state . . . Newest location in **Graybar Electric Company's** nationwide distribution network will be Green Bay, Wis., with complete office, display, and warehouse facilities slated for opening on or about August 17.

Industry People . . .

Personnel changes at David Bogen Company, New York, bring **Vinton K. Ulrich**, formerly with National Union Radio Corporation, in as general sales manager, replacing **W. Walter Jablon**, resigned, and promote **Mortimer Sumberg** to distributor sales manager . . . **Robert P. Lamons** has been upped to sales manager of the Andrew Corporation, Chicago—succeeded as regional sales engineer by **Robert C. Bickel** who headquarters in Ridgewood, N. J. . . . **Brewster Freifeld** named to head up new Chicago office of Audio Devices, Inc. —**Bryce Haynes**, A-D sales vice-president, in announcing new office, stated that the Windy City has become "the third largest market in the country for sound recording materials."

Election of **W. Walter (Wally) Watts** as vice-president in charge of Technical Products, and of **Theodore A. Smith** as vice-president in charge of The Engineering Products Department of the RCA Victor Division, announced by **Walter A. Buck**, vice-president and general manager of the Division . . . **Maj. Gen. Edmond H. Leavey, U. S. A. (Retired)** is newly-elected president of Federal Telecommunication Laboratories, Inc., Nutley, N. J.

Promotions in the sales organization of the Gates Radio Company, Quincy, Ill., find **Lawrence J. Cervone** in as sales manager. Formerly head of the Gates New York sales office, **Cervone** is replaced by **Ed Wilder** in the company's Eastern office . . . **H. H. Seay**, vice-president and general sales manager, Bell Sound Systems, Inc., Columbus, Ohio, receiving congratulations on recent election to board of directors—similarly honored was **R. E. Grace**, Bell's assistant secretary . . . **William A. Ready** has retired as chairman of the board of The National Company, to be succeeded by **Raymond C. Cosgrove**, formerly president of the RTMA . . . After an absence since 1948, **Hugh J. Daly** has returned to Eicor, Inc., as sales manager of the tape recorder division.

New additions to the board of directors of Audio & Video Products Corporation, New York, include: **Kenneth B. Boothe, Martin V. Kiebert, Jr., Bernard B. Smith, and Irving M. Buckley** . . . **George V. Rosenquist** and **Robert C. Jackson** have been added to the management staff of J. C. Warren Corporation, manufacturers of tape recording equipment, Freeport, N. Y.—**Rosenquist** was formerly an executive of Lee Spring Company, while **Jackson** was assistant publisher of Quick magazine . . . **Russell J. Tinkham** has been appointed manager of audio sales by Ampex Corporation—will headquarter in company's Redwood City factory. Formerly head of Ampex's midwestern sales district with headquarters in Chicago, he will be replaced by **E. L. Koller**, until recently manager of field engineering, who in turn will be replaced by **J. L. Stultz**.

ADVERTISING INDEX

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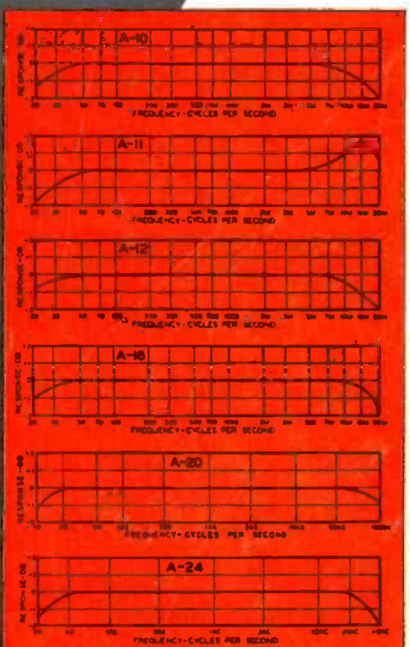
True hum balancing coil structure combined with a high conductivity die cast outer case, effects good inductive shielding.



Type No.	Application	Primary Impedance	Secondary Impedance	List Price
A-10	Low impedance mike, pickup, or multiple line to grid	50, 125/150, 200/250, 333, 500/600 ohms	50 ohms	$16.00
A-11	Low impedance mike pickup, or line to 1 or 2 grids (multiple alloy shields for low hum pickup)	50, 200, 500	50,000 ohms	18.00
A-12	Low impedance mike pickup, or multiple line to grids	50, 125/150, 200/250, 333, 500/600 ohms	80,000 ohms overall, in two sections	16.00
A-14	Dynamic microphone to one or two grids	30 ohms	50,000 ohms overall, in two sections	17.00
A-20	Mixing, mike, pickup, or multiple line to line	50, 125/150, 200/250, 333, 500/600 ohms	50, 125/150, 200/250, 333, 500/600 ohms	16.00
A-21	Mixing, low impedance mike, pickup, or line to line (multiple alloy shields for low hum pickup)	50, 200/250, 500/600	200/250, 500/600	18.00
A-16	Single plate to single grid	15,000 ohms	60,000 ohms, 2:1 ratio	15.00
A-17	Single plate to single grid 8 MA unbalanced D.C.	As above	As above	17.00
A-18	Single plate to two grids. Split primary.	15,000 ohms	80,000 ohms overall, 2.3:1 turn ratio	16.00
A-19	Single plate to two grids. 8 MA unbalanced D.C.	15,000 ohms	80,000 ohms overall, 2.3:1 turn ratio	19.00
A-24	Single plate to multiple line	15,000 ohms	50, 125/150, 200/250, 333, 500/600 ohms	16.00
A-25	Single plate to multiple line 8 MA unbalanced D.C.	15,000 ohms	50, 125/150, 200/250, 333, 500/600 ohms	17.00
A-26	Push pull low level plates to multiple line	30,000 ohms plate to plate	50, 125/150, 200/250, 333, 500/600 ohms	16.00
A-27	Crystal microphone to multiple line	100,000 ohms	50, 125/150, 200/250, 333, 500/600 ohms	16.00
A-30	Audio choke, 250 henrys @ 5 MA 6000 ohms D.C., 65 henrys @ 10 MA 1500 ohms D.C.			12.00
A-32	Filter choke 60 henrys @ 15 MA 2000 ohms D.C., 15 henrys @ 30 MA 500 ohms D.C.			10.00

TYPE A CASE
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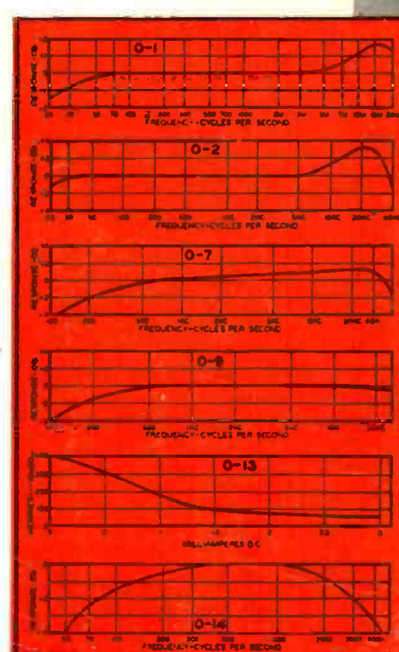
UTC OUNCER components represent the acme in compact quality transformers. These units, which weigh one ounce, are fully impregnated and sealed in a drawn aluminum housing ⅞" diameter...mounting opposite terminal board. High fidelity characteristics are provided, uniform from 40 to 15,000 cycles, except for 0-14, 0-15, and units carrying DC which are intended for voice frequencies from 150 to 4,000 cycles. Maximum level 0 DB.

OUNCER CASE
⅞" Dia. x 1⅛" high

Type No.	Application	Pri. Imp.	Sec. Imp.	List Price
0-1	Mike, pickup or line to 1 grid	50, 200/250 500/600	50,000	$14.00
0-2	Mike, pickup or line to 2 grids	50, 200/250 500/600	50,000	14.00
0-3	Dynamic mike to 1 grid	7.5/30	50,000	13.00
0-4	Single plate to 1 grid	15,000	60,000	11.00
0-5	Plate to grid, D.C. in Pri.	15,000	60,000	11.00
0-6	Single plate to 2 grids	15,000	95,000	13.00
0-7	Plate to 2 grids, D.C. in Pri.	15,000	95,000	13.00
0-8	Single plate to line	15,000	50, 200/250, 500/600	14.00
0-9	Plate to line, D.C. in Pri.	15,000	50, 200/250, 500/600	14.00
0-10	Push pull plates to line	30,000 ohms plate to plate	50, 200/250, 500/600	14.00
0-11	Crystal mike to line	50,000	50, 200/250, 500/600	14.00
0-12	Mixing and matching	50, 200/250	50, 200/250, 500/600	13.00
0-13	Reactor, 300 Hys.—no D.C.; 50 Hys.—3 MA. D.C.,		6000 ohms	10.00
0-14	50:1 mike or line to grid	200	½ megohm	14.00
0-15	10:1 single plate to grid	15,000	1 megohm	14.00



United Transformer Co.

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