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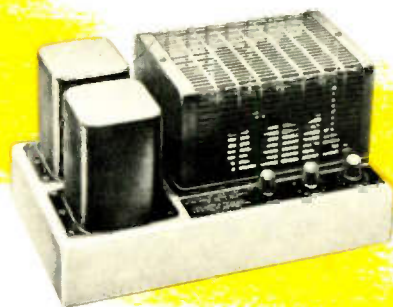
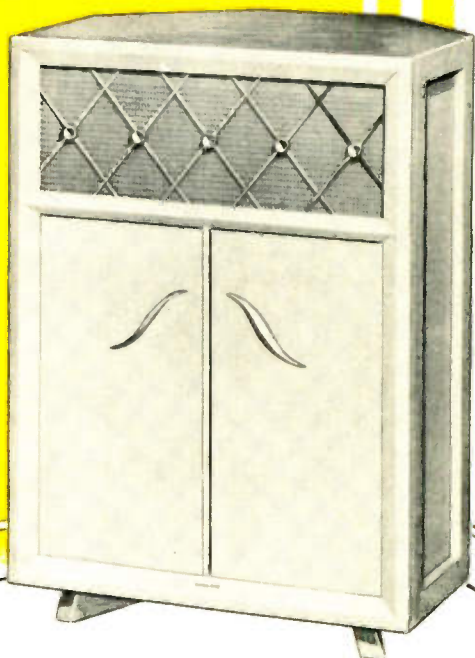




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A selection
of articles
from



FOREWORD

For many years, the editors of RADIO & TELEVISION NEWS have been dedicated to the task of publishing top-flight articles on hi-fi in as great a volume as space would permit. This policy has resulted in the publication of a large quantity of audio material and has established RADIO & TELEVISION NEWS as the leading publication in this field.

Because of the tremendous and growing interest in hi-fi, it was felt that a publication combining the best of the material published in the last couple of years within the covers of a single volume would be welcomed by hi-fi enthusiasts everywhere. This book is the result. It consists of reprints of carefully selected articles chosen for their individual merit and assembled into chapters for ready reference. Sufficient variety is included to satisfy the interests of practically any hi-fi fan, particularly one desiring to build his own equipment.

The material is divided into six chapters. First is a chapter on preamplifiers which includes a number of different designs of varying complexity. Next comes tone-controls and equalizers, presenting information on equalizing for practically every recording curve ever used, including the new AES curve. There is even an article on mixers in this chapter! Chapter 3 contains details on a number of different amplifier designs with power outputs to suit individual needs. The final stage of the audio system, loudspeakers and enclosures, is covered in chapter 4, again with a wide variety of designs included. In chapter 5, information and advice is presented on how to intelligently select the various components of a hi-fi system. Chapter 6 has been reserved for a miscellaneous assortment of audio material which did not fit logically in any of the other chapters.

We sincerely hope that you enjoy this book and urge you to read RADIO & TELEVISION NEWS regularly to keep up-to-date on the latest in hi-fi equipment and techniques.

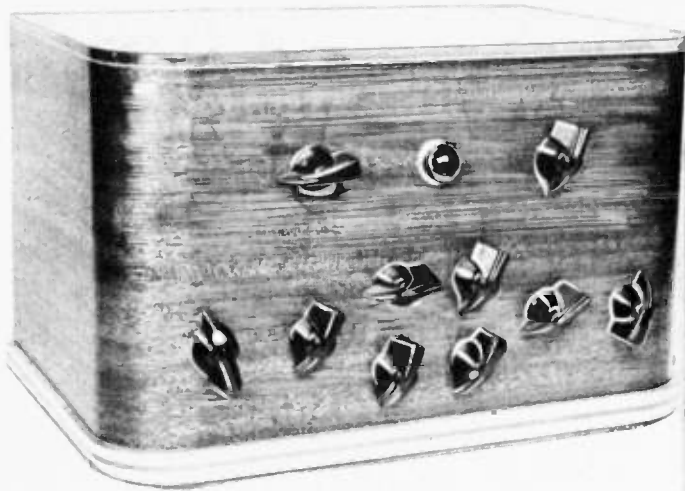
THE EDITORS

CHAPTER 1

PREAMPLIFIERS

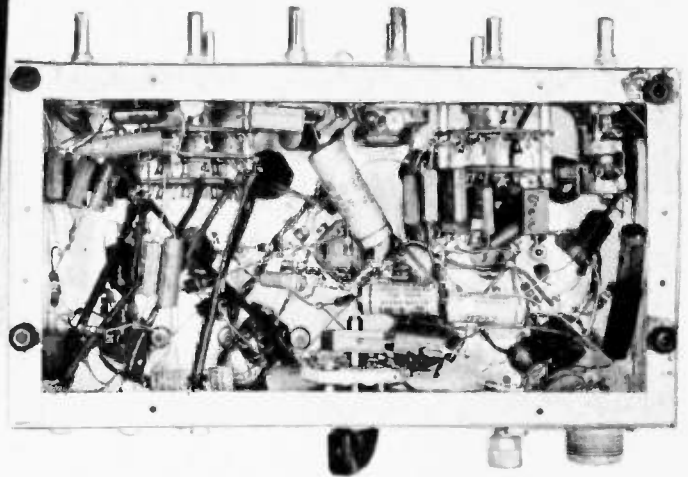
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A TONE-COMPENSATING PREAMP



Author housed his preamp in a standard wood cabinet.

Chassis measures 5 x 9½ x 3 inches over-all.



By JOHN H. DANIEL

This design incorporates, on a single chassis, almost every conceivable tone compensating network. It includes separate circuits for bass and treble controls, additional bass boost and sharp cut-off high-frequency attenuator, and loudness control.

THE high-fidelity enthusiast who builds his own equipment can find numerous articles to guide (or confuse!) him in (1) selecting a suitable design for a high-quality audio amplifier, (2) converting the electrical output of such an amplifier into relatively undistorted (or at least pleasing) sound, and (3) choosing a preamplifier with appropriate equalization curves to be used with magnetic phonograph pickups. For a summary of the three general types of circuits employed, see "A Preamp for Magnetic Pickups," by William Creviston, *RADIO & TELEVISION NEWS*, December 1952. While such a combination of an equalizing preamplifier with a flat-frequency-response amplifier and a well designed speaker-baffle system may possibly give all that is desired for the playing of records, it is probably a rare individual who does not desire further tone adjustment and control, even in this case. Such a combination almost certainly will not satisfy a discriminating listener when used (minus the equalizing preamp, of course) with various AM or FM radio programs, or with TV sound.

To obtain the additional tone variation desired, one can refer to numerous articles on the general principles of, and special circuits for, bass and treble "boosts" and "cuts." However, it appears that somewhat less attention has been devoted to the problem

of integrating these ideas and circuits into a satisfactory over-all system.

The system presented in Fig. 1 is a preamplifier designed to work from the equalizing preamp of the record player, the AM or FM tuner, or the TV discriminator output, into the input of the audio amplifier. It is not intended to be the simplest arrangement that might give a desired tone control, but was made as simple as possible while still furnishing nearly every type of control for which the author has heard a desire expressed by several high-fidelity experimenters. It can be said to be "versatile" not only because of the large number of tonal effects which may be incorporated, but also because it furnishes a basic framework which lends itself to the substitution of alternative circuits to accomplish certain results, as will be discussed in the section on "Modifications."

Features

The schematic of Fig. 1 incorporates the following general features:

1. "Primary" tone controls consisting of continuously variable amounts of bass and treble "boosting" or "cutting" on either side of a middle audio frequency (800 cycles-per-second), with flat frequency response resulting when all controls are in "zero" position.

2. Additional tone controls consist-

ing of a bass boost and a treble boost which can be switched in at any desired frequency (or frequencies, resulting in several steps of boost) and for various amounts.

3. A separate gain control and loudness control, allowing the former to be set for various inputs so that the latter operates on the proper Fletcher-Munson curve, and avoiding the likelihood of saturation in any stage for high level inputs.

4. A sharp treble cut which may be set at various frequencies, as desired, to eliminate record scratch or high frequency noise.

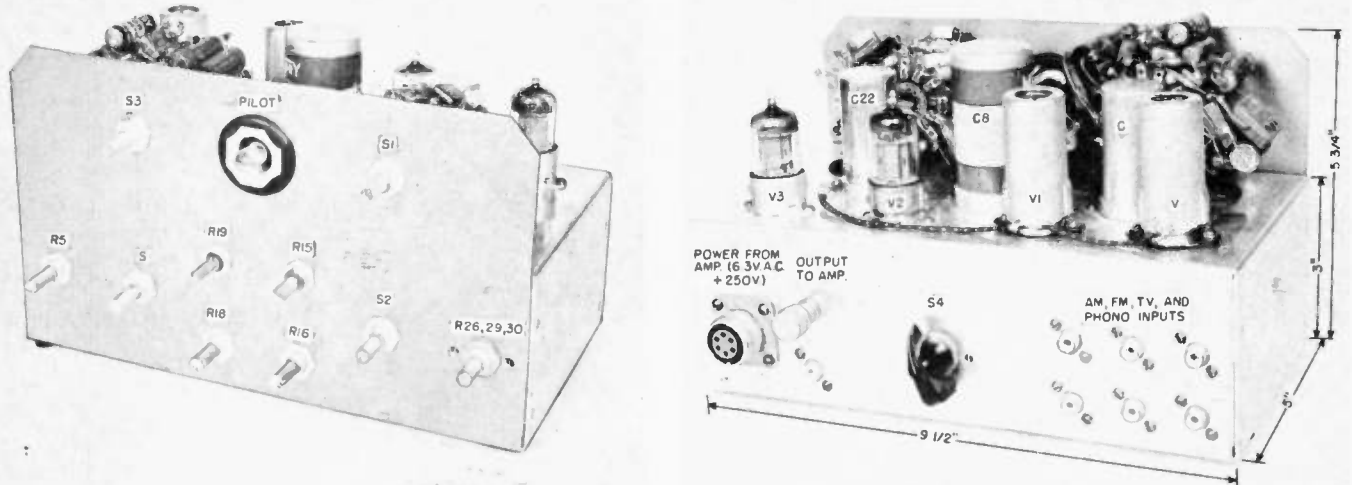
5. A sharp bass cut to be used with record players or programs having excessive turntable rumble or 60 cycle hum.

Primary Tone Controls

The primary tone controls (Feature 1) are realized by the network between the two halves of the 12AU7 tube, V_2 , as indicated in Fig. 1. Because the impedance of this network is low for certain frequencies at certain potentiometer settings, it is driven by a low-impedance cathode-follower source which, with its high input impedance, serves as a buffer between the network and the high impedances associated with the output of the 5879 tube, V_1 . The frequency response curves for this section are shown in Fig. 3. The four potentiometers, one each for bass boost, bass cut, treble boost, and treble cut, are arranged so that the boosts turn clockwise from the zero or flat position, and the cuts counterclockwise from that position.

Additional Tone Controls

The adjustable treble boost of Feature 2 is accomplished by shunting networks such as R_1 , R_2 , C_1 , C_2 (Fig. 1) across the 0.47 megohm resistor R_{10} . With no shunt, this resistor, in series with R_{11} , cuts the gain of the preamp



Front and rear views of the tone-compensating preamp. See Fig. 1 for identification of controls and parts.

to about a fifth. Shunting R_{10} partially restores the gain for those frequencies for which the shunting impedance is low. Examples of the effect of several network parameters R_1 , R_2 , C_1 , and C_2 are shown in Fig. 2, curves 1-5.

The adjustable bass boost of Feature 2 is accomplished by feedback networks R_3 , R_4 , C_3 , C_4 , etc. (Fig. 1) which feed the output of the second half (V_{2n}) of the 12AX7 back to its grid input. Its operation may be understood if it is remembered that the gain of this stage is approximately equal to the ratio of the total feedback impedance between plate and grid to the resistance R_{15} . For flat response, network R_3 , C_3 , R_4 , C_4 becomes a direct short (position 1 of switch S_3), giving a constant gain and a low output impedance of approximately 3000 ohms. This low output impedance is desirable if the preamplifier is to be cabled any distance to the amplifier proper, both from the standpoint of reducing 60- or 120-cycle pick up, and of preserving the high frequencies.

To boost the bass below a given frequency, a network, R_3C_3 , having a higher impedance below this frequency, is switched in place of the short. The higher impedance results in less feedback and therefore greater gain below this frequency. Examples of several choices of parameters R and C are given in the frequency curves of Fig. 2, curves 6-14.

Feature 2 furnishes an ideal means of properly equalizing the acoustic output between a woofer and a tweeter. It is not unusual for the high damping factor of a good amplifier-speaker system to be nullified to a large extent (usually for the tweeter) by the insertion of a resistance-type equalizer in the dividing network for the speakers. Feature 2 avoids this by making an equalizer in the dividing network unnecessary. (Theoretically, our method, since it uses essentially a

single element in the feedback loop to produce a 6 db-per-octave slope, is perfectly suited only to a dividing network having a 6 db-per-octave roll-off, such as is produced by a single capacitance in series with the tweeter and inductance in series with the woofer. In practice, however, it seems to work equally well with the 12 db-per-octave networks, probably because the transition region is small, and the departure from flatness within it usually less than 2 db for the equalizations normally required), besides providing additional possibilities for obtaining tonal effects at frequencies other than the one selected for the primary tone control of Feature 1 (and other than the crossover frequency of the dividing network).

Loudness and Gain Controls

The loudness control shown between V_2 and V_3 in Fig. 1 is the one advocated by Johnson¹ with the addition of a 0.1 megohm resistor R_{15} . This addition gives less emphasis on the high frequencies at low volumes, which seems desirable, at least in the author's opinion.

To use the loudness control properly, the operator must associate a given position of the control with its corresponding (and unique) loudness level, preferably choosing the position corresponding to the level at which he is accustomed to listen. With the control in this position, the gain control R_5 is varied to obtain this accustomed loudness level. From this point on, any desired "loudness" change is made with the loudness control, so that proper tonal balance is maintained at all levels.

It is quite properly argued that such tonal balance can usually be maintained by adjustment of the primary tone controls; however, this requires (for finicky listeners) a new "experimental" balance each time the loud-

ness is changed, instead of the one initial balance. Occasionally also the range of the primary controls may be found insufficient to take care of both the primary tone adjustment and the Fletcher-Munson effect.

Sharp Treble Cut

When record scratch or high-frequency noise is reduced by means of a treble cut-off (Feature 4) such as is introduced by switch S_1 in Fig. 1, the sharper the cut-off the more effective the noise elimination before it interferes with quality of reproduction. The high plate impedance of the 5879 pentode as a constant-current source is somewhat more effective in producing a steep slope than a triode would be. In addition, a two-section, low-pass filter is used. The frequency characteristics are given in Fig. 2, curves 15-17.

Sharp Bass Cut

The bass cut (Feature 5) introduced by switch S_2 in Fig. 1 is obtained by means of a two-section, high-pass filter whose characteristics are given in Fig. 2, curves 18 and 19. Should the input impedance of the amplifier driven by this preamp be other than 0.47 megohm, the 0.47 megohm resistor R_{10} of the filter should be changed to equal it. The cut-off frequency, of course, varies inversely as the product of this resistor and the value of the two equal condensers switched in.

The undesirable hum or rumbling which would lead to the use of this feature should, if possible, be eliminated at their sources. However, the expenditure required to produce an amplifier and speaker system which reproduces well at such low frequencies may sometimes be such that the financial limit has been reached without including the considerable amount necessary to replace a hitherto perfectly acceptable record player or

tuner with one meeting the new and more stringent requirements of extended low-frequency coverage. In addition, there are, at times, radio and TV programs with objectionable hum over which the listener has no other means of control.

Layout

The finished unit is shown in the photo on page 57. In its final form the unit includes an equalizing pre-amplifier² for reluctance pickups and a selector switch (for various equalization curves as well as different inputs) in addition to the elements of the circuit of Fig. 1. The plate supplies of 130 and 160 volts for the first and second halves, respectively, of the pre-amp tube were obtained by cascade decoupling filters from the 180 volt supply of the 5879 tube (V₁).

A chassis measuring 5"x9½" allowed easy access to all components with a soldering iron when metallized condensers were used for values of .05 to .25 µfd. Paper condensers would make access more difficult.

With usual judicious placement of leads, no compartmental shielding was found necessary if the input leads to the equalizing preamp and to the 5879 grid were shielded, and tube shields used on the preamp and 5879 tubes. The 250-volt plate and 6.3 volt filament supplies were taken from the main audio amplifier. With the filament supply connected as shown in Fig. 1, 60-cycle hum can be adjusted to a negligible value.

Modifications

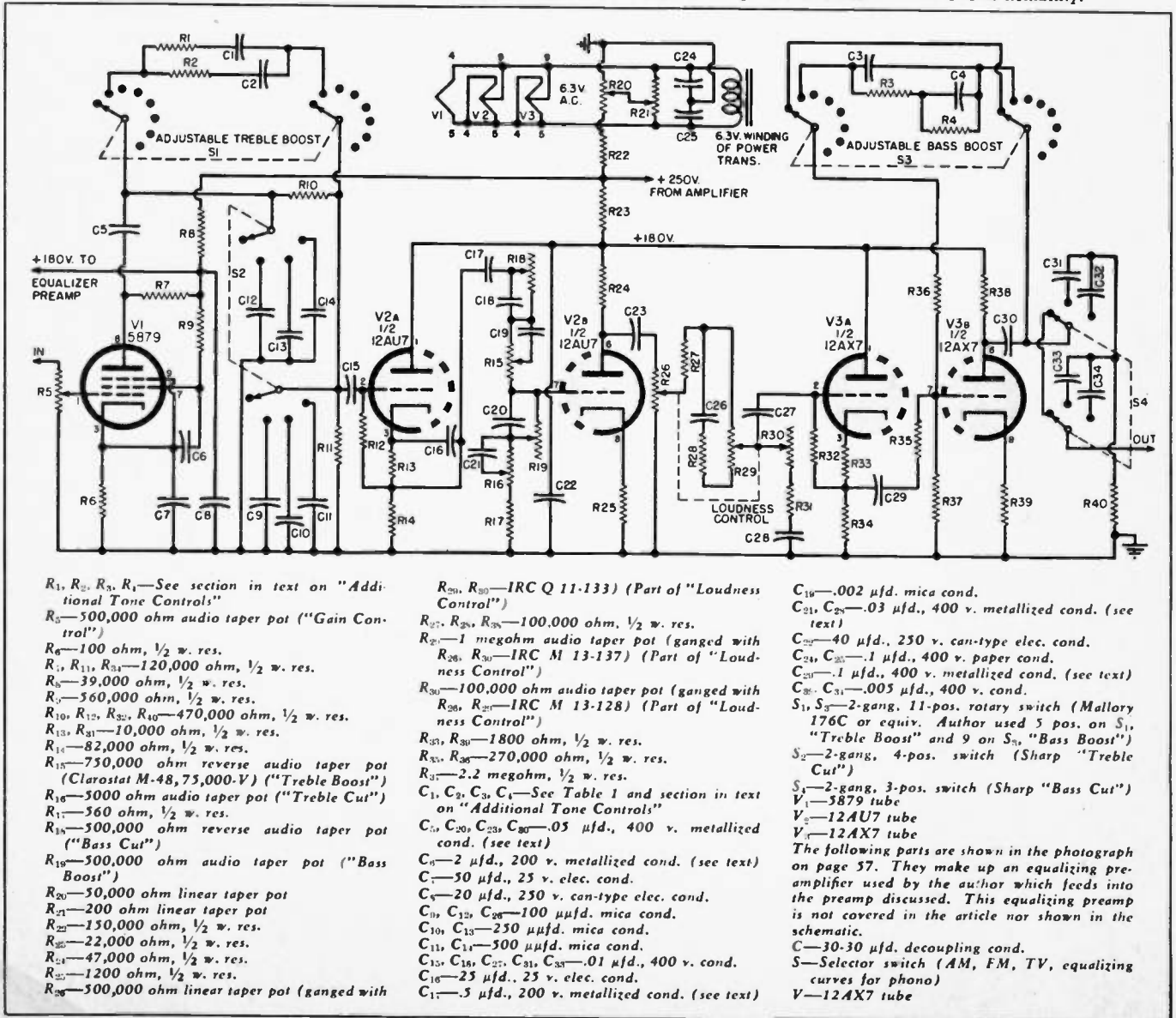
It is a fair assumption that no two people will agree on what effects are desirable from a preamplifier as described here nor the best way to produce any effect they do happen to agree upon as desirable. Consequently, the preamp has been designed to allow substitutions of other types of circuits with minimum disruption.

For example, if it is desired to use a different circuit for obtaining the Fletcher-Munson curves than that shown in Fig. 1, it can undoubtedly be inserted as a direct replacement, since

it is driven by a fairly low impedance and works into the high impedance of a cathode follower.

Similarly, if it is desired to use a primary tone control which has bass and treble controls in series, with each driven separately by a triode section, the two sections of the 12AU7 may be used for this purpose. The Fletcher-Munson circuit may then be inserted between the two halves of the 12AX7. If it requires more than the 0.27 megohm impedance, R₃₅, shown in Fig. 1, this can easily be obtained by increasing the value of both R₃₇ and R₃₈ to the desired impedance, and increasing the grid resistor R₃₁ by the same factor (or not less than ½ this factor). This will, of course, require larger values of R₃R₄, etc., and smaller values of C₃C₄, etc., by the same factor. It may also be desirable to insert between resistors R₃ and R₃₈ a condenser of appropriate value (such that the product of its capacity and resistor R₃₈ gives a time constant of at least 0.025 second) to avoid possible leakage of plate voltage to the grid.

Fig. 1. Circuit diagram and parts list covering the tone-compensating preamplifier. It offers unusual circuit flexibility.



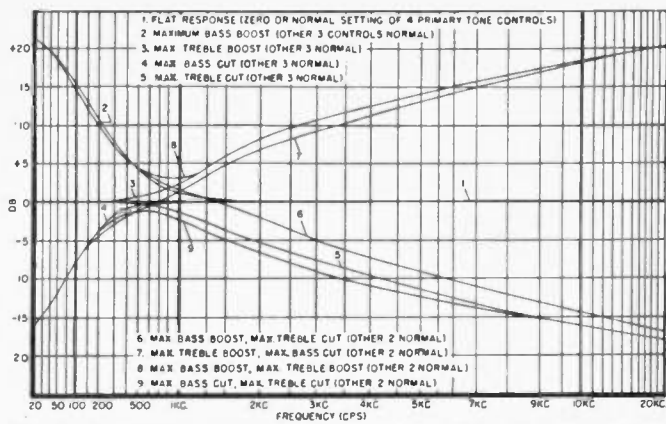
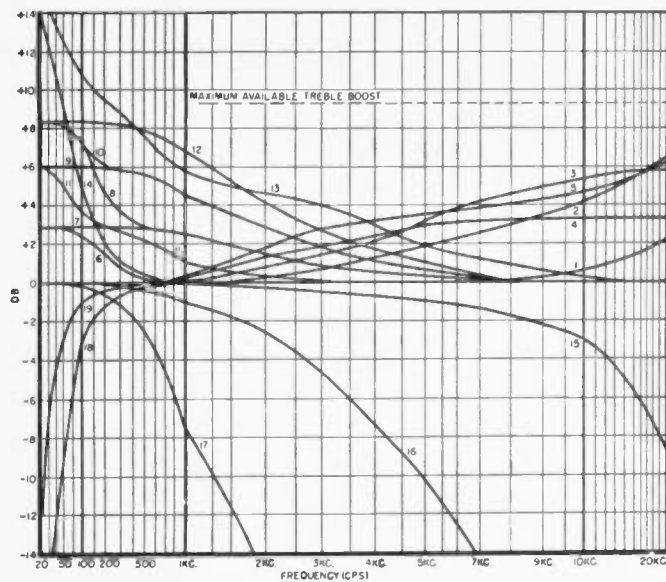


Fig. 3. Frequency response characteristics of primary tone control network. Network shown between halves of 12AU7, V_1 , in Fig. 1.

Fig. 2. Response curves of the additional controls. The curves correspond to the network component values shown in Table 1.

It might be pointed out that the primary tone control scheme of Fig. 1, if it is considered to include the entire 12AU7, has the advantage over any scheme with separately driven bass and treble circuit elements of being able to drive another type of circuit (such as the Fletcher-Munson) directly from its output because of its constant and comparatively low output impedance. It has the possible disadvantage of requiring a cathode bias condenser and a $0.5 \mu\text{fd}$. coupling condenser. Both schemes are equivalent in providing roughly unity gain at the mid-frequencies.

For those who desire step rather than continuous controls, no change in basic design is necessary; the burden of the problem is to provide switching facilities with the desired resistance steps. A simpler means for obtaining essentially the same result is to use calibrated dials on the panel so that the continuous controls can be set accurately and reproducibly to the desired step value.

To effect a slight economy, a type 6AU6 may be used for the pentode in place of the 5879, using circuit component values recommended in the RCA "Receiving Tube Manual," and maintaining the ratios of the resistances in the two-section, low-pass filter to the plate resistor.

For those who have a satisfactory equalizer for their woofer-tweeter dividing network, or who are satisfied with a single speaker, and who do not feel the need for frequency control of bass boost other than that provided by the primary tone control, a simple arrangement is given by merely omitting the 12AX7. The price of the treble boost control and the treble sharp cut-off is that of the RC components and switches, since no tubes can be omitted by leaving them out, although an increase in gain (which is not needed) may be obtained.

It is worth noting that the type of primary tone control used in Fig. 1, when considered as not including the second half of the 12AU7, still has an output impedance not exceeding 5600

ohms for the treble and 75,000 ohms for the bass (except for bass cuts). This is suitable for driving the Fletcher-Munson circuit directly. Thus, probably the most economical version of a preamp with all the features listed would have the primary tone control driving the Fletcher-Munson circuit, which then drives the last half of the 12AU7 in a circuit like that of the last half of the present 12AX7. Alternatively, the 12AX7 can be used with its first half connected as a cathode follower to drive the tone control. This version has less gain than the present arrangement. If more gain were necessary for inputs other than the record player, it could be provided by a

switching arrangement which would place these inputs on the equalizing preamp of the record player and provide it with a flat characteristic. Or, the gain could be increased for the record player as well by omitting the additional treble boost feature and using a single-section filter for the sharp treble cut-off.

Try constructing this unit for good, reliable performance.

REFERENCES

1. Johnson, E. E.: "A Continuously Variable Loudness Control." *Audio Engineering*, December 1950
2. Goodell, John D.: "Problems in Phonograph Record Reproduction." *RADIO & TELEVISION NEWS*, November 1950

Table 1. Component values for the additional tone controls and bass and treble sharp cut-offs. Refer to Fig. 2 for the various response curves obtained from the above listings (Curves 1-19). Only one set of values for R_1 , R_2 , C_1 , C_2 and R_3 , R_4 , C_3 , C_4 are shown in the schematic diagram of Fig. 1. Additional controls may be added at the discretion of the builder. Switches S_1 and S_2 have added contacts as shown in the diagram and may be used for this purpose. The condensers involved in C_9 , C_{12} , C_{10} , C_{13} , C_{11} , C_{14} , and C_{31} , C_{33} , C_{32} , C_{34} are intended to be substituted for those used by the author. They can be added if the switches S_2 and S_3 are changed to provide the additional contacts required.

CURVE	R_2	C_2	R_1	C_1		
1	0	18 μfd .	open			
2	0	50 μfd .	open			
3	100,000	75 μfd .	open			
4	560,000	75 μfd .	open			
5	560,000	75 μfd .	0	50 μfd .		
	R_3	C_3	R_4	C_4	R_n	C_n
6	120,000	.01 μfd .	0		0	
7	120,000	.001 μfd .	0		0	
8	120,000	.001 μfd .	390,000	.003 μfd .	0	
9	270,000	.0005 μfd .	0		0	
10	270,000	.0005 μfd .	220,000	.009 μfd .	0	
11	120,000	.0025 μfd .	180,000	.016 μfd .	0	
12	470,000	.0003 μfd .	0		0	
13	270,000	.002 μfd .	560,000	.001 μfd .	1.2 meg.	.005 μfd
14	1.8 megohm	.005 μfd .	0		0	
	C_9, C_{12}	C_{10}, C_{13}	C_{11}, C_{14}			
15	.0001 μfd .					
16		.0005 μfd .				
17			.002 μfd .			
	C_{31}, C_{33}	C_{32}, C_{34}				
18		.005 μfd .				
19	.01 μfd .					

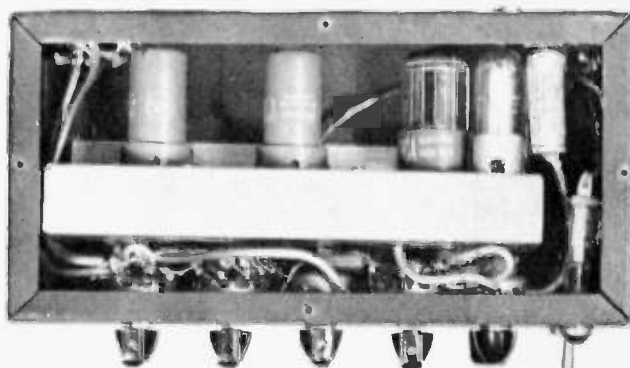
R_n and C_n is an additional parallel combination in series with R_1 .

A SIMPLE—YET GOOD QUALITY—PREAMP



By **ALEXANDER FINLAY**
Battelle Memorial Institute

Front and bottom views of the assembled preamplifier. The chassis measures a mere 5" x 10" x 3" over-all.



Details on a well designed unit for use with magnetic pickups. Amplification of 46 db is provided in preamp.

THE preamplifier in a modern high-fidelity installation is called upon to perform several very important tasks. It is required to supply the amplifying system with most of its voltage gain. It is required to supply the proper gain vs frequency characteristics of the program material, and it must act as the control center for the entire high-fidelity system.

The usual power amplifier in use today requires about one to two volts input for full output. The usual magnetic transducer or pickup used in these installations has an output which may be as low as ten millivolts. This means that in order to be suitable for use with any of the popular types of magnetic pickups, an amplification of about 46 db or 200-to-1 is required from the preamplifier.

The gain vs frequency characteristic of the preamplifier must complement the output vs frequency characteristic of the record and magnetic pickup combination. The usual practice is to record in such a way that the output from a magnetic pickup is proportional to frequency from the lowest recorded frequency to a frequency of about 500 or 800 cycles. The output of the pickup is normally constant with respect to frequency from 500 or 800 cycles to a frequency of about 1590 or 2500 cycles. Above 1590 to 2500 cycles the output from the magnetic pickup is again proportional to frequency. The equalization system of the preamplifier must be such that the over-all frequency response from record to power amplifier input is flat or constant throughout the audible frequencies.

In order to accomplish the desired result, the preamplifier must have a gain inversely proportional to frequency from below audible frequencies to 500 or 800 cycles (turnover frequency). The gain must be constant from the turnover frequency to 1590 or 2500 cycles (roll-off frequency).

Above the roll-off frequency, the gain must once again be inversely proportional to frequency. In order to carry the proper compensation to frequencies below audibility from a turnover point of 800 cycles, for example, a total increase in amplification of about 40 db is required. This amplification plus the over-all gain necessary means that a total amplification of 86 db will be required of the amplifier. The compensation above the roll-off frequency requires a reduction in gain, usually of 6 db per octave but sometimes of 2.5 or 3 db per octave. Thus, high-frequency compensation has no effect on the amplification requirements.

The fact that the preamplifier is the control center makes a small, attractively designed unit very desirable. The unit must have sufficient flexibility to cope with the several recording curves in use today. The ability to operate properly, even though a considerable distance from the remainder of the amplifying system, is a definite advantage.

Electrical Design

There are, of course, many combinations of elements which will produce the required gain. The compensating system can be designed in a number of ways. However, designing the low-frequency compensation into the feedback loop of an inverse-feedback amplifier has the advantage of improving the performance of the amplifier without any disadvantage, such as a requirement for a higher-gain circuit when compared to the "losser" type of equalization system. When this system of compensation is used it does, however, become necessary to be somewhat careful of the design as anyone

who has attempted a similar unit can no doubt attest. Without careful design instability will undoubtedly exist, particularly when as much as 40 db of feedback is applied around more than one stage of amplification.

In order to allow sufficient gain while keeping the number of stages to a minimum, type 6SJ7 tubes were chosen. This allows the necessary 86 db of amplification to be obtained in only two stages. The use of triodes would require at least three stages for the same amplification. The problem of applying the feedback around such a three-stage amplifier would be a much more difficult one than with the pentodes.

The 6SJ7 pentodes are connected in the usual manner except that the cathode bias resistances are quite low. This makes it possible to eliminate cathode bypass condensers without an excessive amount of current feedback and also minimizes the possibility of hum trouble due to the unbypassed cathodes. In addition, the bias is quite low and allows the maximum transconductance to be realized from the tubes.

One of the most common failings of amplifiers of this type is the method of applying the feedback loop. In the normal case the impedance of the feedback loop is a small fraction of the impedance of the plate load of a pentode amplifier. If the feedback loop is connected directly to the plate of a pentode, part of the gain reduction is due to loading of this plate circuit rather than negative feedback. This difficulty has been avoided by taking the feedback from the output of a cathode follower which is driven by the second 6SJ7. This cathode fol-

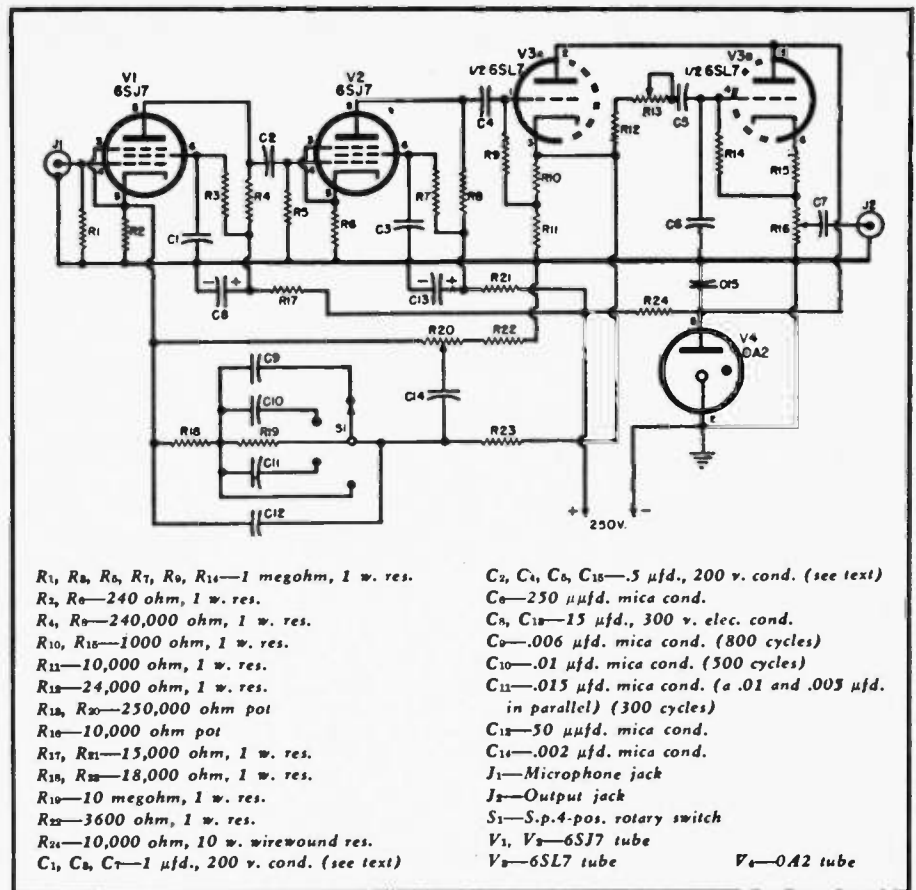
lower input is modified to produce high input impedance in order to minimize phase shift which would cause low-frequency instability.

The original amplifier built along these lines achieved high-frequency compensation by the simple and well-known method of loading the magnetic pickup with resistance. While this method is simple and works well with pickups which have a resistance which is low compared to the reactance of the unit at the roll-off frequency, it has the disadvantage of being suitable for only one type of pickup. Later models have incorporated a type of high-frequency compensation which does not depend on the type of pickup used. This is accomplished by R_{11} and C_4 as shown in the circuit diagram. R_{11} can be made continuously variable to give complete control of the roll-off frequency or can be a step adjustment. In the latter case R_{11} plus R_{12} should be equal to $1/2\pi f_r C_4$ where f_r is the roll-off frequency. Different R_{11} values can then be selected with a switch to give the desired roll-off frequencies.

A second cathode follower is used to provide a low-impedance output system which allows the preamplifier to be remotely located with respect to the power amplifier without any detrimental effects on the frequency response. The two cathode followers cause a reduction in voltage gain of about 3 db which is not serious since the over-all gain is still about 43 db.

The combination R_{20} , R_{21} , and C_{11} allows an adjustment of the high-frequency response. With R_{20} in midposition no modification of high-frequency response exists. At one extreme of R_{20} , half of the feedback loop is shunted by C_{11} . This results in a 6 db reduction of the frequencies above about 6000 cycles. The other extreme of R_{20} bypasses the center of the feedback loop to ground and results in an increase of amplification above about 4000 cycles. With R_{20} in this position, the gain increases to a value determined by the resistance of R_{21} . At intermediate settings of R_{20} , about the midpoint, modifications of the frequency response less than that produced at the extremes will be obtained. This control in conjunction with the roll-off control can be used to approximate 2.5 and 3 db rates of attenuation above the roll-off frequency as shown in the plot of frequency vs amplification.

In any amplifier which has high gain at low frequencies, the problem of making the internal impedance of the power supply low enough so that low-frequency oscillation or "motorboating" does not occur is a serious one. In this instance it was necessary to use a voltage-regulator tube to isolate the cathode-follower plate circuits from the rest of the amplifier. By the same token, care must be exercised when utilizing the main amplifier-power supply to supply power to the preamplifier. This can be done but considerable decoupling must be employed in order to avoid low-frequency oscillation. The simplest and best solu-

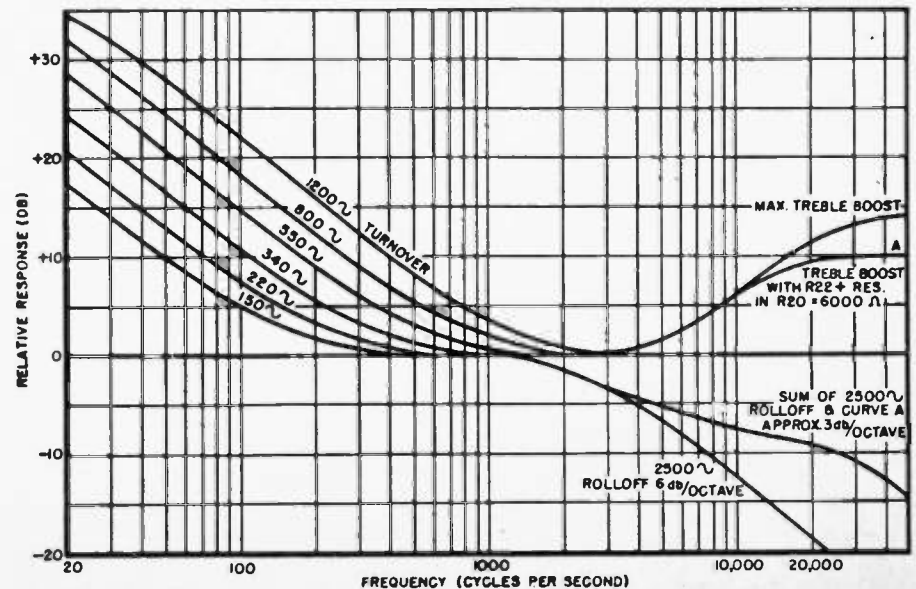


Complete schematic covering a preamplifier designed to be used with magnetic pickups.



The amplifier section was constructed on an "L"-shaped aluminum bracket with shield.

Preamplifier equalization curves. See text for discussion of measurement procedures.



tion is to use a separate power supply if possible.

Considerable attention was devoted to reducing hum in the amplifier. Hum was reduced to inaudibility by a combination of design practices and constructional features. Heater leads to the amplifier tubes are made of Number 22 switchboard wire. This wire has a very thin insulating covering and can be twisted very tightly. The wire also forms well and can be placed close to the metal base. Ceramic tube sockets were used. Mica-filled Bakelite should also be satisfactory, but trouble with leakage is sometimes experienced with ordinary Bakelite sockets. In addition to these constructional features, a potentiometer is placed across the heater circuit to allow a balance with respect to ground to be obtained. The rotor of this potentiometer is returned to about 20 volts positive rather than to ground potential.

The amplifier displays no hum and very little random-noise output with ordinary type 6SJ7 tubes. The 5693, which is a premium version of the 6SJ7, has also been used. This tube has considerably less microphonic output than the 6SJ7 even though microphonics have not been a problem with the 6SJ7. In addition, the 5693 ap-

pears to have a very low hum content. This was apparent since with 5693 tubes in place no hum was observed, irrespective of the position of the hum-balancing potentiometer. With the 6SJ7 tubes the balancing control had to be used to eliminate hum.

Operating Characteristics

The original unit has been in use for over a year. The results have been outstanding. Universal comment among those who have heard it in comparison with other amplifiers has been most favorable. Most noticeable is the exceptional "cleanness" and "clarity" of the highs and the "depth" of the low frequencies.

The unit has not been provided with a bass-boost and attenuation control. However, there is less need for one when proper equalizing curves are available. There is a way provided to produce a similar result by switching to a higher or lower turnover frequency. In the opinion of many who have listened to the amplifier, this produces a more pleasing effect. The original unit was built with six turnover frequencies available. The accompanying plot of frequency vs amplification shows these characteristics. If more turnovers than those shown in the circuit are desired they can be

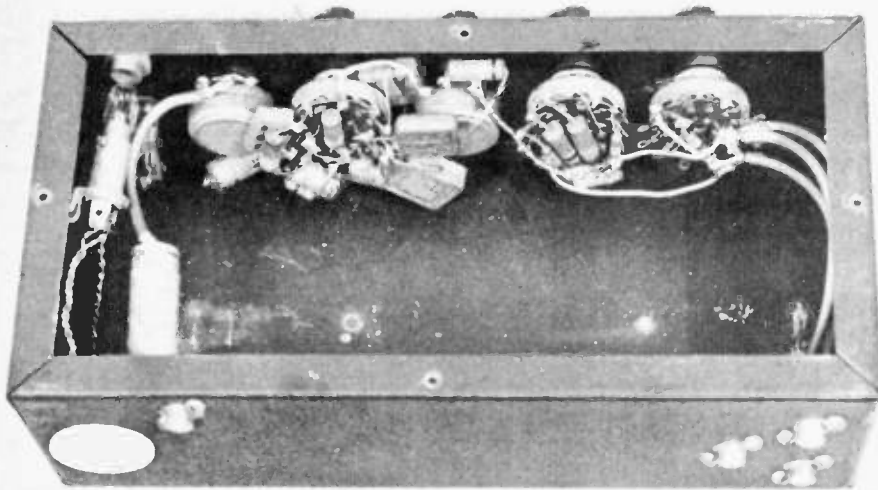
added very simply by the addition of switch positions on the turnover control. These switch positions should insert a capacitance in series with the feedback loop equal to $1/4\pi f_i R_{10}$ where f_i is the turnover frequency, in cycles-per-second, desired.

A series of performance tests has been run on the preamplifier. A *Hewlett-Packard* Model 202A low-frequency function generator was available for these tests. The sine-wave response of the amplifier with all frequency-selective circuits inoperative extended from one cycle-per-second to over 300 kilocycles. Square-wave response was checked within the limits of the 202A function generator. Square waves showed negligible tilt at four cycles-per-second. At 1200 cycles, which is the upper limit of the generator used, square waves showed no overshoot or ringing and no trace of rounding. Advancing the high-frequency boost control produced no ringing effects. Intermodulation distortion was checked at an output level of four volts. No deflection of the intermodulation meter could be observed on the three percent full-scale range. An inspection of the equalization curves shows that the objective of extending equalization below audible frequencies has been met.

The general mechanical features of the preamplifier can be seen in the accompanying photographs of the original unit. The equalization system and other controls are mounted in a 5" x 10" x 3" chassis which serves as a cabinet for the amplifier. The amplifier itself is constructed on an aluminum bracket. This bracket is assembled with a shield and placed in the chassis. Connections to the equalizing network and other controls are made with shielded leads through the aluminum shield. As with any amplifier of this type, care must be taken when wiring to insure proper operation. Metallized paper condensers were used throughout. The small size of the preamplifier is largely due to their use. Power is brought in through an octal tube socket in the rear of the unit.

Several units have been built using the same mechanical features and layout so that any duplication of this preamplifier would probably be most successful if constructed along the same lines.

Interior of cabinet with amplifier removed to show equalization network and controls.



A FOUR-CHANNEL AUDIO MIXER



The four-channel audio mixer. All of the controls are identified. The "not-used" toggle switch was incorporated to provide an extra switch should it be required. It could serve as a power supply "on-off" control, if desired.

By JOHN S. CARROLL

Construction details on a professional-type unit which uses only four audio transformers instead of the usual eleven.

EVERY recording studio except the very smallest has, on occasion, need for audio mixing facilities. Unfortunately, most of the inexpensive mixing equipment that is commercially available is, at best, of amateur quality while professional-type equipment is usually priced in direct proportion to its quality.

The problem faced by the author in building the unit to be described was one of small budget and the fact that the mixer was to be used with professional-type recording equipment. The mixer could not be permitted to add materially to either the noise level or distortion of the system. Thus, such shortcuts as had to be made to keep the cost to a minimum, had to be taken keeping this performance requirement firmly in mind.

The particular studio for which this unit was intended was that of a producer of documentary motion pictures and the facilities were designed for recording sound tracks by the "live-mix" method. That is, the narrator's (or at times, two narrators') voice would be recorded on, at most, two microphones while simultaneously the musical score would be dubbed from discs or tape. A total of four channels were therefore required, two for microphones and two for phonographs.

Under normal circumstances a professional-type, four-channel mixer would have a total of nine transformers up to the mixer stage, four input transformers to the preamplifiers, four output transformers from the preamplifiers to the line-mixer potentiometers, and a line-to-grid transformer to the mixer stage. Two additional transformers would be required, between the mixer stage and the output of the "line" amplifier and an output transformer to line, making a

total of eleven audio transformers required in the whole unit.

In the mixer unit to be described the total number of audio transformers was reduced to four by resorting to the following expedients. It was decided at the beginning that the phonograph channels would be fed by a pair of the new, high-fidelity crystal pickups, eliminating the need for preamplification for the two phono stages. Thus, only two input transformers were required.

With the preamplifiers and mixing amplifier on a single chassis, there appeared to be little objection to high-independence mixing, especially since the associated power supply would be remotely placed. Using molded carbon-element potentiometers (*Ohmite* or *Allen-Bradley* types) cut down on the slider noise and "frying" until it was virtually inaudible as compared to ordinary broadcast radio-type volume controls. In any case, the mixing is done at a point in the circuit where the signal levels are high and not too much amplification follows the mixer potentiometers.

Many references in current audio literature point out that a mixer circuit of this type tends to have some interaction between the channel controls. Our experience with this unit indicates that this particular problem has perhaps been unduly exaggerated. If any interaction exists, it is so slight as to be completely negligible. Some variation in the noise level was found at a point about two-thirds of clock-

wise rotation of the master gain control. This was eliminated by a slight circuit change which will be explained in detail later in this article. Interaction between the channels is apparently completely eliminated by the isolation resistors connected in series with each of the potentiometers.

The basic circuit is the one recommended by *United Transformer Company* for a portable remote broadcast amplifier. The two phonograph channels were added by simply paralleling them with the original two mixer potentiometers. Construction is straightforward but due to the very high gain of the unit certain precautions must be taken. For the lowest possible hum level, the ground points are placed as near the two inputs as possible. These, then, are the only two places where the ground connects to the chassis. A ground bus is used between these two points and the "B—" terminal and is completely insulated from the chassis at all other points. All of the other grounds, including the condenser cans, are insulated from the chassis and connected to the ground bus.

The heater circuit is not grounded at any point. The center tap of the 6.3 volt winding is returned to a tap on the voltage divider across the power supply. This places a fairly high positive bias on all heaters and prevents hum due to heater-cathode leakage or heater emission. With these precautions it proved unnecessary to use d.c. on the heaters of the preamplifier tubes. If any residual hum is present

despite this precaution, it can be minimized by increasing the value of the cathode bypass condensers, C_1 and C_{14} , to 100 $\mu\text{fd.}$ or higher.

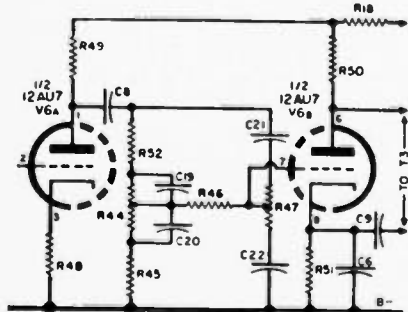
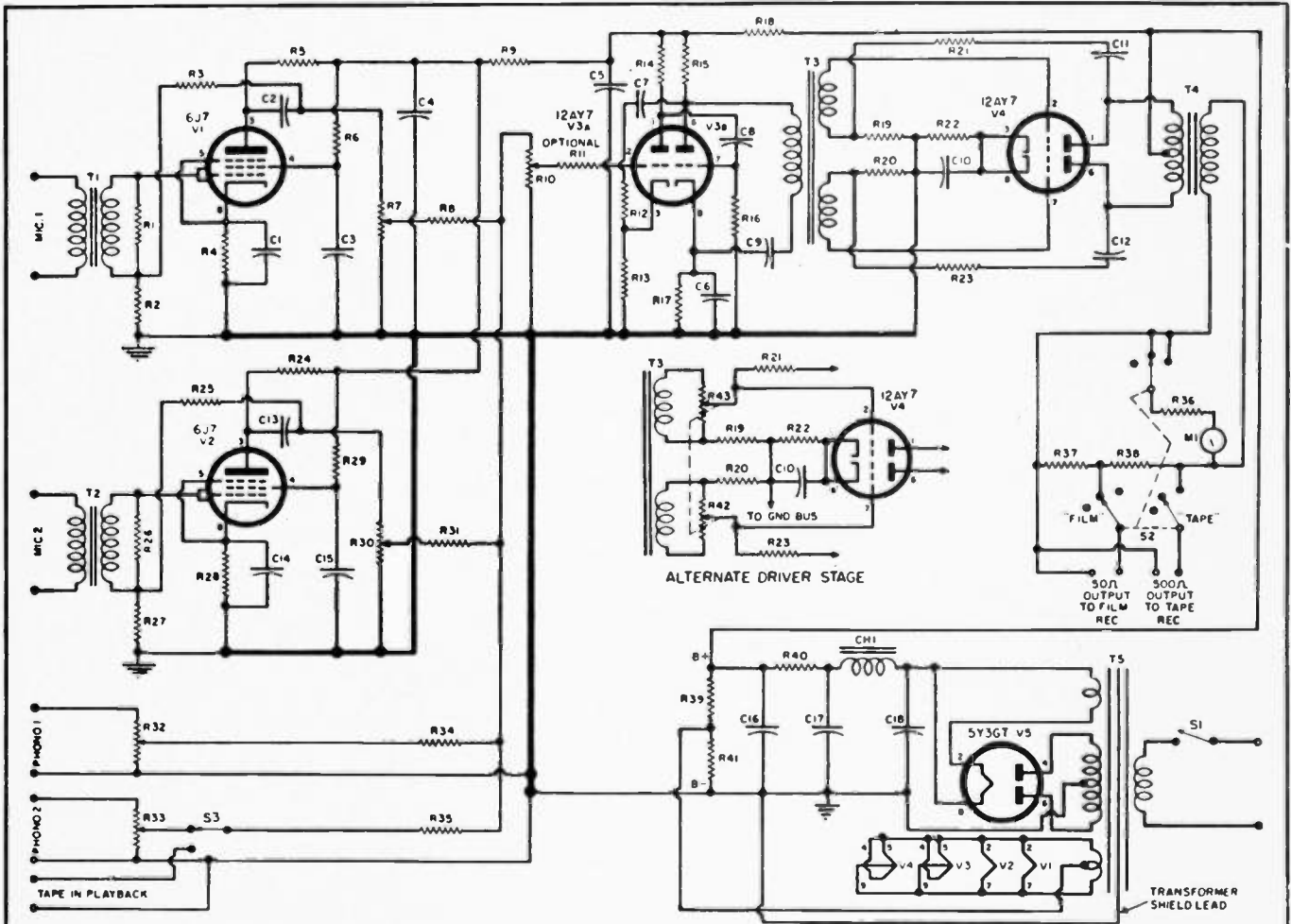
In order to insure minimum distortion, feedback is used around the preamplifiers, around the mixer amplifier stage, and around the output stage. There is little or no danger of in-

stability due to the loops around the preamplifiers and, in this case, the output stage also exhibits good stability. However, some resonance between the interstage transformer windings and capacitances in the mixer stage caused an ultrasonic oscillation which, while not audible, threw the vu meter completely off scale. This oscil-

lation was suppressed by placing small phase-shifting condensers (50 to 100 $\mu\text{fd.}$) across either R_{12} or R_{13} , or both.

If the builder wishes to add an equalizer to the unit, this may be placed between the two mixer amplifier stages and the feedback loop eliminated to provide sufficient gain for the equalizer. A 12AU7 was used in

Complete wiring diagram for the four-channel audio mixer. If only two inputs, Mic. 1 and Mic 2, are required, the schematic with its original driver stage will work without any difficulty. Should Phono 1 and Phono 2 inputs be required in addition, interaction may occur between the various controls. To eliminate this, it is suggested that the alternate driver stage shown below be used in place of the original circuit. When this is done, the master gain control R_{10} is replaced by a standard 3-megohm, $\frac{1}{2}$ -watt resistor. The dual potentiometer, R_{11} - R_{12} , then becomes the new master gain control. Resistor R_{11} should be connected to the top of the newly-connected 3-megohm resistor to obtain maximum gain. Also shown below is an alternate equalizer-mixer stage. This has not been incorporated in the audio mixer unit. It should, however, work out advantageously and may be worth the installation effort involved.



- ALTERNATE EQUALIZER-MIXER STAGE**
- R_{11} , R_{12} —68,000 ohm, $\frac{1}{2}$ w. res.
 - R_{24} , R_{27} , R_{41} —10,000 ohm, $\frac{1}{2}$ w. res.
 - R_{30} , R_{31} , R_{32} , R_{33} —1.5 megohm, $\frac{1}{2}$ w. res.
 - R_{44} , R_{45} —1800 ohm, $\frac{1}{2}$ w. res.
 - R_{70} , R_{71} —270,000 ohm, $\frac{1}{2}$ w. res.
 - R_{72} , R_{73} , R_{74} , R_{75} —500,000 ohm audio taper pot
 - R_{80} , R_{81} , R_{82} , R_{83} —470,000 ohm, $\frac{1}{2}$ w. res.

- R_7 —20,000 ohm, $\frac{1}{2}$ w. res.
- R_{10} —3 megohm pot
- R_{11} —100,000 ohm, $\frac{1}{2}$ w. res. (if required)
- R_{12} —120,000 ohm, $\frac{1}{2}$ w. res.
- R_{13} , R_{17} —820 ohm, 1 w. res.
- R_{14} , R_{15} —47,000 ohm, 1 w. res.
- R_{16} —1 megohm, $\frac{1}{2}$ w. res.
- R_{18} —5600 ohm, 1 w. res.
- R_{19} , R_{20} —18,000 ohm, $\frac{1}{2}$ w. res.
- R_{21} , R_{22} —82,000 ohm, $\frac{1}{2}$ w. res.
- R_{23} —290 ohm, 1 w. res.
- R_{24} —3600 ohm, $\frac{1}{2}$ w. res.
- R_{25} —56 ohm, $\frac{1}{2}$ w. res.
- R_{26} —470 ohm, $\frac{1}{2}$ w. res.
- R_{27} —24,000 ohm, 2 w. res.
- R_{40} —5000 ohm, 5 w. res.
- R_{41} —6000 ohm, 2 w. res.
- R_{42} , R_{43} —500,000 ohm dual pot
- R_{44} , R_{45} —1 megohm linear taper pot
- R_{46} , R_{47} , R_{48} —100,000 ohm, $\frac{1}{2}$ w. res.
- R_{49} —3300 ohm, $\frac{1}{2}$ w. res.
- R_{50} —22,000 ohm, $\frac{1}{2}$ w. res.
- R_{51} —2200 ohm, $\frac{1}{2}$ w. res.
- C_1 , C_2 , C_{10} , C_{11} —50 $\mu\text{fd.}$, 25 v. elec. cond.
- C_3 , C_{18} —1 $\mu\text{fd.}$, 400 v. cond.

- C_3 , C_{10} —1 $\mu\text{fd.}$, 200 v. metallized cond.
- C_4 , C_7 —40/40 $\mu\text{fd.}$, 350 v. elec. cond.
- C_5 , C_6 —5 $\mu\text{fd.}$, 400 v. metallized cond.
- C_8 , C_{11} , C_{12} —.05 $\mu\text{fd.}$, 400 v. cond.
- C_{10} , C_{17} , C_{18} —20/20/20 $\mu\text{fd.}$, 450 v. elec. cond.
- C_{19} , C_{22} —.002 $\mu\text{fd.}$ mica cond.
- C_{20} —.02 $\mu\text{fd.}$, 400 v. cond.
- C_{21} —.0002 $\mu\text{fd.}$ mica cond.
- T_1 , T_2 —Input trans. 50 ohms to single grid (UTC A-10 or A-11)
- T_3 —Interstage trans. Single plate to push-pull grids, split secondary (UTC HA-106)
- T_4 —Output trans. push-pull plates to 500 ohm line (UTC HA-114)
- T_5 —Power trans. 240-0-240 v. @ 40 ma., 6.3 v.c.t. @ 2 amps; 5 v. @ 2 amps. (Thoradson TS-24R00 or equiv.)
- CH_1 —7 hy., 50 ma. choke
- M_1 —Vu meter (see text)
- S_1 —S.p.s.t. toggle switch
- S_2 —3 p. 3t. rotary switch
- S_3 —S.p.d.t. toggle switch
- V_1 , V_2 —6J7 tube (1620 preferred)
- V_3 , V_4 —12AY7 tube
- V_5 —5Y3GT tube V_6 —12AU7 tube

preference to the 12AY7 for the equalizer. The circuit of this altered section is included in the diagram.

As previously mentioned, while there is no interaction between the four channels, there was some between the channel potentiometers and the master gain control. This was eliminated by feeding the combined output of the four mixer potentiometers directly to the grid of the mixer-equalizer stage and by placing the master gain control in the grid circuit of the output stage. A double potentiometer is used to control the push-pull grids simultaneously, as shown in the alternative schematic.

The output of the unit was designed to feed either a tape recording setup with a 500-600-ohm line bridging input or a 16 mm film recorder with a 50-ohm line input. To avoid complicated switching of the output transformer secondary taps, a simple matching pad was placed on a switch on the output, so that switching from "tape" to "film" automatically adjusts the output impedance. It is evident that this pad causes a loss of output level of about 16 db; however, the film recorder in question had ample gain and the loss was not important. Using the pad permitted the vu meter to work across a 500-ohm line at all times. In addition, the pad has the effect of terminating the line, even when the output is disconnected. Thus, rehearsals can be run while feeding only a bridging monitor amplifier having a 20,000-ohm input.

The vu meter can be switched off during preliminary set-ups so that accidental jolts while moving the microphone will not damage the needle. The 3600-ohm resistor in series with the meter is the standard calibrating resistor usually used with vu meters. If variation in output level is required, or changes in meter range are desirable, this resistor can be replaced by a 7500/3900-ohm variable pad to extend the meter scale. Such pads are available from the manufacturer of such meters.

In this article great emphasis has been placed on instability and the causes and cures for it. This may come as a surprise to those audiophiles who are not accustomed to the extremely high-gain circuits used in recording work. The tube complement is based on the use of low-noise types throughout. If 6J7's are used instead of 1620's, it may be necessary to select the quietest from a number of tubes. The same thing applies to the 12AU7 if substituted for the 12AY7 in the mixer stage. The output stage should, in any case, be a 12AY7. The 12AX7 is not a good substitute due to its higher mu and plate resistance. The over-all gain of the unit is such that the thermal agitation noise of the first stage sets the limit on following amplification. If all recommendations are followed, this hiss should be stronger than any other noise, residual hum, etc.

The power supply, which is not shown in the photographs, should be built on a separate chassis and the



Rear view of unit. Note particularly the type of jacks and plugs that were used in the construction of this unit. It is important that all of these components be of the shielded type to prevent hum pickup. The jacks and plugs are, from left to right: Mic. 1, Mic. 2, power cable connector, (far right, top) 50 and 500 ohm outputs, (far right, bottom) Phono 1, Phono 2, and tape playback inputs.

same rule that applies to the mixer followed here. Ground the chassis at only one point and do not ground the center tap of the heater circuit. The power supply should be kept as far from the mixer unit as possible. If this is not feasible, it may be necessary to use triple-shielded input transformers.

If the power supply can be isolated the power transformer lead shown as shielded and grounded can be omitted. Many transformers are made today without this shield. Should your unit not come so equipped, you can ignore this requirement—if the power supply is isolated.

Just a few final suggestions regarding the construction before you whip out your soldering iron and start to build this unit. In the interests of economy and to avoid purchasing unneeded components, decide at the start whether for your purpose the original circuit, shown in the schematic, is to be used or whether one or both of the alternate circuits are to be incorporated. Some of the parts specified in the original diagram will be omitted if one of the alternate circuits is used and *vice versa*.

Another point which cannot be emphasized too strongly is the matter of

grounding discussed earlier. If you are seeking trouble-free performance from this unit, it is imperative that the author's suggestions be followed to the letter.

It goes without saying that in this application, at least, quality transformers will have to be used. The builder will find that compromises are too expensive in the long run to be afforded. Just keep in mind that this circuit incorporates only four audio transformers which are doing the duty of the eleven units normally encountered in commercial versions of four-channel audio mixers.

This is one instance when "bargain" transformers will prove to be no bargain and the prospective builder will do well to stick to quality components. The transformers specified in the parts list are recommended by the author as they have proved entirely satisfactory in this application. Use units of similar quality for best results.

While all of these precautions may sound like undue "fussing" to the novice builder of audio gear, they are being handed along as "gospel" by one who has been through the mill. Good luck in building this unit. It is well worth the effort.

-30-

Under chassis view of mixer. Wiring is critical so check author's suggestions regarding grounding and the location of such ground points before construction.

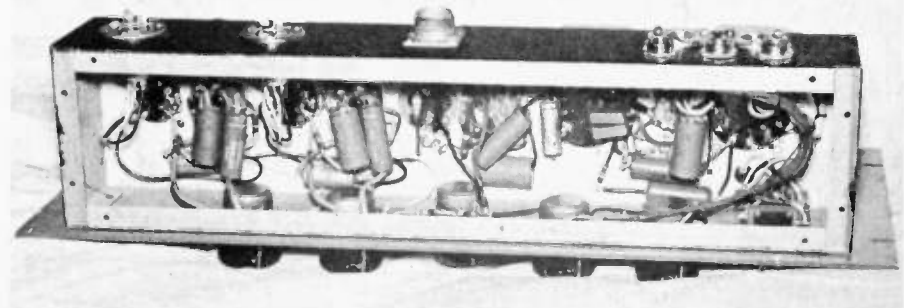


Fig. 1. The Printed Electronic Circuit preamp which measures 5½"x3"x2½". With equalizers using standard components, the over-all size of the unit will have to be enlarged.



MINIATURE PREAMP

By
ROBERT WIENER
Product Engineer
Centralab, Div. of Globe-Union, Inc.

This compact preamplifier was designed to demonstrate how miniaturization can be accomplished through use of printed circuits. Although it cannot be duplicated exactly in the form shown, it can be reproduced to a reasonable degree by substituting standard radio parts for the printed circuits which are not currently available at parts distributors.

THE miniature preamp to be described was designed, primarily, to demonstrate the practicality of printed circuits in audio gear rather than to present any radically new or startling circuitry.

The fact that the final version of this experimental circuit required only two dual-triodes and a minimum of controls was a pleasant "bonus."

The most important feature of the unit is its small size which has been achieved by using Centralab PEC's (Printed Electronic Circuits) wherever possible. With the exception of the five equalizer circuits, all of the listed components are readily available. By making the chassis slightly larger, standard components can be sub-

stituted for the printed equalizer networks.

Simplicity of operation is obtained by using only three controls: a selector-equalizer switch and a dual control consisting of a "Compentrol" (Centralab's compensated loudness control) and a gain control. The selector-equalizer switch has six settings, the first five being phono equalizer positions and the sixth bypassing the equalizers for other sources. More inputs may be added but, for the purposes of this article, only one additional position is included. The selection of equalizer positions, as shown in Fig. 4, will provide most settings normally required.

Centralab has recently introduced the "Senior Compentrol." This is a dual concentric unit with the compensated loudness control mounted on the rear and operated by the inner shaft (front knob). A level-set unit, the front control, is connected to the "Compentrol" and is operated by the outer shaft (rear knob).

The debate over compensated loudness controls vs tone controls has been, is, and always will be waged. However, based on the author's experience in building various control systems, he has found that usually bass and treble

controls are set for flat response or some degree of boost—seldom for cut. The "Compentrol" offers a combined bass and treble boost in itself and, with the judicious use of the level set, a flexible tonal range is obtained. Despite the controversy, the reader is urged to try the unit himself and make up his own mind.

The schematic of the preamp is given in Fig. 4. There are, as mentioned previously, two dual-triodes—a 12AX7 and a 12AU7. The 12AX7 serves as the equalizer section and the 12AU7 is the preamp.

The two switch decks of the single ganged switch (S_1) in the preamp section serve separate functions. The section which switches the inputs is designated as S_{1A} . The first five positions are shorted together since these positions are the five phonograph equalizer positions controlled by the other deck. The sixth position connects directly to the input J_2 and R_1 . Switch S_{1B} is used to select any of the five available equalizer circuits and in its sixth position disables this portion of the circuit.

The first input, J_1 , is for phono. The second input, J_2 , is for radio, TV, tape, or any input having the higher voltage output commonly associated with these program sources. This jack is connected to the level set, R_1 . This level set, as well as R_2 for the phono input, is a Centralab Model 1 control used in hearing aids and miniature radio receivers. The one used in the construction was specially modified with mounting studs so that the control could be attached directly to the chassis, with the screwdriver shaft available as a semi-permanent adjustment. A standard midget hearing aid control (the Centralab B16-118) can be used equally as well.

The switch rotor of S_{1A} is connected

Fig. 2. Equivalent circuits for the printed circuit equalizers shown dotted in Fig. 4.

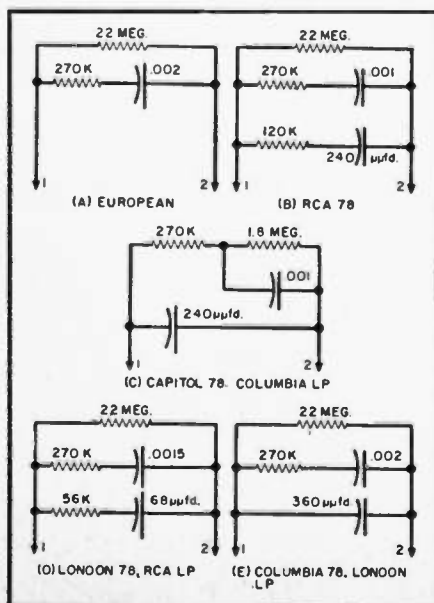
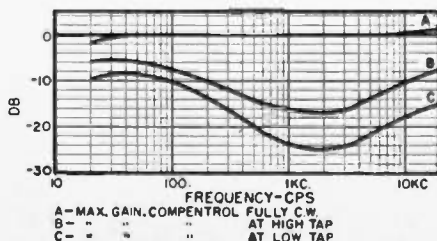


Fig. 3. Gain vs frequency of preamp equalizer. Input is .245 v. at J_2 , 0 db level 24.5 v. at 1 kc. "B+" is 240 volts d.c.



to the low-frequency filter composed of C_6 , C_7 , C_8 , R_{12} , R_{13} , and R_{14} . Following this is the first section of the 12AU7 feeding the volume control R_{16} . R_{16} was inserted in the stage preceding the "Compentrol" to do away with a grid resistor for V_{2B} , as well as to decrease some noise commonly encountered in high-gain amplifiers. Ordinarily the "Senior Compentrol" is to be used with the level-set control feeding the "Compentrol" directly. This is the manner in which they are supplied, but the user may make this change if he feels that it is desirable.

The center terminal of the level-set control is tied to the grid of the output stage, V_{2B} , and its plate a.c.-coupled to the right terminal of the "Compentrol." There are several points to be noted about placing the "Compentrol" in the output of the preamp. With this connection, the grid resistor of the following stage, usually the first audio stage of the power amplifier, should be removed. That is, the "Compentrol" should feed directly into the following grid. In addition, the device is one which exhibits decreasing impedance with increasing frequency and thus, aside from exceedingly low frequencies which are filtered out in the low-frequency filter, offers a low-impedance output.

The impedance level may not be comparable to that of a cathode follower when connected with long cables to the power amplifier but with this control in the output, the gain that would have been lost in the cathode follower is retained and some measure of low-impedance output is obtained. A phono jack, J_3 , tied to the center terminal of the "Compentrol," is the output take-off to the power amplifier.

The response of the preamp portion of the circuit is shown in the graph of Fig. 3. The response was obtained with the input at J_2 and the gain maximum, i.e., level set, volume, and "Compentrol" full on. Curve A of Fig. 3 shows the response of these two stages. At 20 cps, the output is down about 2 db while at 20,000 cps it is up about 1½ db. The response is quite good but may be modified if desired. The droop at the low-frequency end of the spectrum is due to the low-frequency filter taking over. Unfortunately, none of the audio generators available to the author furnished a lower frequency than 18 cps so the effect of the filter cannot be demonstrated here.

The high-frequency rise could be decreased by changing the value of C_9 , the cathode bypass condenser for V_{2A} . This is a .01 μ fd. ceramic unit and can be increased in value by some 50 percent to drop the response. Other means can be employed but since they are familiar to the hi-fi enthusiast, they will not be discussed here.

Curves B and C (Fig. 3) show the output curve when the "Compentrol" is set at the high (62½ per-cent rotation) and the low tap (37½ per-cent rotation) respectively.

Referring to Fig. 3, the gain figure

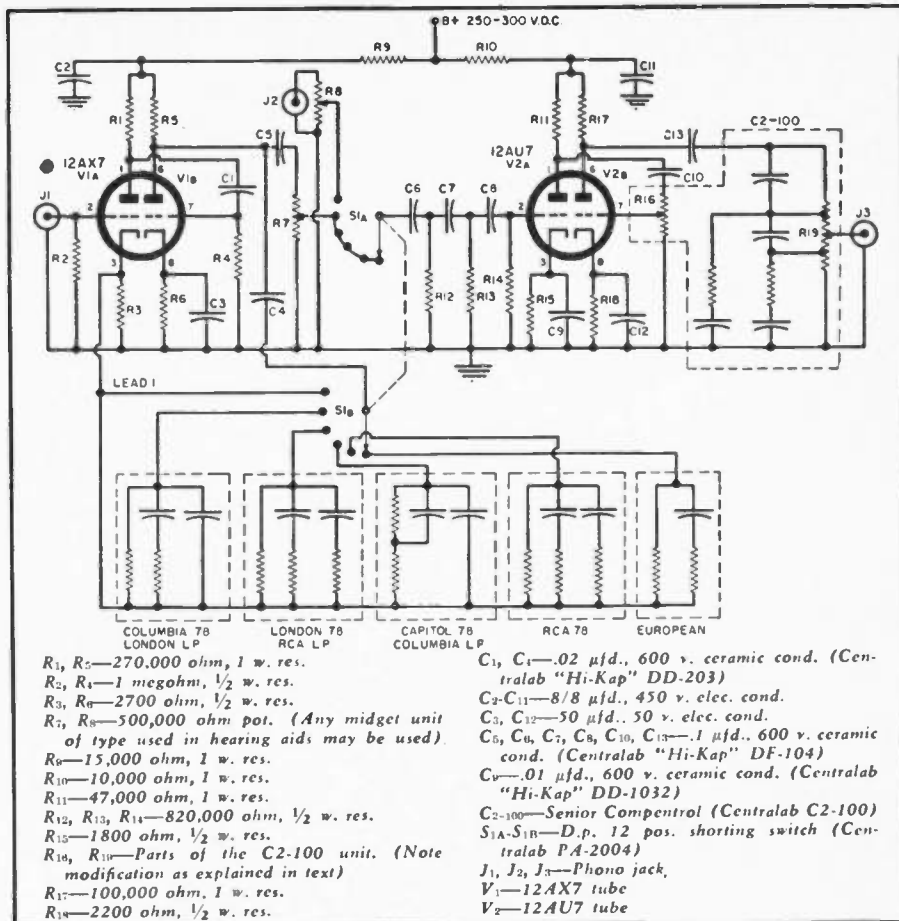
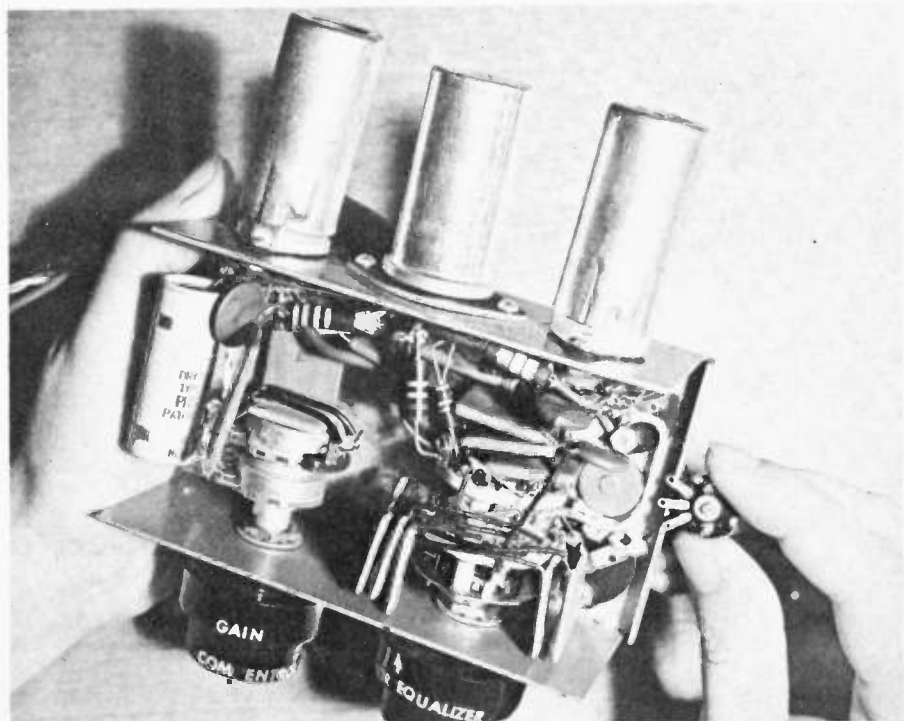


Fig. 4. Complete schematic of the printed circuit preamp. The equalizer networks, shown dotted at lower left, are not commercially available but suitable networks can be substituted by following the diagrams and parts values shown in Fig. 2.

is 100 or 40 db since the input voltage at J_2 was 245 volt (1 kc.) and the output voltage at J_3 was 24.5 volts.

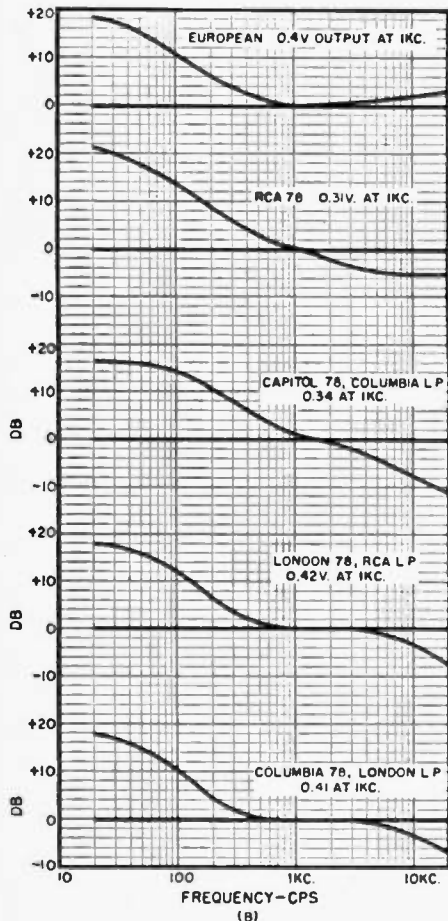
The equalizer circuit is basically the one used in the H. H. Scott 120-A equalizer. The deviation is the output

Fig. 5. Interior view of the preamp showing compact arrangement of components, use of the PEC plates, and the special Model 1 subminiature control (held in hand). The equalizer switch's arrow is pointed to the "European" equalizer.



of the second stage V_{1B} . In the original the outputs are separated into positions for the standard (78 rpm) records and for the LP's (both 33 and 45 rpm). The output for the 78's is divided down for the lower output voltage while that for LP is not. Component-wise the circuit follows the Scott circuit but the feedback net-

Fig. 6. Gain vs frequency of the equalizer with input at J_1 equal to 1 mv. See text.



works are made in the style of the Centralab "Couplate." These five "Couplates" are special units and are not currently available. They are shown in connection with this piece of equipment merely to demonstrate the adaptability of printed circuits to the audio field. There are five of these circuits. The components which these replace are shown within the dotted lines at the lower left-hand corner of Fig. 4. Fig. 2 is the schematic for these five plates with the component values indicated. From this diagram the reader can see just how many connections and components are saved by using printed circuitry.

Response curves for the complete unit are shown in Fig. 6. The input was fed into J_1 at 1 mv. and the output was read at J_2 . Gain was maximum throughout. These curves include the total response for the unit and should not be compared precisely to the curves published by manufacturers. The output voltages at 1 kc. are listed on the curves and thus are the levels taken for 0 db. Generally they run at .40 volt out and thus show that the over-all gain at 1 kc. is .40/.001 or 400. This is 52 db. At the low-frequency point (20 cps) the boost amounts to 20 db in most curves and the total is then about 70 db. These figures are representative of the requirements for most power amplifiers. Since the output of most magnetic cartridges is actually on the order of 10 mv. or more at the maximum gain setting, the curves would result in about 4 volts output—far more than required for living room listening unless the neighbor's living room is included in your audio system.

The photograph, refer to Fig. 5, illustrates wires that are used to connect the unit to the required external power source; "B+", heater, and ground. The jack, J_1 , is shown with J_2

directly below it. The screwdriver adjustments for the level sets are to the right and above the respective inputs. The output jack, J_2 , can be seen at the extreme lower right of the photo. The dual knobs, labeled "gain" and "Compentrol" control those functions while the dummy dual knob labeled "selector-equalizer" controls that switching function. The arrow in Fig. 5 points to the "European" record setting and is switched clockwise through all the remaining functions of the selector.

The five equalizer plates are visible in Fig. 5 at the sides of the equalizer switch.

The plates at the rear of the switch are the .1 μ fd. ceramic condensers, C_5 , C_6 , C_7 , C_8 , C_{10} , and C_{12} . Note that the switch is actually a 2 to 12 position one and since, in this case, only six positions were used the remaining terminal points were put into service as tie points for "B+", filter resistors, and other components that would have otherwise required extra terminal points.

The reader will note that the chassis is not crowded so there is no need to "cheat" on the assembly. Those who want to duplicate this construction using components for the not-available equalizer plates will have to use a slightly larger chassis since obviously something will have to give.

It is hoped that the prospective builder will be pleased by the quality of reproduction as well as by the ease of operation. It is left for him to incorporate whatever additional inputs, a.c. switching, etc. he desires. It is left for the manufacturer of equipment to determine whether his customers want gadgets and controls or simplicity along with their listening pleasure.

REFERENCE

1. Wiener, R. & Nelson, N.: "The Compentrol—A Compact Loudness Attenuator," RADIO & TELEVISION NEWS, November 1953.

A PRACTICAL TRANSISTOR PREAMP

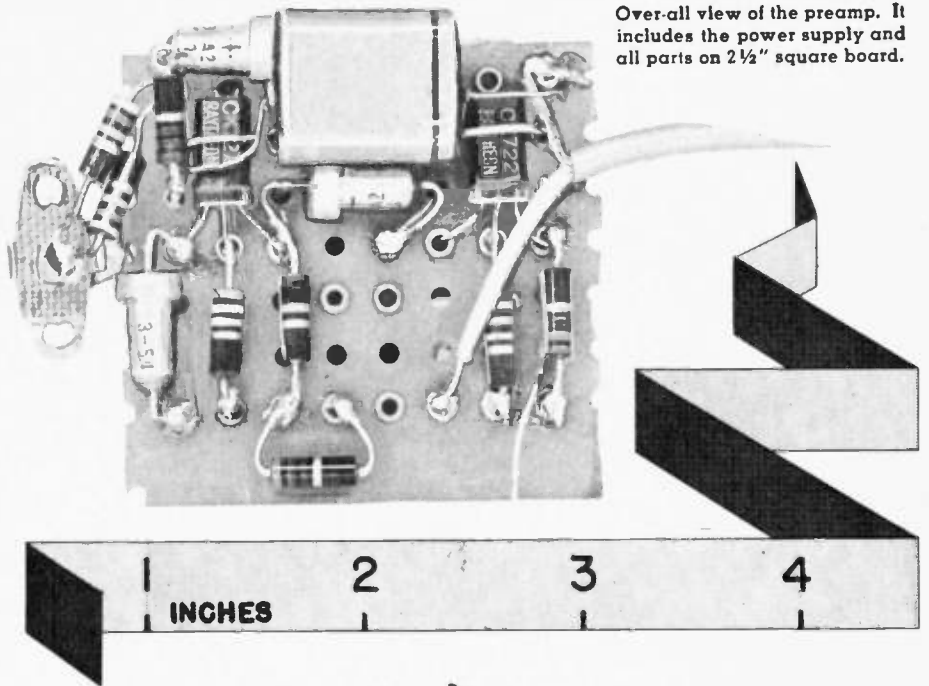
RECENTLY many articles have appeared describing practical applications for transistors. Still, work with transistor circuitry has been avoided by the audio experimenter and hobbyist. Skepticism came with the advent of the transistor (i.e., noise, cost, "temporary" battery supplies) and has remained to discourage much-needed exploitation of the new field.

In a short time, transistors have come a long way. To you, the hobbyist, a simple preamplifier can be constructed in a matter of hours, for a cost equal to that of the commercially available single-tube magnetic pickup preamplifiers. The complete unit may be mounted on a board $2\frac{1}{2}$ inches square, power supply and all. Internal noise is essentially inaudible at more than comfortable music-listening levels. A practical 6-volt battery supply (consisting of *Mallory* mercury cells) will provide better than 2000 hours of service. This means that with an average of four hours usage per day, you can forget about the supply for a year and a half! All parts are easily available on the open market, or in the junk-box.

Several working models of the preamp have been built, with no custom tailoring necessary. A straightforward circuit is used, with grounded emitters and no tricky feedback (see Fig. 1). The *Raytheon* CK727 low-noise transistor is used in stage 1, followed by the popular CK722 in a similar circuit arrangement. In resistance coupling of transistors, there must be a compromise made on the values of load resistance. The impedance match must be as close as possible (about 20,000 ohms optimum), yet this impedance must be low enough to allow proper collector current to flow without demanding too high a supply voltage "pump."

Although transformers solve the problem, they are bulky, expensive, and limited in their frequency response. But with grounded emitters, there is still gain to burn. The db wasted due to compromise will never be missed. This preamp still ends up with a gain of 2000 on voltage. The base resistors were set for low noise, low distortion, and best gain for the stage. The values are not highly critical, and 10% parts are used throughout. The coupling capacitors are large in value, since they work into the low impedance base circuits. Five or eight μ fd. electrolytics at very low voltage ratings are small and fairly inexpensive. The author's model uses tantalums, but this is not necessary. Notice that the output impedance is very low, and long leads from the preamp cause no losses. The output capacitor may be smaller, since it is assumed that the unit will work into a high impedance input of the main amplifier.

The simplest solution to the equalization



By
BERNARD EDELMAN
Dynamics Test Engineer, Convair

The development of the CK727 low-noise transistor makes the construction of this compact unit possible.

ization problem was to equalize right at the pickup. A *G-E* was used for tests, aiming at the NARTB curve. The low impedance input of the CK727 was a help rather than a hindrance, since the equalization components are also of low impedance, and there was no stray pickup or hum problems. The series resistor to the pickup both completed the equalizer design, and presented the proper load to the pickup. The equalization circuit may be glorified into a selector switch choice of a number of the curves, with no evils of high impedance. The attenuation and insertion loss of this equalizer are of no consequence to the preamp following, with its gain of 2000. The resultant output from LP's is close to one volt delivered across the 6800 ohm load resistor.

Total battery drain from 6 volts is .9 milliampere. The batteries (4 series

cells) were mounted right on the reverse side of the small fiber board used as a chassis.

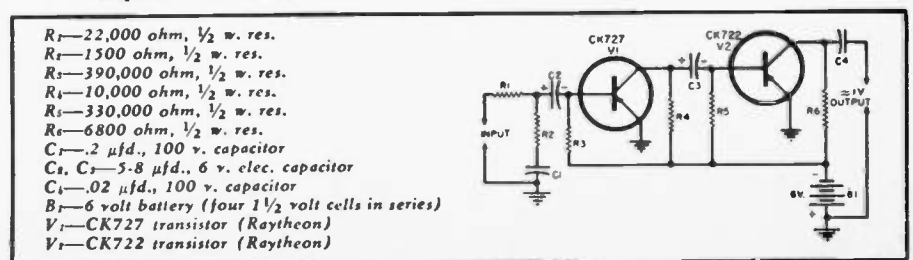
The over-all result will be an equalized one volt of signal from your turntable, humless and virtually distortionless. The changer (or turntable) may be taken for granted as it always has been, but now you may put your amplifier in the basement or attic if you so choose.

Just keep the soldering iron off those little wire pigtails. Don't count on a pair of pliers pinching the lead to draw off the heat. Heed the voice of sad experience, and use sockets of the store-bought or home-made variety.

Remember, we can help those little wonders to come down in price and up in variety, if we put them to just a few of the untold thousands of uses of which they are capable.

—30—

Complete schematic diagram and parts list for the compact transistor preamp.



Author's control amplifier. Controls from left to right: input selector switch, master gain control, expression potentiometer, treble, and bass.



HI-FI CONTROL AMPLIFIER WITH "EXPRESSION"

By **MAURICE P. JOHNSON**
Engr. Dept., Station WAAM

Details on a five-tube unit which features an "expressor" circuit for volume expansion and compression to enhance the dynamic range of reproduction on recorded material.

THE ultimate objective of most audio equipment design is to produce a greater degree of realism in the reproduction of sound. A perusal of past issues of this magazine reveals that considerable material has been published concerning various circuitry germane to achieving this goal. A comparison between present designs and those prevalent only a few years ago will show that much progress has been made. Harmonic and intermodulation distortion in audio amplifiers have been reduced to microscopic amounts. Speaker damping and controlled feedback loops have improved transient response and provided extended linear frequency characteristics. Loudspeakers, enclosures, pickup devices, and preamplifiers have likewise been immeasurably improved. An appreciation of the benefits of "high fidelity" in the home, ambiguous as the term may be, is indicative of the increased interest in high quality sound reproduction, even by the layman.

A major program source for home audio reproduction is derived from recorded material, supplied directly by playing phonograph records, or indirectly via radio broadcasting. The reproduction may well be free from distortion and be "clean," hum and noise may be at a minimum, and still not compare favorably with an original live performance.

The restricted dynamic range of most recorded material is a definite

factor contributing to the destruction of realism of such reproduction. The feeling of "presence" can often be enhanced by artificially increasing the dynamic range of recorded reproduction. An electronic circuit for such action is referred to as a "volume expander." The relative merit of volume expansion in audio reproducing equipment has been the subject of much discussion by engineers and others, but a few basic facts should be mentioned.

Even present-day disc recording techniques are such that the volume range on the record must be restricted in order to prevent excessive cutter swing on peaks which would otherwise overcut the groove. Likewise, the minimum audio level is limited by the signal-to-noise ratio of the finished disc, although this has been considerably improved by modern plastic pressing materials.

Radio stations indulge in "gain riding" even during recorded programs, in order to keep their average modulation high at the transmitter. To this end also, limiting amplifiers are used on the transmitter feed, with the result that these techniques further alter the original dynamic range of the program.

Of course, no electronic circuit to date will restore the exact dynamic range of the original live performance to material that has been subjected to such volume restrictions. Nevertheless, most listeners and audio enthu-

siasts will agree that a degree of volume expansion will do much toward increasing the brilliance and life-like character of the reproduction.

This article will be concerned with a control amplifier which is intended for use with a home audio system, incorporating a versatile expander circuit together with several other desirable features.

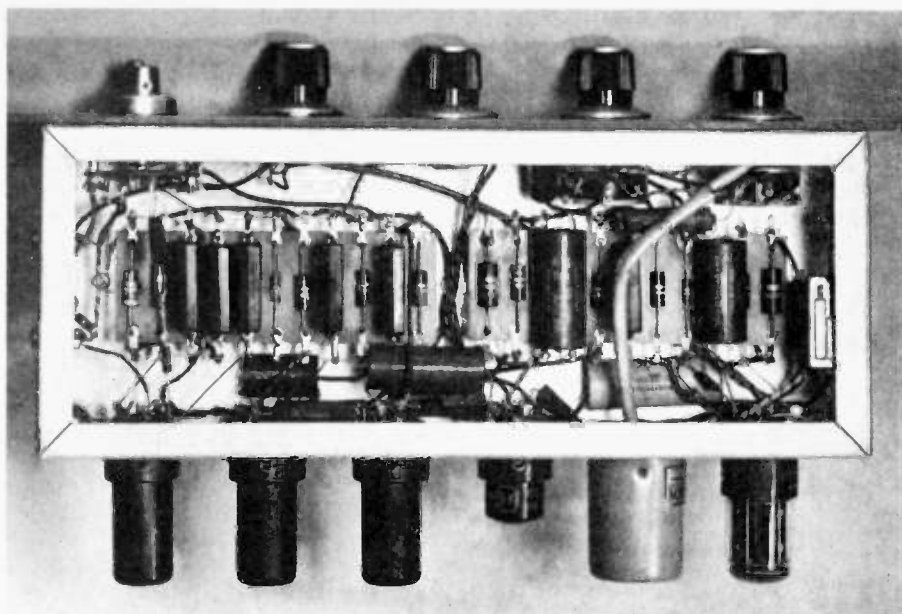
The control amplifier was designed with certain specific requirements in mind. The power amplifier to be used in the system, of the Williamson type, requires an audio input of approximately one volt r.m.s. This level was needed from a low impedance output, to allow operation of the control at a point remote from the power amplifier. Such output was obtained by use of a cathode-follower stage, which allows the control and power amplifiers to be separated by almost any reasonable distance without hum pickup or high frequency roll-off due to cable capacity shunting effects. It should be noted that, although the cathode-follower output impedance is low, the input to the power amplifier should be of high impedance, in the neighborhood of 100,000 ohms or more, for proper operation.

The control amplifier functions as a centralized focal point for all signals used with the complete audio system. For such use, it was necessary to include three inputs for relatively high-level signal sources. These three inputs accommodate feeds from the tape recorder playback and an AM-FM tuner, as well as a TV tuner chassis. The inputs require approximately 0.2 to 0.5 volt r.m.s. signals for the proper expander operation, as will be discussed.

A fourth input is included with a simple, yet effective preamp for use with magnetic phonograph pickups capable of outputs of 0.01 volt r.m.s., such as the *Pickering, Audak*, and others. A single pentode tube is used in the preamplifier stage, connected in

the conventional manner for a high-gain stage. The equalization to compensate for modern recording characteristics is accomplished by shunting the preamplifier plate load with an RC circuit formed by R_8 and C_8 in series. This method is explained in detail by Herb Matthews, in "Design Considerations for High-Quality Reproducing Systems," Part 2, RADIO & TELEVISION NEWS, May, 1950. A single turnover frequency of 800 cycles, in conjunction with the tone controls to be covered, was found adequate for the majority of recordings. Some persons may desire other turnovers, which can be adjusted by the choice of the value of C_8 . No high frequency roll-off has been included in the equalization, since this can be approximated by proper settings of the treble tone control.

Shunt-type equalizers are encountered in broadcast work and many designers have devised circuitry for exact correction for nearly all recording curves. (See "An Improved Equalizer-Preamp" by Charles Boegli, RADIO & TELEVISION NEWS, April, 1951.) Adherents of feedback-style equalizers may readily modify the stage in the manner illustrated by Lawrence Fleming in *Audio Engineering*, March, 1950 which is further improved by George Augspurger in his article "4 Problem



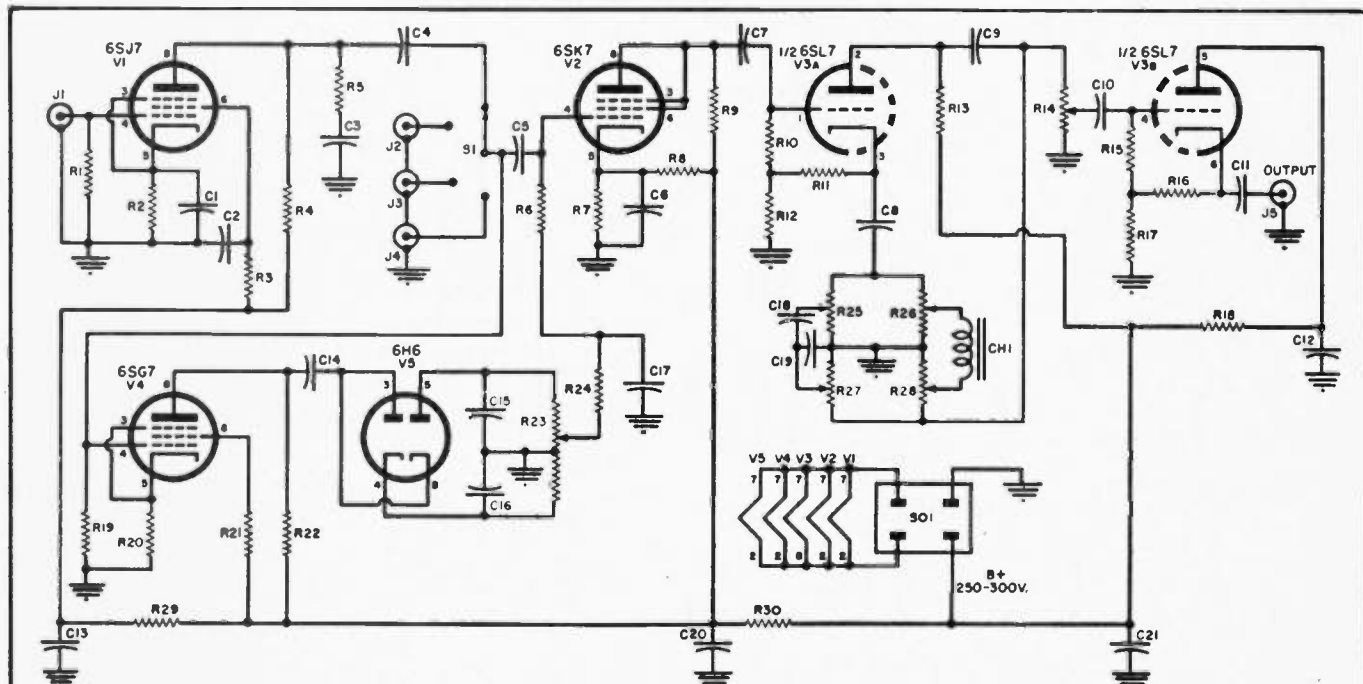
Under chassis view of unit. Note that terminal strips are used in construction to insure short leads. Left-to-right (bottom) are 6SJ7, 6SK7, 6SG7, 6H6, choke, and 6SL7.

Preamps" in RADIO & TELEVISION NEWS, November, 1952.

An input selector switch in the control amplifier is used to choose the desired signal, simultaneously grounding the other feeds to prevent leakage and

crosstalk. The selected signal is thus routed into the heart of the control amplifier which, of course, is the volume expander circuit. Here, three tubes are utilized for both volume expansion and compression, hence the

Complete schematic diagram of the control amplifier. Power for the unit is obtained from amplifier with which it is used.



R_1 —47,000 ohm, $\frac{1}{2}$ w. res. (or other proper load for pickup used)
 R_2 —2200 ohm, $\frac{1}{2}$ w. carbon or low-noise res.
 R_3 —2.2 megohm, $\frac{1}{2}$ w. carbon res.
 R_4 —470,000 ohm, 1 w. carbon or low-noise res.
 R_5 —10,000 ohm, $\frac{1}{2}$ w. carbon or low-noise res.
 R_6 —1 megohm, $\frac{1}{2}$ w. carbon res.
 R_7 —2700 ohm, $\frac{1}{2}$ w. carbon res.
 R_8 —18,000 ohm, 2 w. carbon res.
 R_9 —22,000 ohm, 1 w. carbon or low-noise res.
 R_{10}, R_{11}, R_{12} —470,000 ohm, $\frac{1}{2}$ w. carbon res.
 R_{13} —1000 ohm, 1 w. carbon res.
 R_{14}, R_{15} —22,000 ohm, 1 w. carbon res.
 R_{16} —500,000 ohm audio taper pot
 R_{17} —470 ohm, $\frac{1}{2}$ w. carbon res.
 R_{18}, R_{19} —22,000 ohm, 1 w. carbon res.
 R_{20} —1 megohm, $\frac{1}{2}$ w. carbon res.

R_{21} —2700 ohm, $\frac{1}{2}$ w. carbon res.
 R_{22} —330,000 ohm, 1 w. carbon res.
 R_{23} —100,000 ohm, 1 w. carbon res.
 R_{24} —1 megohm linear taper pot with center tap
 $R_{25}-R_{27}, R_{28}-R_{30}$ —Dual tone control pot (Thordarson #R-1068X)
 R_{29} —10,000 ohm, 1 w. carbon res.
 R_{30} —18,000 ohm, 1 w. carbon res.
 C_1 —50 μ fd., 25 v. elec. capacitor
 C_2 —5 μ fd., 400 v. capacitor
 C_3, C_4 —0.1 μ fd., 400 v. capacitor
 $C_5, C_6, C_7, C_8, C_9, C_{10}, C_{11}$ —1 μ fd., 400 v. capacitor
 C_{12} —25 μ fd., 25 v. elec. capacitor
 C_{13} —10 μ fd., 200 v. elec. capacitor
 C_{14} —25 μ fd., 200 v. capacitor
 C_{15} —25 μ fd., 400 v. capacitor
 C_{16} —20 μ fd., 450 v. elec. capacitor

C_{17}, C_{18} —0.1 μ fd., 200 v. capacitor
 C_{19} —1 μ fd., 200 v. capacitor
 C_{20} —0.3 μ fd., 400 v. capacitor
 C_{21} —0.1 μ fd., 400 v. capacitor
 CH_1 —22 hy. tone control choke (Thordarson 20C74)
 S_1 —Selector switch (Centralab 1400 series, type G wafer. All unused positions shorted to ground)
 J_1, J_2, J_3, J_4, J_5 —Phono jack
 SO —4-prong power connector
 V_1 —6SJ7 tube
 V_2 —6SK7 tube
 V_3 —6SL7 tube
 V_4 —6SG7 tube
 V_5 —6H6 tube

circuit function is called "expression."

The principle of operation is quite easily understood. A signal amplifier stage makes use of a remote cut-off, variable- μ tube. This tube characteristic is due to the special construction of the grid. Very little stage gain is needed, so the tube is triode-connected by joining together the plate, suppressor, and screen. A rather stiff fixed-bias is supplied to the cathode from the ten-volt point of a voltage divider. Under these conditions, the tube (which is a 6SK7, incidentally) will act as a conventional low-gain audio amplifier.

To obtain the "expression" effects, an additional bias voltage is applied to the grid along with the signal. This bias varies in direct accordance with the signal, thereby changing the stage gain as a function of the applied signal. This control bias is generated by the remaining two tubes of the "expressor" circuit.

In addition to feeding the 6SK7 signal amplifier, the signal input is also applied to the 6SG7 side amplifier. This pentode operates as a high-gain stage to amplify the signal, which is then connected to a twin-diode 6H6 rectifier. The amplified signal is fed to the plate of one section and to the cathode of the other section of the 6H6. The remaining plate and cathode connect to the extremes of a center-tapped potentiometer, with the tap grounded. The signal is rectified by the twin diode, while the potentiometer allows selection of the resulting positive and negative voltage used as d.c. control bias for the 6SK7 tube. Positive bias counteracts some of the initial fixed bias, increasing the stage gain to provide expansion. Naturally, the negative control bias is responsible for compression when desired. The degree of expansion or compression is determined by the setting of the control arm, or the effect can be removed completely at the midsetting of the potentiometer. At this setting, no control bias will be fed to the 6SK7 stage.

The RC values in the control bias circuits are such that the bias follows the general audio levels, rather than the audio itself. The time constants of the circuit are rather critical and are best chosen by careful listening tests. The attack or "build-up" time is determined largely by R_{21} and C_{11} , while the release or "fall-off" time is controlled by R_{22} , C_{12} , and C_{13} as well. The values indicated have been found suitable for the usual program material.

In addition to the phonograph equalization already covered, it is desirable to have control over the bass and treble portions of the audio spectrum, to adjust the response for room acoustics, speaker characteristics and, of course, listener preferences. Both boost and cut are required at each end of the range, with independent control, so dual tone controls are needed. Boegli has discussed fully the degenerative-type tone control in the June,

1951 issue of RADIO & TELEVISION NEWS, and in fact has included such a circuit in his "Improved Kappler Amplifier" which appeared in the October, 1953 issue of this magazine.

A degenerative tone control, with dual potentiometers utilizing Thordarson components, is included in this control amplifier. The action of the circuit has been well covered in the literature, and will not be repeated here. Suffice it to say that 16 db boost to 28 db attenuation at 60 cycles is possible, and 18 db boost or 35 db cut at 10 kc. is attained. At the mid-frequencies the stage operates with unity gain from the half 6SL7 used in the control stage. This results in approximately one to two volts r.m.s. output at this point. The master gain control is located after the tone control in order to improve the signal-to-noise ratio of the degenerative circuit.

Although miniature equivalents are available, metal tubes were used in the control amplifier with the exception, of course, of a glass 6SL7 output tube. Metal tubes were considered more desirable because they have, in general, proved to be less microphonic, longer lived, less subject to gas troubles, and are shielded inherently.

The inclusion of the compression action may be questioned by some readers. As stated, expansion of the dynamic range is the logical use of the "expressor" for playback purposes. However, this control amplifier is occasionally used to feed a disc recorder, and for this mode of operation the compression is advantageous. It may also find application as an automatic volume control for late evening listening or for background music when full dynamic range may be undesirable. However, such operation is definitely not to be misconstrued as "high fidelity" playback.

Construction

The complete control amplifier, as discussed, was constructed for developmental work. Reference to the photographs will show how this unit was built. A 13 x 5 x 3 inch aluminum chassis was used, attached to a 3½ x 19 inch rack panel. Symmetrically spaced along the back side of the chassis are the tubes and tone control choke. These are arranged in the order of their appearance in the circuit, i.e., 6SJ7, 6SK7, 6SG7, 6H6, choke, 6SL7. Four RCA phono connectors are used for the inputs, located along the pre-amplifier end of the chassis. A similar connector for the output, as well as a four-prong power connector occupies the opposite chassis end. Controls on the front panel are (from left to right): input selector switch, master gain control, expression potentiometer, treble control, and bass control.

Beneath the chassis, terminal strips are attached along the center of the open area between front and rear edges of the chassis. Most components going to ground are wired directly to

the associated sockets, while the terminal boards support interstage coupling capacitors, plate resistors, decoupling networks, and the like. Parts placement is not too critical, although layout is directed toward short leads as much as possible. Shielded wire is used for the volume control and tone choke leads.

Operation

A few comments on operation of the control unit may be in order. Power for the unit is obtained from the power amplifier. Approximately 250-300 volts should be supplied, as well as 6.3 volts a.c. for the filaments. Current drain is low and no motorboating has been encountered with the usual well-decoupled supply. The tone control choke can pick up hum from power transformers or turntable motors, so it is advisable to keep it isolated from strong a.c. fields.

As previously mentioned, amplifier input levels should be correct for the proper "expressor" operation. Signals at the selector switch should be 0.2 to 0.5 volt, while at the phono pickup jack a voltage of at least 0.01 volt is required. Voltages less than these values will be insufficient to create enough control bias for maximum expansion. On the other hand, voltages in excess of the stated amounts may create such a great bias that distortion can be caused at extreme settings of the expression control.

Proper input levels can generally be obtained by use of the volume controls in the tuners and tape recorder, or the inputs may be padded down to the correct value. Alternatively, level-setting potentiometers could be incorporated into the control amplifier itself, if so desired.

With input levels of the required values, a maximum control bias of five to eight volts should be developed on peaks, which will act against the ten-volt fixed bias on the 6SK7. This will produce more than usable amounts of expansion without driving the grid positive. Likewise, the same amounts of additional negative bias are available for compression. It is quite possible to adjust the compression bias to produce a constant peak level at the output. This particular a.g.c. action might be of interest to hams, since much can be said in favor of compression rather than peak clipping for automatic modulation control for amateur transmitters.

The finished control unit has been more than satisfactory in its intended operation, and very definitely serves to increase the "presence" in reproduction of recorded program material. Most listeners have considered the expansion a desirable addition to the home audio system.

While this control unit might appear to be a rather elaborate unit—most audiophiles will agree that it's worth the time and trouble involved in its construction.

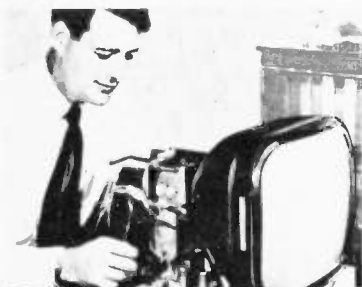


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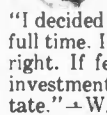
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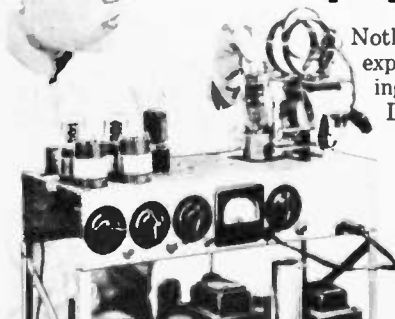
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CHAPTER 2

EQUALIZERS AND TONE CONTROLS

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SIMPLIFIED DESIGN OF FEEDBACK

EQUALIZERS

By ARTHUR J. ROSE



Fig. 1. Front panel view of the author's home-built "universal equalizer" which provides equalization for all type discs.

Design charts for an infinite variety of playback curves, an analysis of equalizer problems, and a suitable circuit.

EVERYONE has his own ideas about recording equalization. From the recording company brain trusts to the designer responsible for undoing the record cutter's handiwork, opinion is varied and outspoken. Consequently, high-fidelity enthusiasts are faced with a bewildering array of factual as well as fictional recording characteristics.

This problem, in part, stems from the big question mark that accom-

panies many curves published by other than official sources. Fortunately, this confusion is gradually disappearing and perhaps within a few years all will be an open book. This revelation is coming about as manufacturers realize the maturity of the present not-so-average record buyer and because of some persuasive arm twisting by audio equipment designers.

Of course there will be those who insist on being arty. Once the record owner is addicted to providing the "recommended" curve for some particular label, he will doubtless find that some super equalization has been applied to give a thrilling "high fidelity" effect. The orchestra sounds as if it were right inside your ear, for example.

While it is true that by merely applying the mirror of the recording response as generated by the pickup one is able to obtain the coveted "flat" response, one is limited by choice rather than by technology. Almost any curve can be effected. The problem lies in the question: "What curves shall I include?" Attempts to design universal switching arrangements are complicated by sheer numbers.

One solution that comes to mind is to use two switches. One sets the low-frequency response, while the other sets the high-frequency characteristic. With, for example, six position switches, thirty-six different combinations are possible! Somewhere within this number should be the correct response for the record in question. Imagination will supply variations of the number of switches and number of posi-

tions. Thus unlimited possibilities present themselves.

It is not entirely the purpose of this article to give a horse laugh to the equalization snafu. Most people are serious as to why "such and such" a curve is ideal. Well meaning groups have met the problem with sensible middle-of-the-road solutions. It is not funny to the guy trying to get his sound system to come up to expectations—and he never will unless he properly equalizes his records.

Rather than stress the difficulties of equalizer design, straightforward design procedures will be given. Then after the reader understands the easily-made graphical and arithmetic solutions that have been worked out, he can return to the more perplexing aspects. Suggestions for designing workable equalizers will precede the description of a "universal equalizer."

Circuit

Throughout the ensuing discussion it will be assumed that the feedback is great enough at all frequencies concerned so that the equalizer gain is dependent only upon the degree of feedback. Aside from simplifying the analysis, three important results stem from this assumption:

a. Wide variations of tube characteristics and supply voltages can be tolerated.

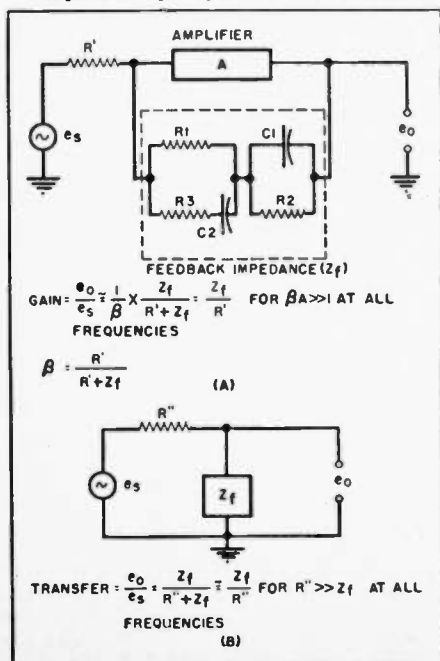
b. Distortion is reduced by an appreciable amount, even at the lowest frequencies.

c. Maximum possible bass boost is realizable.

In practice, this is accomplished by using a good deal more feedback than is apparently required. With a 500 cps turnover frequency, assuming 6 db-per-octave rise, the resultant boost at 50 cps is 20 db. In a linear system this is easily done. However, it has been shown elsewhere¹ that feedback systems with less than infinite ratios of open loop gain to effective midband gain flatten out at the lower frequencies. For close to ideal results, a ratio of 80 is satisfactory.

Although subject to controversy, ratios of 20 to 1 have been used². This

Fig. 2. A feedback equalizer and its equivalent passive network. Either type may be designed using the procedure given in text.



ratio is the basis for many passive equalizer circuits that give bass rise curves asymptotic to 26 db, reaching a maximum boost of 26 db at zero cps. In an ideal system where both recording and playback units are compensated by such a network, nothing is lost. In actual use, there is a definite loss of lower bass frequencies.

Fig. 2A shows a feedback equalizer. At midband, only R_1 of the complex feedback impedance, Z_f , is effective. Therefore, the midband gain is R_1/R' . For satisfactory operation, the ratio of open loop gain, A , to midband gain should be 80 or greater.

An equivalent passive equalizer, shown in Fig. 2B, should have R''/R_1 equal to 80 or more.

In either circuit, bass boost is produced by R_1 and C_1 and bass leveling by R_2 . High-frequency de-emphasis is produced by R_1 and C_2 . Introduction of R_3 lowers the slope of the de-emphasis curve and causes two other effects that will be discussed shortly.

A simplification of the feedback impedance, Z_f , in Fig. 3B shows effective networks throughout the spectrum. Equations relate circuit constants to applicable frequencies.

Ideal Equalization Curve

The exaggerated response curve of Fig. 3A shows the scope of the analysis. Were it not for f_c and f_r , the matter would reduce to the familiar simple case where R_2 and R_3 are not included. This gives a straight bass rise below the turnover frequency, f_1 , and a straight treble de-emphasis above the starting frequency, f_s . Both approach 6 db-per-octave in rate. Introduction of bass de-emphasis below f_c and treble de-emphasis with less than a 6 db-per-octave rate demands consideration of the entire feedback impedance. The RCA "New Orthophonic" and London ffr 78 curves are of the latter type, whereas the NARTB and AES are of the simpler type.

Constants for the feedback impedance for any desired curve shape are derived from the vector gain equations of the circuit in Fig. 2. These equations will be given as a matter of record. However, their solutions have been greatly simplified by plotting various parameters. As a result, circuit constants can be rapidly and easily found even by persons with a limited mathematical background. Sample problems following the derivation of the design charts will serve to illustrate this.

Impedance Equations

By virtue of the previously made assumption, the gain at any frequency is dependent only upon the feedback and therefore upon Z_f of Fig. 3B. Relative response then is merely the ratio of the Z_f 's at different frequencies. For convenience, all gain is referred to the gain where Z_f equals R_1 .

A further simplification is made by splitting Z_f into a low-frequency equivalent (Z_f greater than R_1), and a high-

frequency equivalent (Z_f less than R_1). Interaction, although slight, will have to be considered in some cases.

Gain at low frequencies can be expressed by

$$db = 10 \log \left[1 + r^2 \left(\frac{f}{f_1} \right)^2 + 1 \right] \dots (1)$$

where: $r = \frac{f_1}{f_c} = \frac{R_2}{R_1}$

if $R_2 = \infty$ then:

$$db = 10 \log \left[1 + \left(\frac{f}{f_1} \right)^2 \right] \dots (2)$$

Plotting equation (1) for various values of the parameter r (see Fig. 4) shows responses to be expected for changing values of R_2 . It can be seen that the effect of this resistance is to decrease the initial slope of the curve and then level it off below f_c . It has no effect on the turnover frequency.

In order to retain some perspective of the curves in familiar form, f_1 is shown as if it were 500 cps—the most prevalent turnover frequency. For other turnovers, the boost values hold equally well, but the curves have to be relocated by shifting the frequency axis. This will give the more commonly experienced picture.

There is no particular problem associated with the bass region. Usually the turnover frequency and the frequency where leveling occurs are given. Circuit constants to give these frequencies are obtained by their defining equations.

However, when a particular response is desired and f_c is not known, then the curves are most useful. Here, one must match the desired curve with one in Fig. 4 and extract a value of r .

Quite different is the situation encountered in the high-frequency region. Here, the addition of the series resistor, R_3 , has the effect of lowering the frequency at which de-emphasis starts as well as altering the curve shape. Since the exact starting frequency is more-or-less obscure, the frequency that yields a response 3 db down is used for the reference frequency, f_s . This applies, to some extent, in the bass region as the turnover frequency occurs with the response up 3 db.

Gain at higher frequencies can be expressed by $-db =$

$$10 \log \left[1 + \left(\frac{f}{f_1} \right)^2 \right] - 10 \log \left[1 + \left(\frac{f}{f_s} \right)^2 \right] \dots (3)$$

Here there is a subtraction of two curves. One is the de-emphasis produced by $10 \log [1 + (f/f_1)^2]$ and the other is a rise produced by $10 \log [1 + (f/f_s)^2]$. When $R_3=0$, the second curve disappears and a simple de-emphasis curve results. (See Fig. 6).

In terms of time constants, the high-frequency response is:

$$-db = 10 \log [1 + \omega^2 t_1^2] - 10 \log [1 + \omega^2 t_2^2] \dots (4)$$

where:

$$t_1 = R_1 C_2 \dots (5A)$$

$$t_2 = R_3 C_2 \dots (5B)$$

$$t = t_1 + t_2 \dots (5C)$$

$$f_s = \frac{.159}{t} \dots (5D)$$

These equations show how f_s is low-

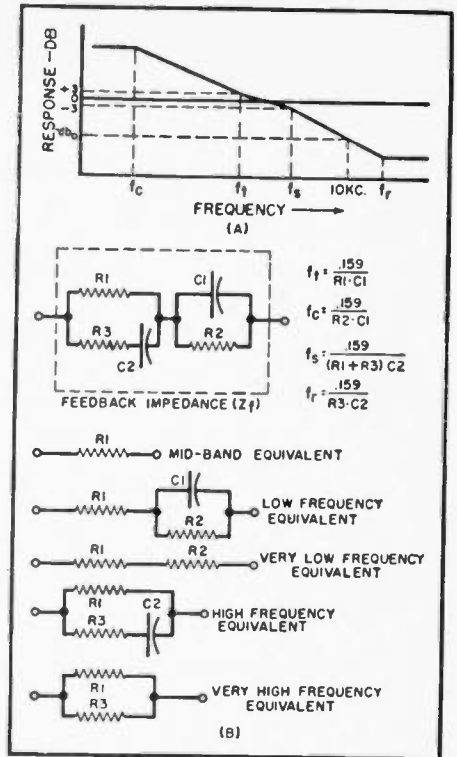


Fig. 3. (A) Idealized equalizer response curve with significant frequencies and response points shown. (B) Feedback impedance network. Equations relate components to the frequencies given in (A) of diagram.

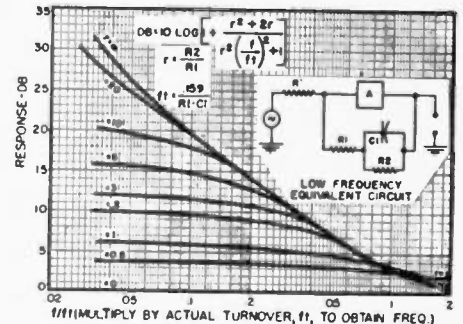
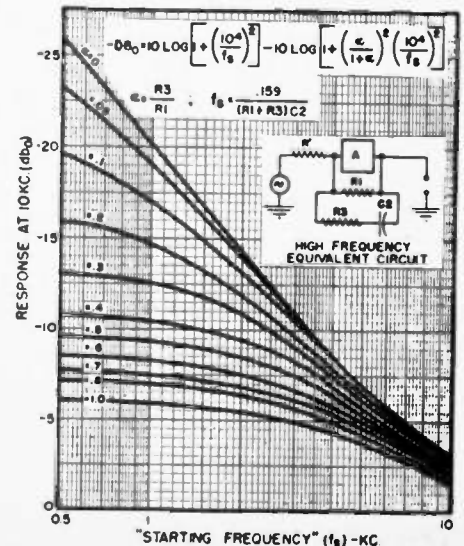


Fig. 4. Bass response curves for values of parameter r . Text describes how resistor R_2 is used to flatten the response curves.

Fig. 5. Response at 10 kc. for values of parameter α . Text describes how R_3 affects slope and shape of de-emphasis curves.



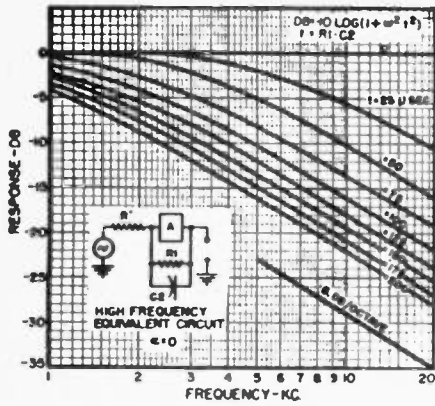


Fig. 6. De-emphasis curves when $\alpha = 0$.

ered as R_2 increases. If $R_2 = 0$, $t = t_1$. As R_2 is increased, t becomes greater by the increasing t_1 . Hence f_c becomes smaller.

If a factor α is introduced such that $\alpha = R_2/R_1$ it is evident that $t_1 = t/\alpha$, and $t = \alpha t_1/(1 + \alpha)$. Substituting in equations 3, 4, and 5 and combining,

$$-db = 10 \log \frac{1 + \left(\frac{f}{f_c}\right)^2}{1 + \left(\frac{\alpha}{1 + \alpha}\right)^2 \left(\frac{f}{f_c}\right)^2} \quad (6)$$

As customary in treating de-emphasis curves, the response at 10 kc. is specified. If this response, db_0 , is plotted versus f_c for significant values of the parameter α , one obtains a useful, general guide to de-emphasis curves. This is done in Fig. 5. Constants for a particular curve are easily obtained once α is known.

Design Procedure

1. Write down all available information about curve.
2. Draw curve on semi-log graph paper.
3. Determine value of R_1 from gain considerations given earlier. 100,000 ohms is a good, all-around value (for passive network, use 10,000 ohms).
4. Find C_1 . Since the bass turnover frequency, f_c , is always known:

$$C_1 = \frac{.159}{f_c R_1} \quad (7)$$

where: f_c is in cps, R_1 in megohms, and C_1 in microfarads.

5. Determine R_2 by one of the following methods:

(a) If bass curve is straight and approaches 6 db-per-octave, then $R_2 = \infty$

(b) If frequency where bass de-emphasis begins, f_c , is known, then

$$R_2 = \frac{.159}{f_c C_1} \quad (8)$$

or:

$$R_2 = \left(\frac{f_c}{f_c}\right) R_1 \quad (9)$$

where: f_c is in cps, R_1 and R_2 in megohms; C_1 in microfarads.

(c) If f_c is not known, match desired curve to one in Fig. 4. Interpolate if necessary. Fill in frequency scale according to turnover frequency (e.g., if f_c is 800 cps, then $0.1 f_c = 80$ cps, $0.2 f_c = 160$ cps, etc.) Determine r from chosen curve. Then

$$R_2 = r R_1 \quad (10)$$

$$f_c = \frac{f_c}{r} \quad (11)$$

6. Determine R_1 and C_2 by one of the following methods:

Case I. Curve approaches 6 db-per-octave droop. $R_2 = 0$

(a) If de-emphasis time constant, t , is known (e.g., 100 microseconds), then:

$$C_2 = \frac{t}{R_1} \quad (12)$$

where: t is in microseconds, R_1 in megohms, C_2 in micromicrofarads.

(b) If starting frequency, f_c , is known, then

$$C_2 = \frac{.159}{f_c R_1} \quad (13)$$

where: f_c is in cps, R_1 in megohms, C_2 in microfarads.

(c) If response at 10 kc., db_0 , is known, see Fig. 5. Find f_c for given db_0 on $\alpha = 0$ curve. Proceed as in (b).

Case II. Curve is less than 6 db-per-octave. R_2 is finite.

From starting frequency, f_c , and response at 10 kc., db_0 , determine suitable α from Fig. 5. Then,

$$R_2 = \alpha R_1 \quad (14)$$

and:

$$C_2 = \frac{.159}{f_c (R_1 + R_2)} \quad (15)$$

where: f_c is in cps, R_1 and R_2 in megohms, C_2 in microfarads.

Typical Curve Problems

I. AES (Fig. 7A)

This curve will be defined as having a straight 400 cps bass turnover frequency and treble de-emphasis of sim-

ply -12 db at 10 kc. $f_c = 400$ cps, $db_0 = -12$. R_1 is chosen as 0.1 megohm. Using Procedures 4, 5a, and 6 (Case Ic).

$$R_2 = \infty, R_1 = 0$$

$$C_1 = \frac{.159}{400 \times 0.1} = .004 \mu\text{fd.}$$

From Fig. 5, $\alpha = 0$, $f_c = 2650$

$$C_2 = \frac{.159}{2650 \times 0.1} = .0006 \mu\text{fd.}$$

II. RCA "New Orthophonic" (Fig. 7B)

Here there is a bass turnover frequency of 500 cps with de-emphasis starting at 50 cps. Treble de-emphasis is simply 75 microseconds. R_1 is chosen as 0.1 megohm. Procedures 4, 5b, and 6 (Case Ia) are used.

$$R_2 = 0$$

$$C_1 = \frac{.159}{500 \times 0.1} = .0032 \mu\text{fd.}$$

$$R_2 = \frac{.159}{50 \times .0032} = 1 \text{ megohm or}$$

$$R_2 = \frac{500}{50} \times 0.1 = 1 \text{ megohm}$$

$$C_2 = \frac{75}{0.1} = 750 \mu\text{fd.}$$

III. Columbia LP (Fig. 7C)

For purposes of instruction, the bass section of this curve will be defined as "looking like the $r = 5$ curve of Fig. 4." The turnover frequency is 500 cps. Treble de-emphasis is 3 db down at 1590 cps and falls at a rate approaching 6 db-per-octave. $f_c = 500$, $r = 5$, $f_c = 1590$ cps.

Again choose R_1 as 0.1 megohm. Procedures 4, 5c, and 6 (Case Ib) are used.

$$R_2 = 0$$

$$C_1 = \frac{.159}{500 \times 0.1} = .0032 \mu\text{fd.}$$

$$R_2 = 5 \times 0.1 = 0.5 \text{ megohm}$$

$$f_c = \frac{500}{5} = 100 \text{ cps}$$

$$C_2 = \frac{.159}{1590 \times 0.1} = .001 \mu\text{fd.}$$

IV. London frr 78 (Fig. 7D)

The bass curve is simply a straight 400 cps turnover and can be derived as outlined previously. However, the treble section is an example of Procedure 6 Case II.

De-emphasis emerges at 3 kc. and is down 5 db at 10 kc. Inspection of Fig. 7, curve D, shows 6 kc. as the 3 db down frequency. Using $f_c = 6$ kc. and $db_0 = -5$, Fig. 5 gives α as 0.4.

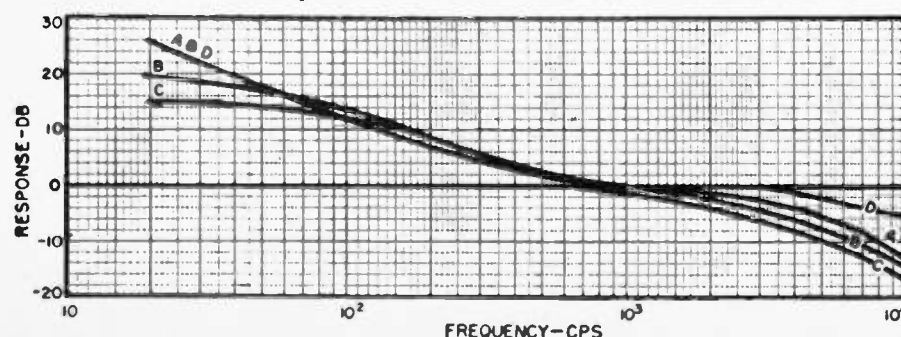
$$R_2 = 0.4 \times 0.1 = 0.04 \text{ megohm}$$

$$C_2 = \frac{.159}{6000 \times 0.14} = .00019 \mu\text{fd.}$$

Other Aids To Design

Although much time has been spent in preparing a comprehensive analysis of equalizer circuits, only the general results can be shown here due to space limitations. However, the designer can make full use of them by spending

Fig. 7. Playback curves. (A) AES, (B) Columbia LP, (C) RCA "New Orthophonic," and (D) London frr 78 rpm discs. See author's discussion in text for full details.



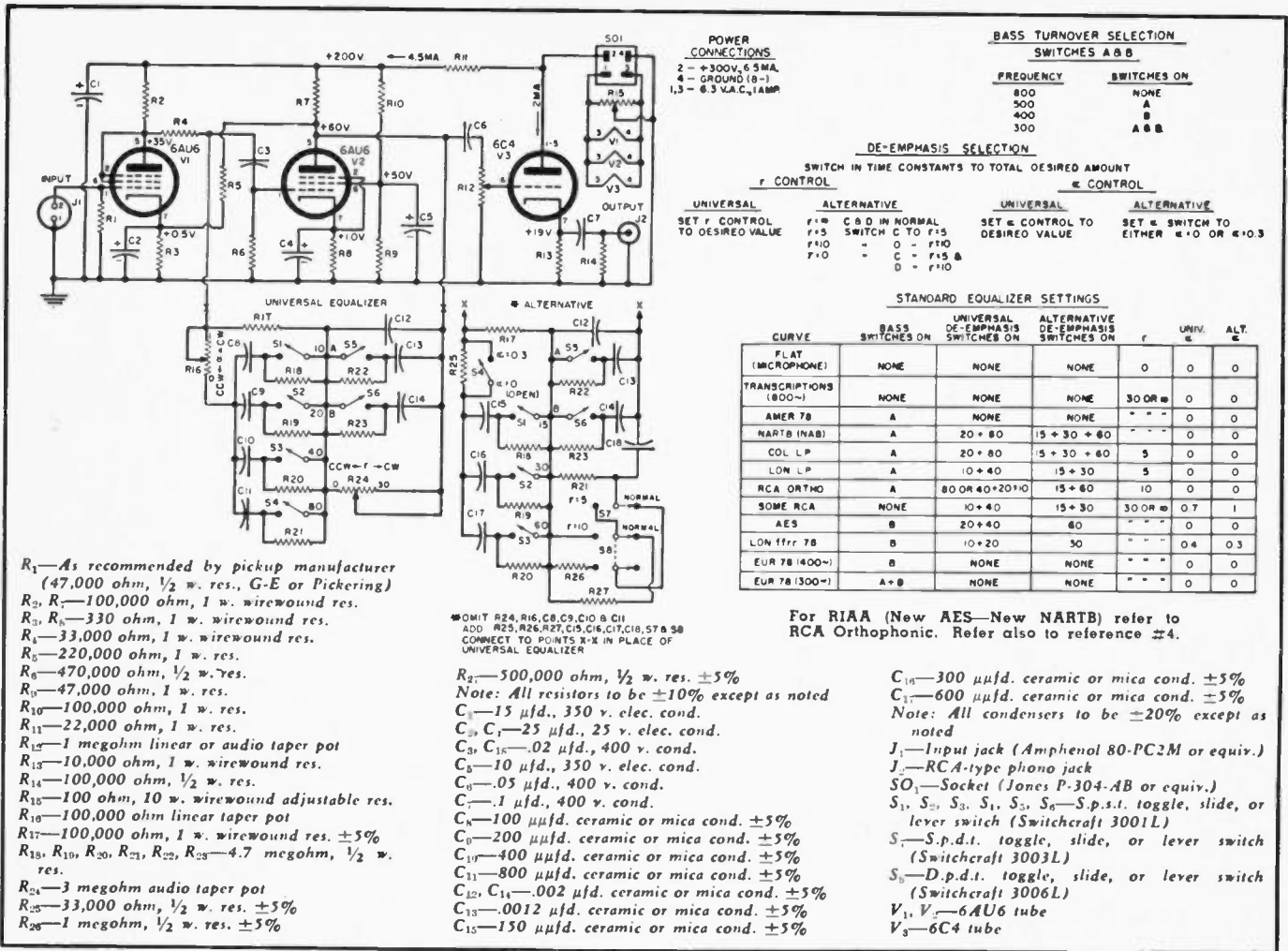


Fig. 8. Complete schematic for a "universal equalizer" and an alternative switching arrangement. Charts show the switching operations and suitable settings for the standard playback curves.

enough time to fully understand the general equations and the significance of various interrelationships. For example, the relationship

$$\frac{\alpha}{1 + \alpha} = \frac{f_s}{f_r} \quad (16)$$

is useful in making a quick diagnosis of a complex de-emphasis curve. By examining the curve, the 3 db down frequency, f_s , and the leveling frequency, f_r , can be determined. Using equation (16), α is obtained. Then R_3 and C_2 are found by solving equations (14) and (15).

In a great many cases simple de-emphasis, where $\alpha = 0$, is required. Fig. 6 shows actual response curves for this condition. C_2 for a particular response is found by solving equation (12).

It is to be pointed out that although there are but a few standard curves in which α is other than zero, the use of variable or switched resistors for R_{16} is most useful in producing modifications of de-emphasis slope often required to exactly equalize some recordings. This is equally true in the bass region where adjustment of R_{24} accomplishes equalization correction. See Fig. 8.

Practical Equalizers

Returning to the problem that was

presented earlier, it can be readily seen that the greatest difficulty now facing the designer concerns the choice of curves he is to use and the means he should use to switch them. Before he can begin, however, he must obtain accurate definitions of these curves in terms of the factors that determine their circuit constants. This task is made difficult because of conflicting information from different sources. The equalizer recommendations given in Fig. 8 are based on the most reliable information available to the writer. However, this does not preclude conflict between these sources or between those recommended in Fig. 8 and sources available to the reader.

Once a satisfactory agreement has been reached as to the nature of the curves, the designer can turn to the problem of switching. Most widely used methods fall into two groups:

- Single switching of complete playback curve
- Dual switching—one switch selects the bass while the other selects the treble

A more recent innovation falls into a third category:

- Sets of push-buttons or two position switches that individually select a complete curve and in combination give additive curves.

Following naturally from this is a

scheme that the writer feels to be most flexible and one that is slated for increased popularity:

- Dual sets of push-buttons or two position switches that give desired curves by adding separate bass and treble curves.

The distinction between methods (c) and (d) may not be too clear. Method (c), for example, uses a number of two-position slide switches. Each switch selects a particular complete curve such as the AES or London. Using more than one switch at a time adds together whatever curves are chosen and a new curve results. This method has an advantage of simplicity— 2^n curves can be formed with n switches—but curves formed by addition of standard curves may be useless and there are bound to be duplications and wide gaps in the range of curves.

Method (d) can handle as many curves as (c), but each curve represents a small incremental change from the preceding one. Thus the entire range of standard curves can be included as part of a system that offers extreme flexibility. To accomplish this, several switches are assigned to the bass section only and several more to the treble. This method is the basis of a universal equalizer.

Universal Equalizer

In the interest of greatest flexibility, a means for an ordered control of the various factors that contribute to a playback curve has been devised. Such an arrangement can be termed a "universal equalizer." With this device, one is able to effect any desired playback characteristic and an infinite number of variations about that characteristic.

Although originally meant as a means for confirming the preceding design data, the "universal equalizer" has proven invaluable in the investigation of the effects of factors such as acoustics at the recording location and losses in the recording process. These factors are instrumental in changing the effective recording characteristic.³ It has been found that many different recordings made by the same manufacturer require different compensation to sound right. Often the required compensation is a slight deviation from the generally accepted one. Sometimes it varies greatly. This type of correction cannot be accomplished by the usual preamplifier tone controls.

After careful consideration of the many factors that enter into high-gain preamplifier design, the configuration shown in Fig. 8 has been chosen. 6AU6's are exceptionally quiet and dependable tubes. The equalizer stage has a gain of 240, reduced to 3 at mid-band by feedback. Preceding this stage, a triode connected 6AU6 with a gain of 30 brings the over-all mid-band gain to 90. Thus the unit can directly drive a final power amplifier if need be. Normally, a switching and tone compensating amplifier is used between the two.

Other combinations of triodes and pentodes can be successfully used, but most combinations will not give as high a gain.¹

Equalizer switching is negotiated by either the "universal" or "alternative" methods of Fig. 8. Both schemes use the same bass turnover selector switches. The first method gives de-emphasis from 0 to 150 microseconds in 10 microsecond steps, whereas the alternative goes from 0 to 105 microseconds in steps of 15 microseconds.

Selection of α and r differs in that the "universal equalizer" offers continuously variable control of these factors over the entire wide range while the alternative gives the extreme ends and popular middle settings. The "universal" α and r dials can be panel calibrated, if desired, with an ohmmeter. In both cases, the calibration marking represents the fraction or multiple that is placed into the circuit by R_3 and R_{16} . Some constructors may wish to calibrate their dials in terms of $\frac{1}{r}$ and $\frac{1+\alpha}{\alpha}$ in order to more readily aid mental calculation of f_c and f_r .

$$f_c = \frac{1}{r} f_s$$

$$f_r = \frac{1+\alpha}{\alpha} f_s$$

Potentiometers for these controls should be wired so that a clockwise rotation gives maximum response.

Operation of either equalizer is conducive to rapid and easy selection of any desired curve. Bass turnover frequencies are selected by switches A and B. Treble de-emphasis is selected by throwing the switches marked directly in microseconds. The total de-emphasis is the sum of the individually chosen ones. α and r are directly selected.

It may occur to the reader that these schemes rely upon the user's detailed knowledge of playback curves.

For this reason, these means of selection are unsuitable for the uninitiated. Charts such as the one that accompanies the schematic can be prepared for their use until the user becomes familiar with the scheme of things.

Component tolerances must be carefully watched if results are to be as expected. Commercially available condensers will deviate as much as 100% from their marked value and should be bridged beforehand. Resistors should also be bridged for very best results.

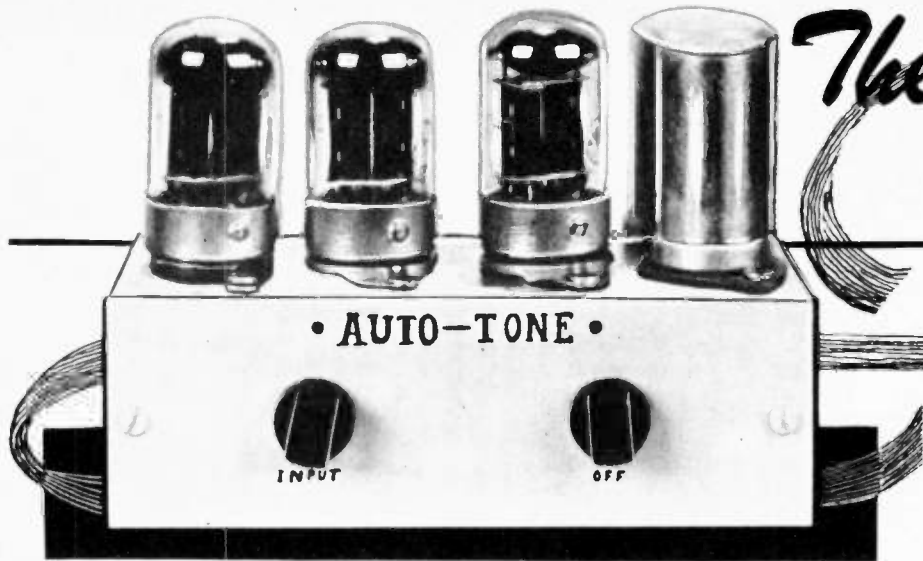
Soldering of these components must be done with care lest they change value with excessive heat. Do not overlook either of these suggestions as most complaints of poor operation can be traced directly to carelessness on these counts.

Since the output impedance of the equalizer stage is fairly low at the higher frequencies, the cathode follower output is not necessary where short output cable lengths will be used or where the unit forms part of a complete equalizer-preamplifier. However, for general use, the cathode follower is desirable.

Liberal use has been made of high resistance "click suppressors" across all switching points. R_3 forms part of a rumble filter.¹ It is recommended that the reference 1 be consulted by those wishing to construct the "universal equalizer." Suggestions for building the equalizer-preamplifier described there are applicable here as well. Some may prefer to use the equalizer schemes of this article in place of the one in reference 1.

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3. Wortman, Leon A.: "Audio Equalization", RADIO & TELEVISION NEWS, May 1953
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Front panel view of the "Auto-Tone." Three 7F7's are used although almost any twin-triode can be substituted if desired.

In effect a loudness control—uses three tubes in a single circuit. Provides automatic bass and treble control to compensate for the Fletcher-Munson effect.

NEARLY everyone is aware that the effective response of the human ear is not the same at all volume levels, it being more nearly flat at high volume levels, and having less sensitivity to the lower and higher ends of the audio spectrum on low intensity sounds. Because the design engineers were aware of this fact they introduced the tapped volume control which compensates for the reduced sensitivity of the ear to the lower frequencies at low volume control settings by boosting the bass. The volume control, in some cases, has combined a treble-boost control with a bass-boost on the same shaft as the volume potentiometer. Obviously, a sound of constant volume could be handled satisfactorily by one of these arrangements, but speech or music, or any of the sounds normally encountered in audio amplifiers do not have constant volume, but may, and do, have passages, words or syllables of much lower than average level. The need, therefore, is for some automatic method of response control: An amplifier that would introduce considerable bass and treble boost at low inputs, and some method of automatically reducing the amount of boost as the volume increases.

In the simple bass-boost circuit shown in Fig. 1A, the setting of R determines whether the full capabilities of boost are used or not. When this resistor is at its maximum (assuming its maximum resistance is equal to, or greater than, the reactance of C at the lowest frequency to be amplified) the boost will also be greatest. When R equals zero, C is shorted and the purely resistive network of R_1 and R_2 does not discriminate frequencies and the response will be essentially flat. If R

is replaced by the plate-to-ground resistance of a vacuum tube, varying the d.c. bias will vary the plate current and likewise its effective resistance. It is possible, then, to control the bass boost by varying the bias on the grid of such a control tube. If the bias is obtained by rectifying the incoming audio voltage, the amount of bass boost can be controlled by the sound level.

If control of the bass was all that was desired, the basic circuit of Fig. 1B would suffice. Because the plate voltage on V_1 , and hence its plate resistance will not, in practice, drop to zero, and its effective maximum resistance can not exceed the value of its plate resistor, the ratio of maximum to minimum resistance across C will not be nearly as high as R in Fig. 1A. This means that with the circuit designed to provide a nearly flat frequency response with V_1 conducting its maximum (the plate resistance of V_1 at minimum), the maximum bass boost that can be expected will be just slightly more than 6 db at 100 cycles over the response at 1000 cycles. If no corresponding treble boost is incorporated, that amount will do. But if treble boost is also used, a boost of both treble and bass of about 12 db above the 1000 cycle level should be available at low volume levels. This can be obtained, as will be explained later. Fig. 1D shows a typical curve for the circuit of Fig. 1B, with the control tube, V_1 , inoperative (low volume in-

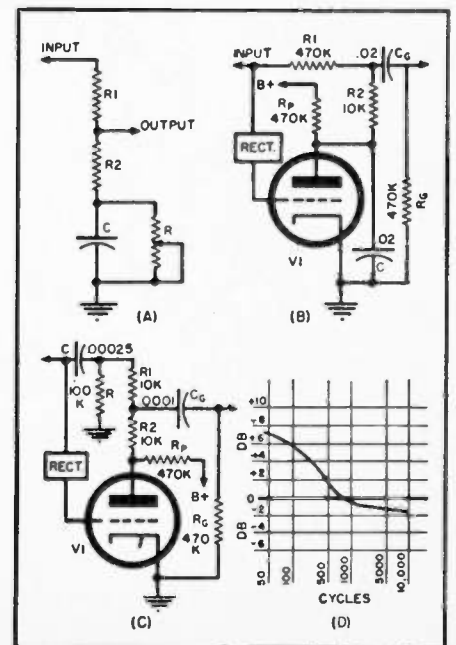
Fig. 1. (A) A simple bass-boost circuit. (B) Simplified bass control circuit using a tube. (C) Basic treble-boost control circuit. (D) Curve for the circuit of (B). See text for details on these circuits.

The AUTO-TONE

By ED. C. MILLER

put). Fig. 1C is a basic treble-boost control circuit, wherein the amount of boost is controlled by the components C and R , and the control tube attenuates all frequencies in the same proportion, instead of leaving frequencies that are not boosted unchanged while varying the amount of boost, as was done in the bass circuit just described. For that reason it could not be used alone, but must be used in conjunction with the bass booster or linear attenuator. This method of control for the high frequencies was chosen because the control tube can be operated in the same fashion as the bass control tube with its cathode grounded, and because it makes it possible to obtain the desired 12 db boost for both the highs and the lows. This is accomplished by applying the output of the low boost and the high boost to a mixer tube in such a manner that they will be in a ratio of 2:1 in amplitude and 180 degrees out-of-phase at 1000 cycles, with the control tubes inoperative. This gives us an additional 6 db attenuation at 1000 cycles which, added to the 6 db boost at 100 cycles, results in an effective boost of the lows and highs of about 12 db.

The output of Fig. 1B is approximately 3% of the input amplitude, with a phase shift of minus 40 degrees at 1000 cycles, when the plate resistance of V_1 is 12,000 ohms, which it is



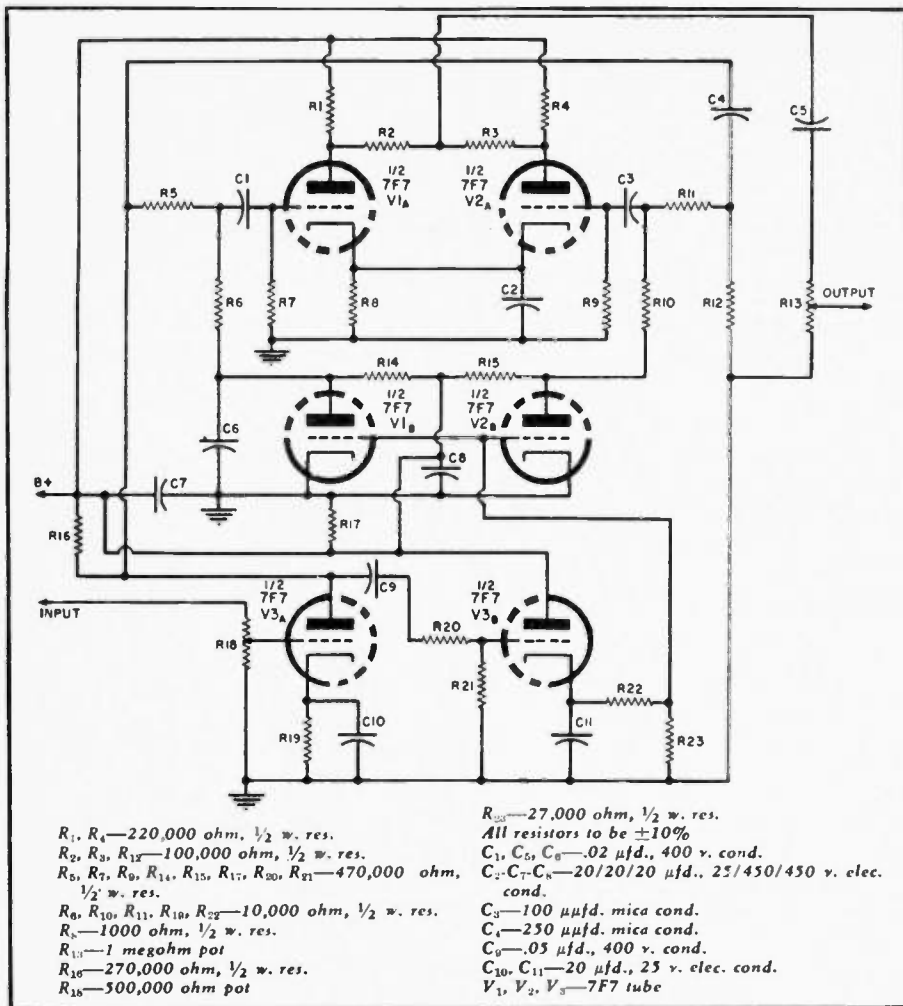


Fig. 2. Complete schematic of the "Auto-Tone." See text for possible alterations.

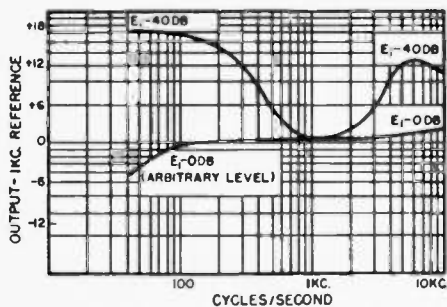


Fig. 3. Measured curves of the "Auto-Tone."

when there is no output from the rectifier. A single condenser-resistor network such as RC in Fig. 1B cannot produce more than a positive phase shift approaching 90 degrees. In fact the phase of the voltage at the junction of R_1 and R_2 is slightly less than 80 degrees at 1000 cycles. It is necessary to use another RC combination, namely C_9 - R_9 to provide the additional 60 degree shift to make the output of Fig. 1C directly out-of-phase with the output of Fig. 1B. The output phases of these two circuits do not change

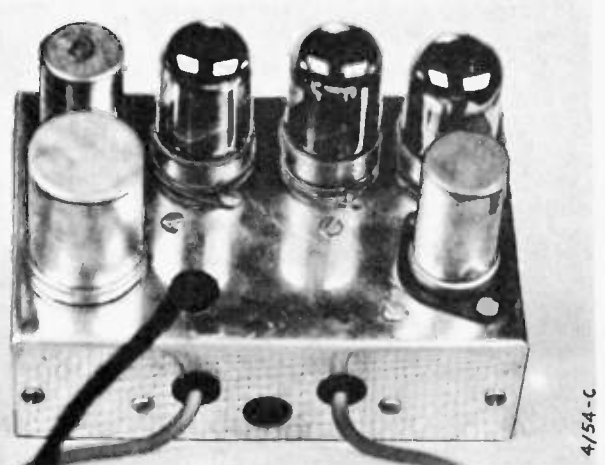
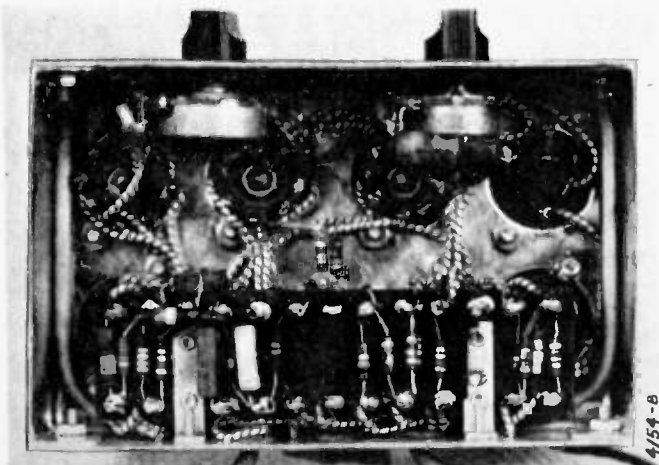
correspondingly with frequency, so that at some frequency above 1000 cycles, where the voltage outputs are equal in amplitude, they are no longer out-of-phase, so there is no frequency where the resultant output is less than the output at 1000 cycles.

In the "Auto-Tone" amplifier pictured, three type 7F7 tubes were used. If octal types are preferred by the constructor, type 6SL7GT tubes may be substituted with no circuit or parts value changes. Almost any twin-triodes can be used with this circuit, but the values of R_{22} , R_{21} , R_{14} , and R_{15} would have to be changed if the characteristics of the desired tube types differ much from the 7F7. In the complete schematic of Fig. 2, the first section of V_{3A} is used as a straight "class A" amplifier, the output of which is fed to the bass-boost network, the treble-boost network, and the control rectifier. The rectifier is the second section of V_{3B} , and is of the high-impedance, cathode-follower type. The static conditions of this tube are such that it has cathode bias slightly higher than if the tube were operating as a "class A" amplifier. Audio signals of small amplitude, i.e., whose negative peaks do not cut off plate current in the tube, will cause no change in the d.c. voltage at the cathode. As the applied signal strength increases beyond cutoff, the negative portion of the audio will have less and less of a nullifying effect, and the voltage on the cathode will increase in a positive direction. C_{11} is merely a filter condenser; R_{22} limits the grid current in the control tubes, and R_{23} determines the voltage on the grids of the control tubes at minimum signal level. V_{1B} is the bass-boost control tube and V_{1A} is the bass mixer. The treble-boost control and mixer are V_{2B} and V_{2A} , respectively.

Power requirements are not critical. The heater drain is .9 amp at 6.3 volts, and the plate current drain is less than 5 ma. at any voltage from 250 to 400 volts. 300 volts is about optimum. Terminal board construction, such as is pictured makes construction, testing, and any future servicing, easy and is to be recommended. As long as the usual care in wiring is observed, no

Under chassis view. Terminal board construction is used.

Rear view showing how chassis is laid out for compactness.



troubles should develop. As it actually makes little difference whether the minimum boost frequency is exactly 1000 cycles or not, all components need have no better tolerance than 10%. If this frequency is not as close to 1000 cycles as the constructor would desire when the unit is tested, changes in C_1 and/or R_1 may be made up to 50% of their value to correct it.

The "Auto-Tone" was designed primarily for use with record players, and when so used the output of the cartridge or preamp is connected across

R_{15} , and the arm of R_{12} is connected to the amplifier input. It should be noted here, perhaps, that this circuit does not replace the tone control circuits in the amplifier or preamp. With full input, the "Auto-Tone's" response is essentially flat from 100 cycles to 10,000 cycles and more, and at that input the tone controls on the main amplifier should be adjusted for the most pleasing results at full room volume. The main amplifier controls may be left at those settings, and R_{15} used to control the volume. The amount of bass and

treble boost will then be controlled directly by the amount of sound coming from the speakers, being maximum when the level is lowest. Fig. 3 shows the measured curves of the "Auto-Tone" pictured.

The preceding information, along with the schematic, should explain the operation. The curves show the measured results. Hearing it in use is the only real way to appreciate the added listening pleasure the "Auto-Tone" can contribute.

-30-

AN OUTBOARD BASS-BOOSTER

By ALLAN M. FERRES

Add a bass boost circuit to your equipment without changing existing wiring. Power for booster comes from the amplifier.

"EVERYTHING sounds fine, but I'd like just a little more bass."

Sometimes this comment is heard after a custom installation is completed, and if you hesitate to dig into factory-made equipment to correct the trouble, this booster will produce the

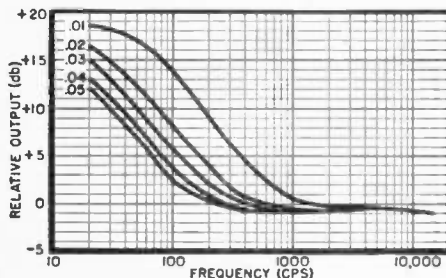


Fig. 1. Response of booster with different condenser values from .01 to .05 μ fd. The zero level is .775 volt in this instance.

required additional bass with a minimum of effort on your part. It is simple to demonstrate and often can be sold to old customers after other repairs or adjustments have been made on their sets.

The circuit is simple and the few parts required can be mounted in an aluminum can 2 x 2 x 4 inches, small enough to fit into a corner of the amplifier or tuner compartment of almost any set. The booster uses a 6C4 triode as a resistance-coupled amplifier with an RC frequency-response-correcting network in the grid circuit. The tube furnishes just enough gain

to compensate for the loss in the network so that the mid- and high-frequency amplification is zero. The booster can be connected to the input of the power amplifier so as to increase the bass response for all the signal inputs or connected to the output of the record player, AM-FM tuner, or TV set, if only one of these requires the additional bass. In order to reduce the number of parts, no controls are included.

Power to operate the booster is obtained from the power amplifier of the installation, either by means of an adapter plug, placed under a 6V6, 6L6, 6K6GT, or similar tube, or by running flexible leads to the underside of a socket. The plate voltage is taken from the screen pin of the tube. The ground connection is obtained from the shield of the output lead, so no other lead need be run to the chassis. If motor-boating is experienced due to poor voltage regulation of the output tube's screen supply, then the plate voltage for the booster can be picked off some other well-filtered high-voltage point in the circuit.

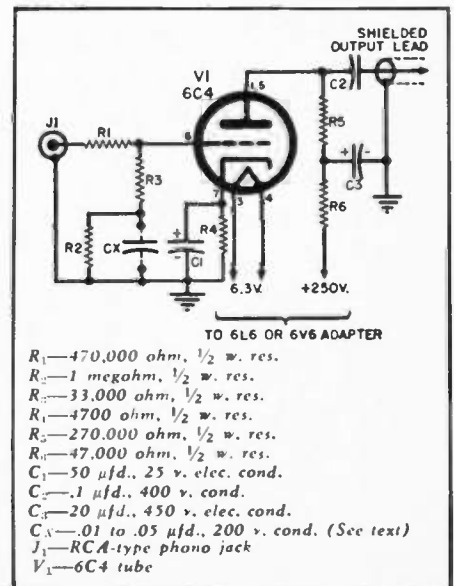
C_X is mounted to a dual tie point, one lug connected to R_2 and R_3 and the other lug grounded. After the booster is installed, various values of C_X should be tried until the desired bass response is obtained. Fig. 1 shows the response that is obtained with values from .01 to .05 μ fd. If less boost is required after the proper value of C_X is selected, C_X can be

shunted with a resistor, the value being determined by test. 100,000 to 470,000 ohms are typical values.

The booster should be mounted so that only a foot or two of output cable is required in order to prevent a loss of high-frequency response. Only .15 amp. at 6.3 volts and 1 ma. at 250 volts are required to operate the unit, which will not overload the power amplifier power supply.

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Fig. 2. Complete schematic of the outboard booster. It may be mounted in a 2"x2"x4" can, fitting into the equipment cabinet.

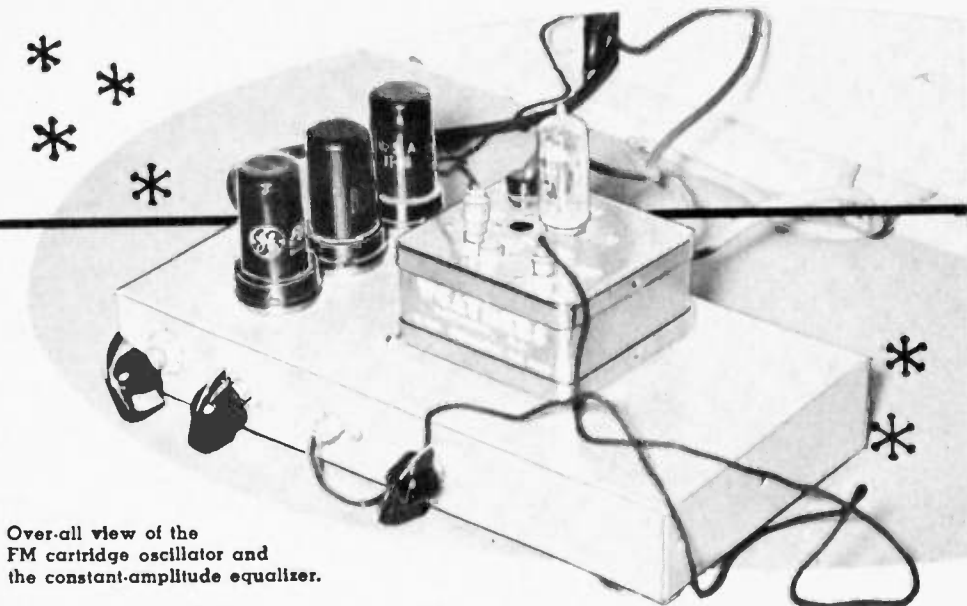


AN EQUALIZER FOR FM PICKUPS

By

JAMES A. MITCHELL

A simple and versatile equalizing circuit which provides correct record compensation for all makes of capacitance FM pickups.



Over-all view of the FM cartridge oscillator and the constant-amplitude equalizer.

A NEW TYPE of phonograph pickup cartridge which has been growing in popularity among hi-fi fans is the *Weathers* FM pickup. The electrical properties of this type of pickup make possible a new equalizing circuit which has many advantages over the circuits generally used in preamps designed for magnetic cartridges. The principles and practical construction of such an equalizer will be described.

First let us consider how the FM pickup generates a voltage in comparison with the magnetic type pickups. All of the magnetic cartridges create a voltage in a coil by changing the magnetic flux through it. The magnetic field is varied through the coil by the movement of the stylus in any of several arrangements. Since the voltage in the coil is determined by the rate of change of the magnetic flux, the voltage is proportional to the rate of movement of the stylus. This system

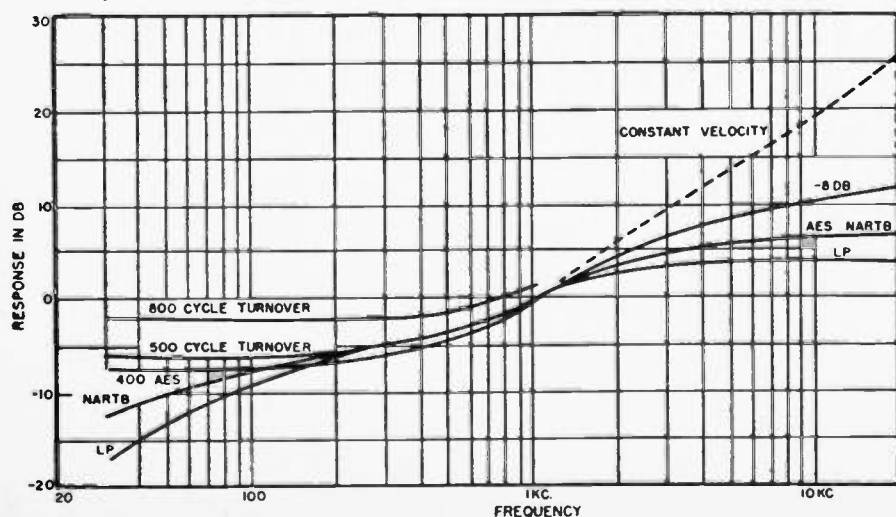
produces a frequency response which is referred to as constant velocity. In the FM capacitance pickup the movement of the stylus creates a change in capacitance between the stylus and a nearby plate. This capacitance is part of the circuit of a radio-frequency oscillator. As the capacitance changes so does the frequency of oscillation. This frequency-modulated signal is converted to an audio frequency by a simple detector. Since the capacitance of the stylus and plate is a function of the distance between them, the voltage produced in the detector is proportional to the distance or amplitude of the stylus movement. The frequency response of this system is termed constant amplitude.

Constant amplitude response differs from constant velocity response in that it has greater output as the frequency is decreased and conversely lower relative output as the frequency is in-

creased. There is no fundamental advantage to either system of response if the preamplifier correctly equalizes the response curve. Actually records are made with a recording curve which is a composite of constant amplitude and constant velocity recording methods. Variations in the recording curves make an adjustable equalizer desirable.

Almost all existing preamplifiers with adjustable controls have been designed for magnetic pickups. If it is desired to use an FM pickup with these preamplifiers the response of the FM pickup must be changed to constant velocity before feeding it into the unit. This is easily done by inserting a small condenser, 50 or 100 $\mu\text{fd.}$, between the oscillator output and the preamplifier input. *Weathers* and most preamp manufacturers show how to do this. However certain inherent advantages of the FM pickup are not fully utilized with this method. First, the FM oscillator has an output of about 500 mv. which, since it has a constant amplitude response, is also the approximate output of the very low frequencies. This should be compared with magnetics which have an output of 15 to 60 mv. at 1000 cycles but only 1 to 5 mv. at the very low frequencies. This makes it possible to obtain an excellent signal-to-noise and hum ratio with the FM pickup. When a condenser is used to convert the response to constant velocity the bass frequencies are reduced to about the same level as the magnetics and the usual elaborate precautions to minimize hum and noise are required in the preamp. Second, it is difficult to design a magnetic-type preamp with the full 6 db-per-octave bass boost. With constant amplitude response available directly from the FM pickup, there is no need to boost the bass at all. So why convert it to constant velocity and then have to

Fig. 1. Required equalization curves for a constant-amplitude pickup to provide flat response for various recording methods. See text for details.



boost the bass back to constant amplitude.

Fig. 1 shows the response curves a preamp should have to provide the proper equalization for a constant amplitude pickup such as the FM type. Fig. 2 shows, for comparison, the same equalization curves as they are required in a preamp for magnetic pickups with constant velocity response. Note that actually less correction is required for the most used recording methods with the constant amplitude pickup. Also that while the usual constant velocity preamp boosts the bass and cuts the treble, the constant amplitude preamp cuts the bass slightly and boosts the highs. With modern high-frequency pre-emphasized recordings little additional boost in the preamp is necessary.

The fundamental RC circuit to produce the equalization needed in the constant amplitude preamp is shown in Fig. 3. C_1 is a small condenser which, in conjunction with R_1 , provides the high-frequency boost. R_1 is a resistor which bypasses C_1 at the low frequencies and controls the turnover to constant amplitude response in the bass. R_2 is a resistor which, depending on its value, limits the high-frequency boost and allows a variable high-frequency control. C_2 is a larger condenser which, if present, restricts the very low frequencies to provide the low bass attenuation required for LP and RCA "Orthophonic" recording curves.

One interesting feature of this circuit in comparison with ordinary preamps is that the elements which are to be varied to control the bass turnover and the high-frequency pre-emphasis are the two resistors R_1 and R_2 . In nearly all magnetic preamps it is the condensers which must be varied. This requires a switching arrangement with many components. With this constant-amplitude equalizer two variable resistors can provide all the turnover and treble control necessary. A switch cutting in C_2 can modify the low bass for the LP pre-emphasis.

Fig. 4 shows a practical set of values for this equalizing circuit. This network can be used either directly connected to the FM detector and before the first amplification stage or it may be used following a triode amplification stage. It provides nearly exact equalization for all the standard recording curves. Fig. 6 shows curves actually obtained at the various settings of the controls. Remember that since the controls are continuous any intermediate high-frequency pre-emphasis correction or bass turnover can be obtained. The only equalization setting which is somewhat short of theoretical is when R_2 is set at zero resistance for the flat constant-velocity curve. The circuit shown is 1.3 db below theoretical at 10,000 cycles and 3.5 db below 20,000 cycles. Since this curve is used only for playing frequency test records or old 78's with no highs anyway this is of little practical consequence. Halving all of the condenser values and doubling the size of all the resistors

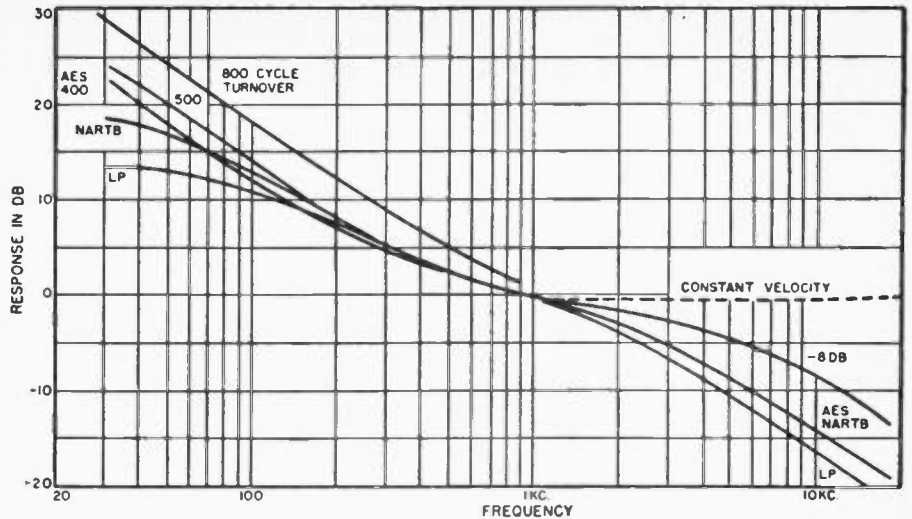


Fig. 2. Required equalization curves for the conventional constant-velocity (magnetic) pickups. Compare this curve with that shown in Fig. 1.

in the circuit (except R_1 , which should not be changed) will make the flat position correct to within 1.3 db at 20,000 cycles but will double the voltage loss in the equalizer. Since the equalization is nearly exact on all the lower settings of the high-frequency control the values given were preferred.

The size of variable resistor R_2 and fixed resistor R_1 have been selected to provide a variable bass turnover of from 280 to 1000 cycles, with a complete swing of the control knob. The value of variable resistor R_2 will provide a correction for records made with no high-frequency pre-emphasis to those made with 20 db pre-emphasis at 10,000 cycles. Switch S_1 has three positions for exact adjustment of low bass pre-emphasis. Position 1, with no condenser, is flat for all constant-amplitude turnovers. Position 2 provides the correct NARTB and RCA "Orthophonic" bass. The bass turnover should be set for 500 cycles on these records. Position 3 matches the LP recording

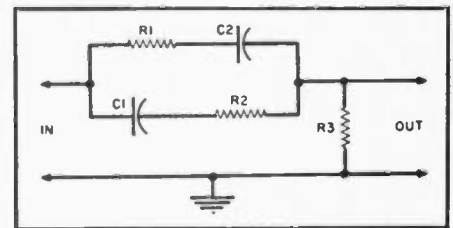
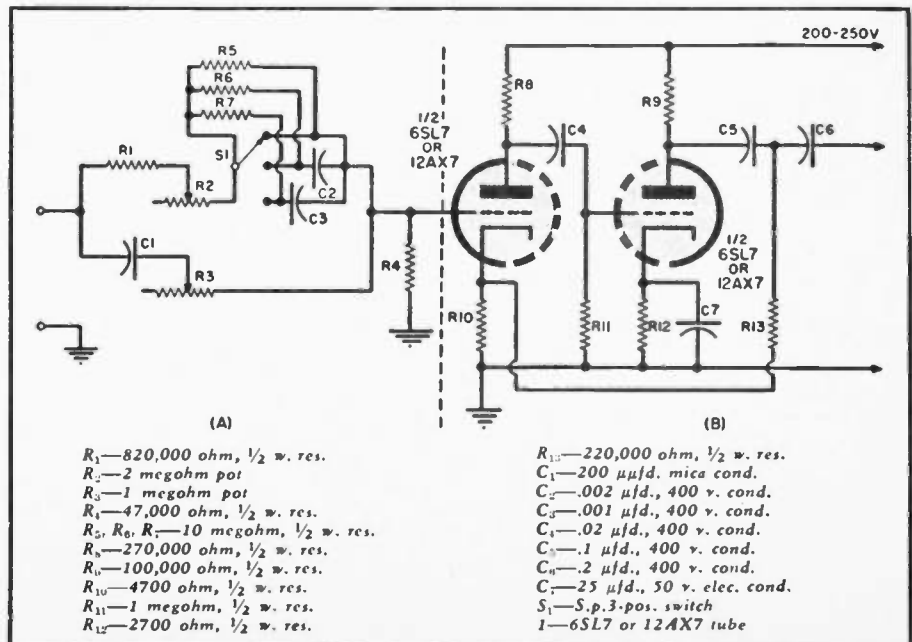


Fig. 3. The fundamental RC circuit for constant-amplitude equalization.

curve with the turnover also set at 500 cycles.

This equalizing circuit has a loss of 24 db at 1000 cycles. The equalized output is about 30 mv. at all frequencies if the circuit is connected directly to the oscillator. This is still about 20 db above the signal level of magnetic pickups in the 30 to 60 cycle range. Hum pickup in the amplification stage following the equalizer is, therefore, 20 db less than when the same tube is used as a magnetic-type equalizer. Fig. 4B shows a simple two-stage voltage am-

Fig. 4. (A) Circuit of adjustable equalizer and (B) the voltage amplifier circuit.



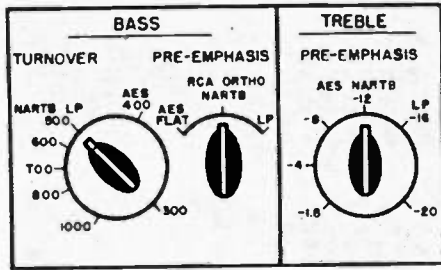


Fig. 5. Sample control panel for the constant-amplitude equalizer unit.

OUTPUT (100 vs 1000—)	BASS TURNOVER
-10 db	300
-8 db	400 (AES)
-6 db	500 (NARTB, LP)
-2 db	800

Table 1. Bass turnover frequency adjustment with 100-cycle output vs 1000 cycles.

OUTPUT (10,000 vs 1000—)	H.F. CORRECTION
+18 db	-2 db
+16 db	-4 db
+12 db	-8 db
+8 db	-12 db (AES, NARTB)
+4 db	-16 db (LP)
+0 db	-20 db

Table 2. High-frequency pre-emphasis control settings (10,000 cycles vs 1000 cycles).

plifier which is suitable for bringing the voltage up to about 1.3 volts, ample for any high-level input. It is outstanding for flat frequency response, low distortion, and low output impedance due to a high ratio of feedback.

The continuous controls should be calibrated and marked in the manner of Fig. 5. Calibration can be accomplished in one of several ways. If an

audio oscillator and an a.c. vacuum-tube voltmeter are available it is possible to trace the frequency response at a number of settings. It is more direct to use the following scheme. Set the high frequency control on maximum high-frequency boost and the low-frequency pre-emphasis switch on flat (no condenser). Adjust the audio oscillator and vacuum-tube voltmeter to get a standard reading at 1000 cycles. Then lower the frequency of the oscillator to 100 cycles. Adjust the bass turnover control (see Table 1) to get output in terms of db below the standard 1000 cycle reading and mark the controls with corresponding turnover.

The high-frequency control is adjusted in the same general manner and marked in terms of correction for high-frequency pre-emphasis. The oscillator is turned to 10,000 cycles and the readings of Table 2 relative to the standard 1000 cycle tone correspond to the given marking of the high-frequency control.

The response curves obtained by switching in C_2 and C_3 should be checked against the curves of Fig. 6. Some selection of condensers may be required to accurately match the desired response with your FM equalizer.

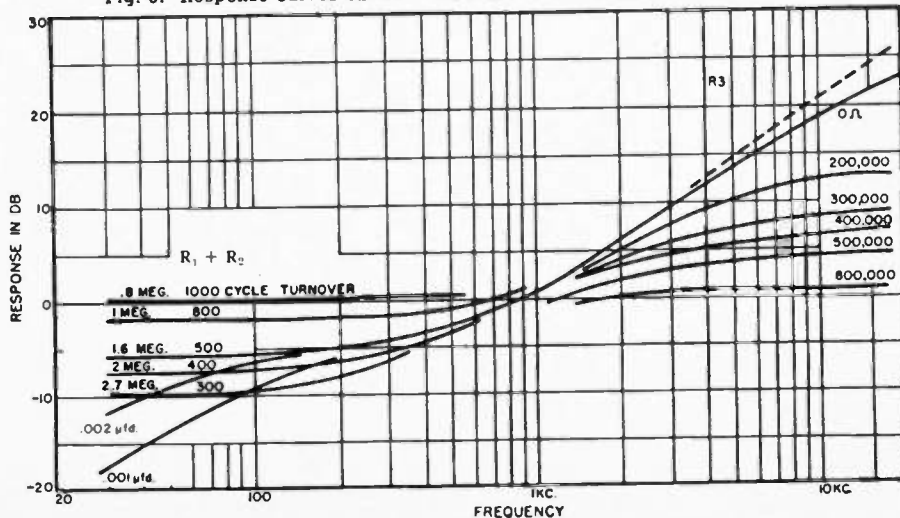
The controls can also be calibrated quite well if only an ohmmeter is available. In this case C_1 should be selected within $\pm 5\%$ of the correct value. The various markings for the controls can then be found from the measured resistances of the variable resistors according to Table 3. Since both R_1 and R_2 comprise the bass turnover control, the resistance measurement should include these two in series for proper calibration.

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BASS TURNOVER CONTROL		H.F. PRE-EMPHASIS CONTROL	
$R_1 + R_2$ (in megohms)	Turnover (cycles)	R_3 (in ohms)	Correction (1000—)
.8	1000	0	-1.5 db
1.0	800	120,000	-4 db
1.6	500	200,000	-8 db
2.0	400	300,000	-12 db
2.7	300	500,000	-16 db
		800,000	-20 db

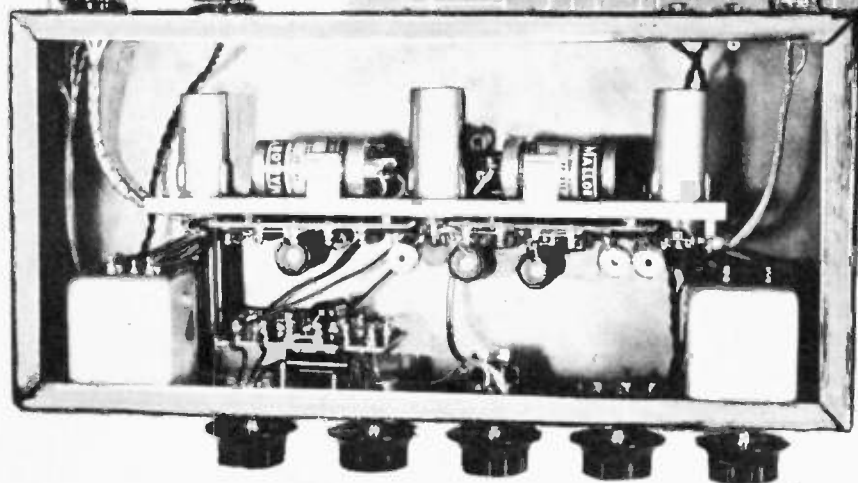
Table 3. How the control can be calibrated using an ohmmeter exclusively.

Fig. 6. Response curves obtained with the constant-amplitude equalizer.



EQUALIZER-PREAMP

By ARTHUR J. ROSE



Over-all view of complete preamplifier showing sub-assembly construction. The bulk of the wiring is completed outside of chassis. Panel controls (left to right) are: low-pass scratch filter, treble, bass, volume-power, program selector, and equalizer.

MUCH of the information included in this article is presented in answer to the many inquiries submitted by those who read the author's earlier article¹ and towards fulfillment of a need for a universally applicable equalizer-preamplifier.

A summarization of the features deserving inclusion in the final design provides a basis for circuit development. Diversified opinions concerning certain auxiliary appliances are manifest among professionals and laymen alike. Therefore, it is not without some degree of subjectiveness that selections are given. However, it is felt that the majority of readers will regard them as the minimum necessary for consistency with the highest grade of equipment used to form the complete audio system. The design "check list" includes:

- (a) Provision for incorporating and matching phono cartridge, radio, television, and tape inputs
- (b) Control of over-all amplification
- (c) Easily switched recording playback equalization
- (d) Continuously variable bass compensation
- (e) Continuously variable treble compensation
- (f) Low-frequency rumble filter
- (g) High-frequency noise filter
- (h) Low impedance output

Along with desirable inclusions, other effects are subject to consideration. Instability, superfluous noises, and distortion are equally important as undesirables. Readers of the previous article will recall the blanket solution to the hum and noise problem. Wirewound resistors and a direct current heater supply are applicable

Design details on a modern control unit engineered to provide flexibility and high quality performance.

in any design. If it were not for reasons to follow shortly, d.c. would be recommended for all applications. As will be appreciated by many, the final circuit permits a.c. operation in many cases. Distortion is maintained at a very low figure by proper tube operation and generous amounts of feedback. It should be noted that every stage, with the exception of the first, uses feedback.

Attention is directed to the schematic of Fig. 1. Preliminary inspection will reveal its sectionalization into a magnetic pickup preamplifier feeding a two-stage feedback amplifier. Tone controls follow this section into a cathode follower. In addition, there is the sharp cut-off low-pass filter that was described in the earlier article. More detailed study shows a six-position equalizer, rumble filter, and the method of using the cathode follower as a buffer between the low-pass filter and the balance of the circuit. When the filter is in the "flat" position, output is taken directly from the cathode follower. Otherwise, the output is taken from the filter where there is no need for a very low impedance output because of the deliberate attenuation of the higher frequencies. The low impedance source of the feedback amplifier and the high impedance loading of the cathode follower form an ideal condition for the tone controls. A direct approach to each section will greatly facilitate an

understanding of the complete unit.

Playback Equalizer Section

There is no question in the writer's mind that feedback type equalizers are much to be preferred over "between the stages" types that offer no outstanding advantage such as reduction of distortion. As low as 20 cps, feedback types have distortion reduction factors great enough to be readily appreciated. Distortion at higher frequencies defies measurement; however, most feedback types are necessarily of the breed that introduces noise because of unbypassed cathodes. The equalizer stage to be presented does not have this difficulty nor does it have another drawback, to be mentioned shortly, that is common to the other types—even those with grounded cathodes. It is no more troublesome to construct although it requires more components.

It has come to be accepted that recordings need to be *exactly* equalized if the maximum of realistic reproduction is to be obtained. Because acoustic systems are subject to such wide variations, all equipment leading to the amplifier output terminals must be of unquestionable performance. This permits adjustment of the acoustic system. With properly equalized recordings, subsequent frequency-selective tone controls can be used effectively in co-ordinating differences in loudness levels and playback *vs* re-

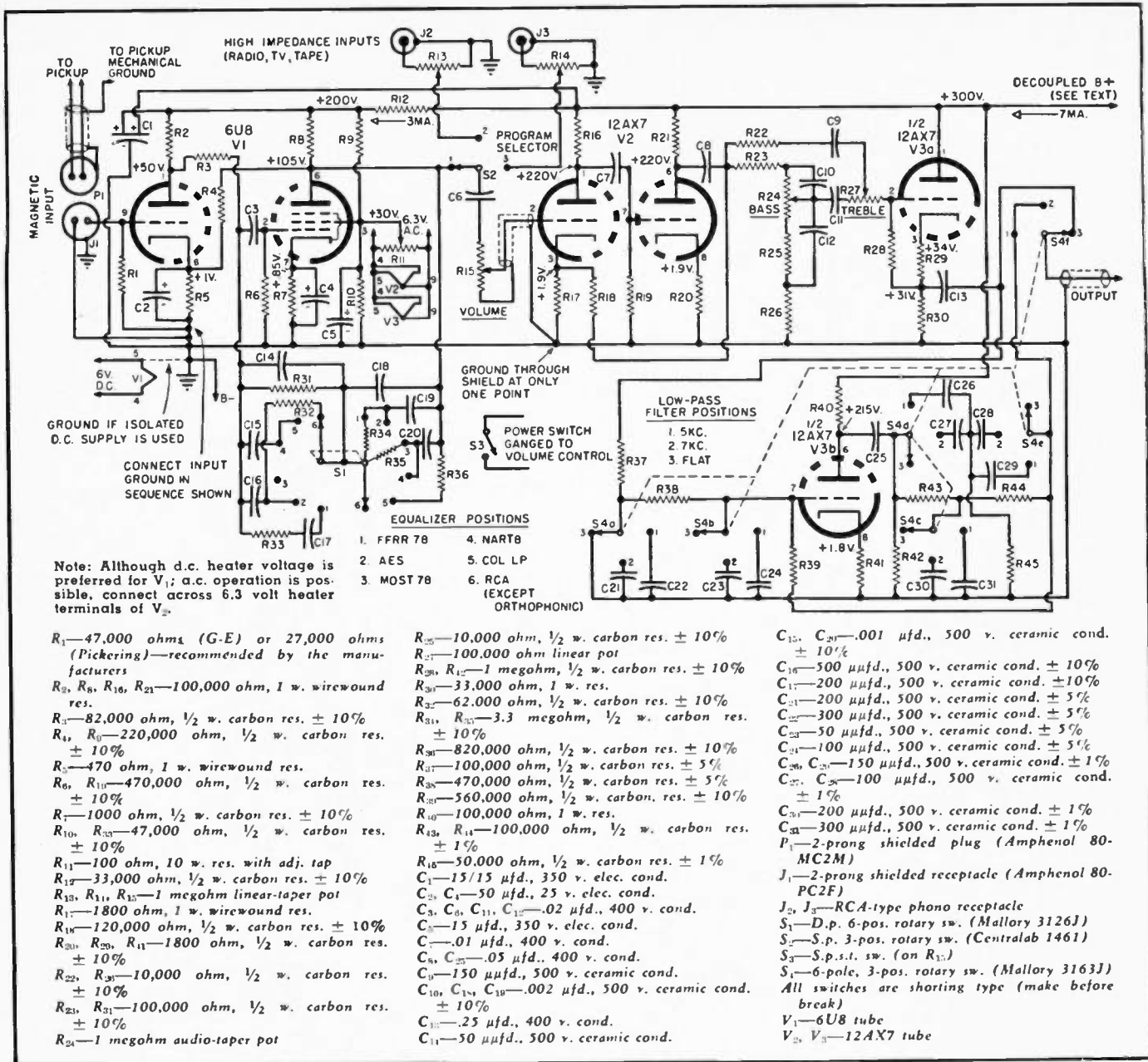


Fig. 1. Complete schematic diagram of equalizer-preamplifier. Precision parts are used throughout to insure quality performance.

ording acoustics. Lack of bass, especially at low levels, is sometimes the fault of the speaker or its acoustic environment. Often, however, it is due to insufficient bass equalization in the recording equalizer stage, in which circumstance even full use of the bass tone control will not prove to be enough.

Familiar twin-triode feedback equalizers are incapable of the 6 db-per-octave rate necessary in many playback requirements for proper bass equalization. This may come as a surprise to many because of the popularity of the various arrangements using triodes. The main reason for this is insufficient loop gain. Under ideal conditions, a gain of 2500 or so can be expected from cascaded 12AX7 sections and about 900 for a 12AY7. This in itself is poor. Aggravation of the situation appears with an unbypassed first cathode resistor that cuts these values by more than 50%. Since this

situation has not been well publicized, it is desirable to present a brief supporting analysis.

An important factor in the maximum available rate of boost from any feedback equalizer is the ratio of midband loop gain (gain without feedback) to the over-all midband gain (gain with feedback). Examination of Fig. 2 will show the response that can be expected for different values of the ratio, A_0/G_0 . These curves are based on stages with sufficient coupling and bypassing well below 10 cps. The 12AX7 example with the commonly used over-all midband gain of 50 can, at best, produce a curve somewhere between the ones shown for ratios of 20 and 40. In like fashion, the 12AY7 will produce the 20 curve. Because the feedback can be considered almost purely reactive (at lower frequencies) and therefore double in value every octave is no cause for a 6 db-per-octave rate here. In fact, for any cir-

cuit to produce this rate, an infinite ratio is required as the equation in Fig. 2 shows. At 50 cps, with a 500 cps turnover, 20 db boost ($C = 10$) corresponds to 6 db-per-octave from the turnover. Substitution of these values in the equation will cause the denominator to go to zero with a subsequent impossible gain requirement.

With this in mind, several alternatives are possible. Ratios in the neighborhood of 80:1 can be obtained by dropping the over-all gain with more feedback or by raising the loop gain by the use of pentodes. To drop the over-all gain with triodes would entail using feedback components that would load the stage to excess and would require condensers of physical proportions too staggering for space requirements. Chances are that the requisite G_0 would be too small for application.

A compromise using a high gain pentode with a sufficient A_0/G_0 ratio and an unfeedback triode first stage

gives proper compensation and permits bypassing of the first cathode—a threefold advantage. Bypassing greatly lowers first stage noise, permits a.c. heater operation and gives higher gain. Attenuation due to the feedback isolation resistor requires that the triode precede the pentode for maximum signal-to-noise ratio.

"Natural" for this application is the 6U8 triode-pentode. Designed primarily for r.f. use, 6U8's are newcomers to audio. They have microphonic tendencies, but no more so than other types used often in similar circuits. Some selection at the parts dealer will help: hold the tube by the pins and tap it with a fingernail while listening for various *bing*s and *bong*s. After about ten hours use, many of these tubes will become much quieter. Their thermal noise naturally varies from tube to tube, but has been found to be quite low especially with the circuit shown. 6AU6's with extremely low hum and noise levels can be substituted where one envelope is not deemed advantageous or where higher gain is desired. This will be further discussed with the next section.

An over-all midband gain of 25 to 30 can be expected from the first section. The pentode has a loop gain of 185 and is reduced by feedback to about 1.25. The triode stage has a gain of 20.

Equalization is provided for what the writer believes to be the majority of the recordings in use. Rather than use a separate network for each position, the switching uses a minimum of parts to achieve the correct values. For those with special requirements, a table is provided in Fig. 3 that will enable quick selection of components. These have been derived by network transformations of values generally given for "between the stages" equalizers.²

Wirewound resistors are specified in all places where their use contributes to lowered noise. Bypassing of R_5 is effective only above 20 cps, therefore this resistor must be wirewound to prevent low frequency noise. C_{11} has been added to prevent high frequency instability. It has no attenuating effect in the audible region.

A single resistor, R_1 , does a lot of work in this circuit. The addition of this component completes the rumble-filter scheme. First, coupling constants throughout the unit were chosen to give roll-off below 20 cps. R_2 and C_2 are similarly chosen. Feedback from the pentode stage via R_1 is operative only when the impedance of the triode cathode rises—at very low frequencies. The combined effect is to give a fairly rapid attenuation below 20 cps without altering response above that frequency. Experiments with parallel-T networks have shown that while the attenuation rate is greater, the improvement does not justify the added complication.

Second Section

Input requirements of the various

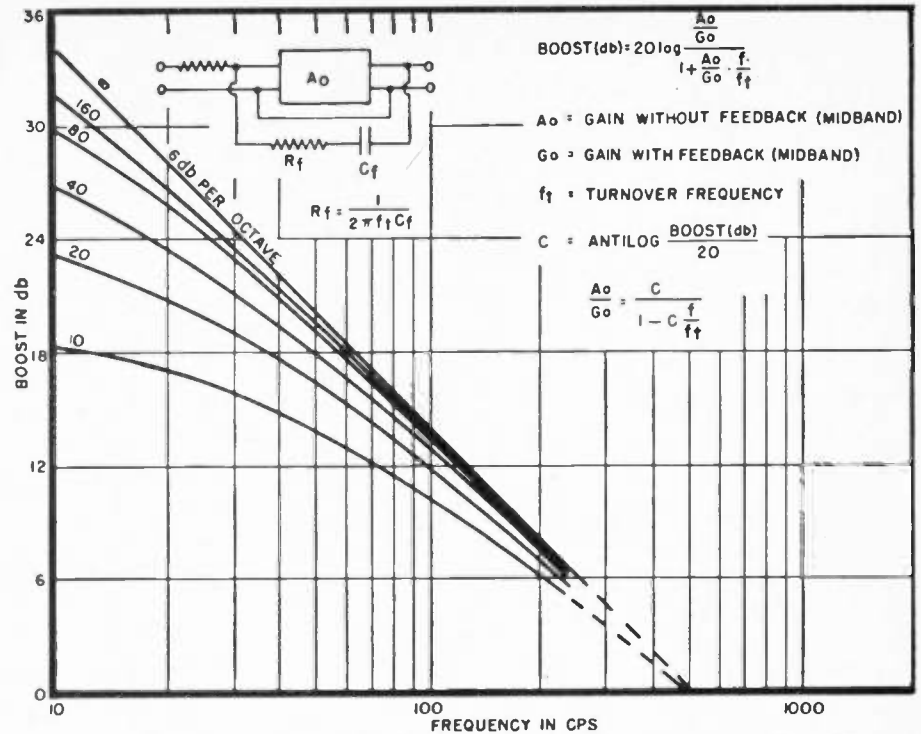


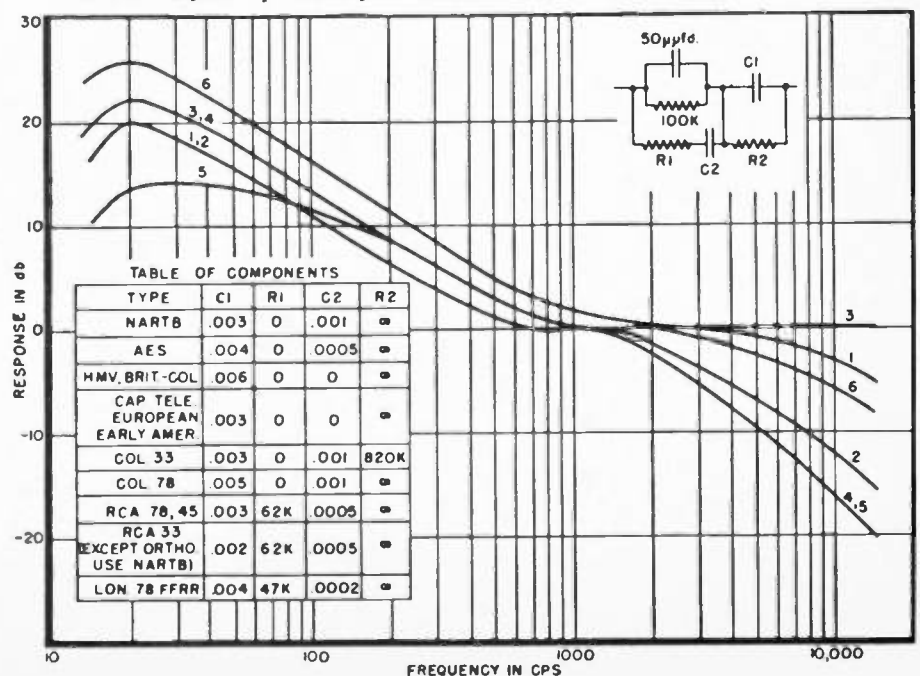
Fig. 2. Rate of bass boost of feedback equalizers depends largely upon the ratio of midband gain with and without feedback. Text points out shortcomings of many equalizers using degeneration and offers basis for development of unique circuit.

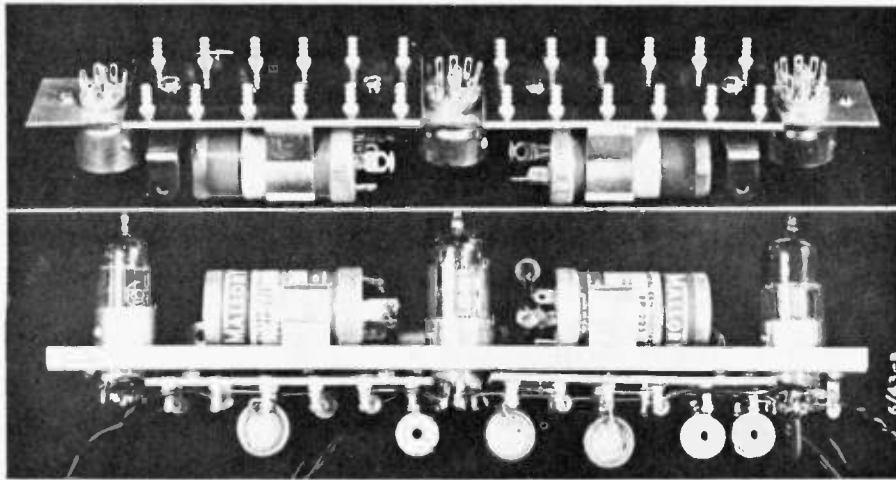
final amplifiers vary from 0.5 to 1.5 volts. The output of the preamplifier is largely determined by the second section.

Again feedback has been called upon to provide amplification that is distortionless and virtually independent of tube characteristic or supply voltage variations. To ensure this, the loop gain must be high enough to have A_0/G_0 large compared to one. The feedback must not, however, be great enough to excite oscillation and some caution is advised. As shown, the gain is nearly 67 and is determined

almost solely by R_{16}/R_{17} . For lower gain, R_{16} has to be lowered and a 12AU7 should be used. For higher gain, bypassing of R_{20} may be required in addition to increasing the value of R_{16} . If higher gain is contemplated, it should be borne in mind that tube noise from this section, as well from the first section, will increase proportionately. 6AU6's will have to be used in the first section with a d.c. heater supply. The d.c. is recommended there, as shown in Fig. 1, because the gain is high by usual preamplifier standards. Lower gains will suffice with cartridge-

Fig. 3. Response of equalizer positions shown in Fig. 1. Special arrangements can be designed by referring to the basic circuit and table of components.





Two views of the sub-assembly at different stages of its construction. See text.

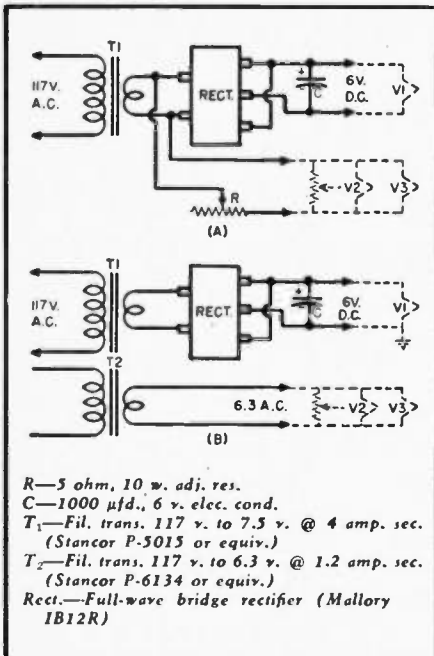
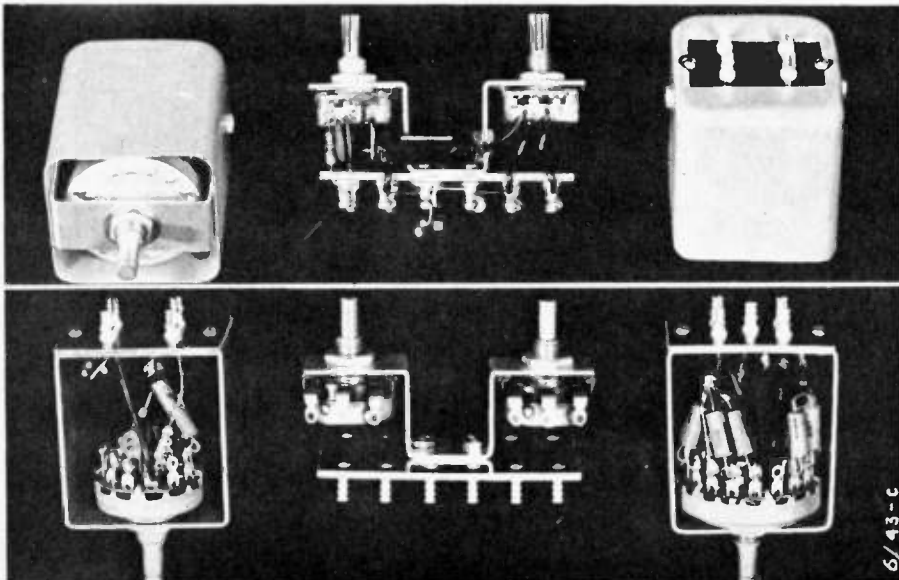


Fig. 4. Two easily-built d.c. heater supplies designed to be used with the first stages of the high-gain preamplifier unit.

Circuits associated with equalizer, tone controls, and low-pass scratch filter are wired to control assemblies. Protective shield cans are not too essential.



provided the source is properly balanced. Usually it is the predominance of a maladjusted tweeter that is to blame when highs cannot be brought under control with a treble control of this type. Known as variable "degree" as opposed to variable "slope," its effect is to lift the middle as well as the top highs, thereby giving a more balanced action. Boost and droop are more applicable to the bass end with a variable slope control. This is to be preferred in that region.

Output from the cathode follower is about 0.85 times its input. The output impedance is nearly 500 ohms. Because of the elevated cathode, heater returns should be made to a similar potential. The pentode screen grid is an ideal place.

Low-Pass Filter

Comment about this stage is limited to information not mentioned in the earlier article. R_{10} and R_{11} have been chosen to bring the gain to unity so there will be no change in level when the filter is switched from flat to cut-off.

Bypassing R_{11} is not recommended as it produces a peak in the response. The loop gain has been increased by the use of a 12AX7 section instead of a 12AU7 and produces an optimum cut-off characteristic for the given component tolerances with a gain of about 25.

Experimenters who wish to investigate the possibilities of a variable slope filter can try variations of the loop gain by placing a potentiometer in the loop or by bridging the T-network with a variable resistor, etc. A wide variation of slope is possible, but full variation from flat to extreme cut-off would seem difficult to obtain. At present, sufficient data is not available to quote definite capabilities.

Power Supply

Duplication of the voltages shown on the schematic is probable within 10% with a supply of 300 volts. Voltages as low as 250 volts can be successfully used. In any event, the supply must be isolated by a resistance greater than 10,000 ohms if the main amplifier is to furnish the plate power. Ideally, a well decoupled point should be chosen in the final amplifier that will supply at least 370 volts with the additional 7 ma. loading of the pre-amplifier.

A separate power supply need only furnish 250 to 300 volts and in the interest of flexibility and sure-fire simplicity this is to be recommended. Self-powered preamplifiers are coming more into their own because of the desire of most experimenters to try their units with many different final amplifiers.

Heaters must be supplied with a separate winding and preferably a separate transformer. Conditions for a.c. or d.c. operation of the playback-equalizer (first) section heaters have been stated. Since the 6U8 draws 450 ma., a dry disc rectifier supply with a

7.5 volt transformer, as shown in Fig. 4A is most applicable when d.c. is required. The remaining heaters should not be operated from the d.c. side of the supply as the ripple would substantially increase. An alternative, shown in Fig. 4B, may appeal to some experimenters. 12AU6's can be series operated from a 150 ma. source if this

type is used instead of the 6U8. An infrequently used method of construction is shown in the accompanying illustrations. This method greatly facilitates wiring and maintenance as well as ensuring a professional appearance. Along with many others, the writer experiences as much, if not greater, satisfaction from the building

process as from the performance of the completed unit. Even so, wiring does become tedious and much thought was given to correcting some of the wiring and parts placement difficulties with a mechanical arrangement that permits simpler and more positive techniques. The sub-assembly method permits the bulk of the parts mounting and wiring to be negotiated in the open with a freedom uncommon to point-to-point chassis wiring.

The writer wishes to thank the many persons who have expressed their interest in the preamplifiers heretofore described and who have offered suggestions for their improvement.

Special gratitude is offered to Mr. Uolevi Lahti of Ann Arbor, Michigan, whose unusual interest and activity in the high fidelity sound reproduction field has made possible the many listening tests and inspiration resulting in the unit described in this article.

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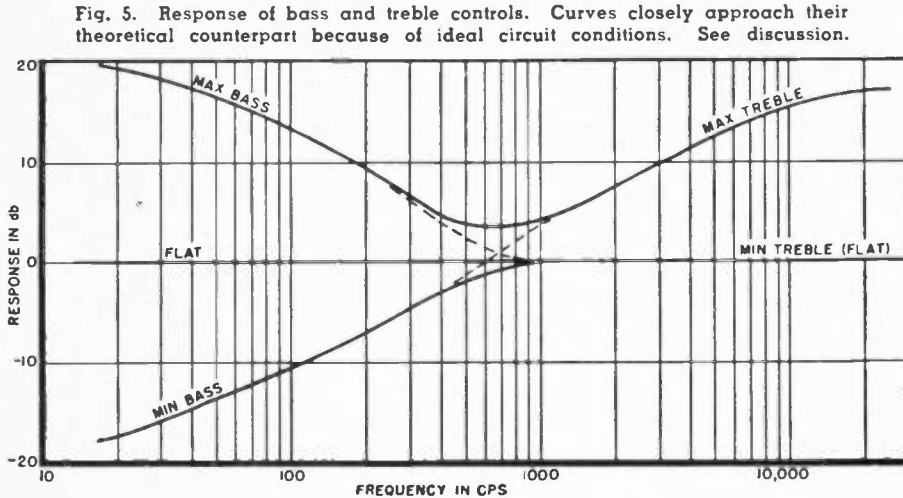


Fig. 5. Response of bass and treble controls. Curves closely approach their theoretical counterpart because of ideal circuit conditions. See discussion.

EQUALIZER-PREAMP MODIFICATIONS

CORRESPONDENCE from readers has indicated the success of the Equalizer-Preamp described in June 1953 issue of RADIO & TELEVISION NEWS. Numerous recent requests for modification of the equalizer to include the RIAA¹ curve are answered in the following. Other changes in the circuit have not been necessary nor desirable.

Since the RIAA and the Col LP curves both derive their bass sections from the shunting of a resistor across the feedback capacitor, thumps arise during switching due to changing d.c. levels unless an additional blocking capacitor and a high resistance charging resistor are employed. Not shown in the original article, these extra components are given in the diagram of Fig. 1.

C_{18} is dictated by the maximum turnover frequency to be used. If, for example, the maximum turnover is to be 800 cps, then C_{18} has to be .002 μ fd. Switch positions for 800 cps require no additional capacitance, but an additional .001 μ fd. is needed for 500 cps and .002 μ fd. for 400 cps. The original equalizer switch used such an arrangement with the exception of the Col LP position. There, the shunting resistor effectively produced the correct curve in spite of the high turnover and thereby saved the expense of an additional .001 μ fd. capacitor.

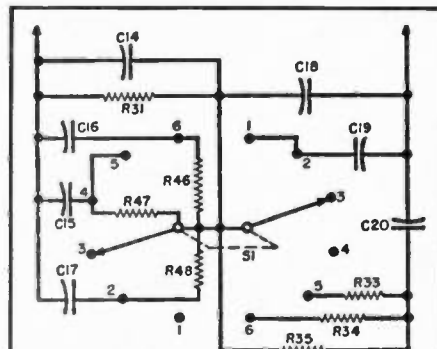
Because the modification replaces the old RCA curve, the 800 cps turnover is no longer needed and C_{15} is increased to the correct value for 500 cps turnovers. The Col LP position uses a different shunting resistance than originally shown for reasons previously inferred.

C_{20} and R_{33} prevent switching thumps. Additional 4.7 megohm charging resistors are employed in the de-emphasis side of the switch to insure completely noiseless switching.

Readers are again cautioned to insure

correctness of capacitor values before using them in equalizers and other frequency correcting networks. Ceramic disc capacitor tolerances are catalogued as $-0 +100\%$. Small mica capacitors should therefore be used where bridging facilities are unavailable.

Fig. 1. Circuit changes to be incorporated in the equalizer-preamp of June, 1953 to provide equalization for new RIAA curve.



- R_{32} —100,000 ohms (no change)
- R_{33} —510,000 ohms (was 47,000)
- R_{31} —1 megohm (was 3.3 megohm)
- R_{35} —4.7 megohm (was 3.3 megohm)
- R_{15}, R_{17}, R_{18} —4.7 megohm, $\frac{1}{2}$ w. res. (not in original)
- C_{15} —50 μ fd. (no change)
- C_{16} —0.01 μ fd. (no change)
- C_{17} —750 μ fd. (was 500 μ fd.)
- C_{18} —600 μ fd. (was 200 μ fd.)
- C_{19} —0.033 μ fd. (was .002 μ fd.)
- C_{20} —0.01 μ fd. (was .002 μ fd.)
- C_{50} —.02 μ fd., 400 v. (was .002 μ fd., 500 v. ceramic)
- S_1 —D.p. 6-pos. switch

POS. CURVE

- 1—Eur., Early Amer. 78's, London frr 78
- 2—AES (pre-RIAA)
- 3—Amer. 78's, old RCA 33 $\frac{1}{2}$'s
- 4—NARTB (pre-RIAA)
- 5—Col. LP's
- 6—RIAA, RCA "Ortha", new NARTB, new AES

¹"The Curve That Conforms", Radio & Television News, July 1954

A BASS-BOOST CONTROL

By **ALLAN M. FERRES**

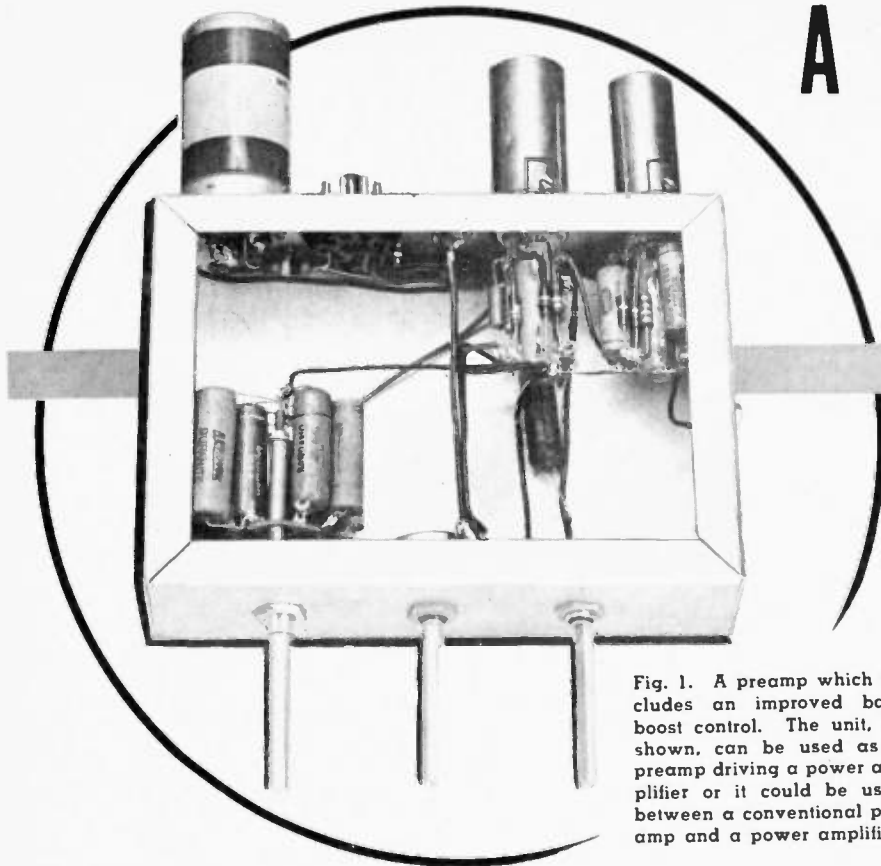


Fig. 1. A preamp which includes an improved bass boost control. The unit, as shown, can be used as a preamp driving a power amplifier or it could be used between a conventional preamp and a power amplifier.

Details on a separate control unit designed to provide a maximum of ten transition frequencies.

TOO many bass-boost controls muffle music and speech. It is a simple matter to change such a control to a better one, or to build a new amplifier using an improved circuit.

The trouble with the usual control is easily explained. Fig. 2 shows the conventional bass-boost circuit and its response curves. The frequency at which the response is increased 3 db is called the "transition frequency" and is that frequency at which the reactance of the condenser is equal to R_1 . As neither the condenser nor R_1 is adjustable, this frequency remains constant.

The total boost available depends upon the value of R_1 , R_2 , and R_3 . As R_1

is the only adjustable element, then a flattening or "shelving" of the response curve occurs when the bass-boost control (R_1) is set at less than its maximum-boost position. (Shelving also occurs at very low frequencies at maximum boost, but this need not be considered here.) Curve A shows the response at full boost, and curve B shows how the curve flattens out at about one-half boost.

This circuit is widely used due to its simplicity, but has two drawbacks which should be given consideration.

The first is that at moderate listening levels, at which, let us say, 8 db of boost at 100 cycles sounds about right, 3.5 db of boost is also obtained at 600

cycles, as shown in curve B. This tends to make a speaking voice sound muffled and upsets the tonal structure of the music produced by instruments whose fundamental tones lie in the mid-frequency region. This causes the type of sound referred to as "muddy" or "indistinct" by laymen.

The second objection is this. Of necessity, the bass-boost control is used in most systems to compensate for a falling-off of the over-all low-frequency response due to deficiencies of the speaker, baffle, or room acoustics. As these deficiencies usually exist at fairly low frequencies only, the boost produced at the mid-frequencies by the conventional control tends to emphasize speaker, baffle, and room resonances and make the system sound much less satisfactory than it should.

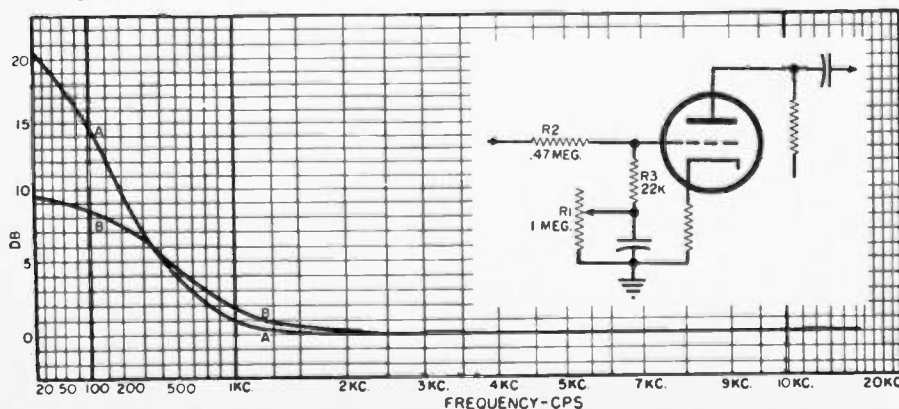
A more satisfactory control would be one which provided a continually-variable transition frequency. With such a control, the extreme low frequencies could be boosted as desired without disturbing the flatness of the response curve in the mid-frequency region.

The control and associated amplifier to be described here does not provide a means of compensation for all low-frequency-response troubles, but it does go a long way toward providing a better control of the bass response of a home radio/phonograph system. This control and amplifier was designed to provide a wide choice of transition frequencies as simply as possible.

As it has been found in practice that a choice of seven or eight transition frequencies is as satisfactory as a continually variable frequency, and as it results in certain circuit simplifications, a tap-switch type of control is used. In this case, a choice of ten frequencies is provided, but the number can be reduced if desired.

As mentioned previously, the transition frequency depends upon the values of the condenser and resistor, R_1 , of Fig. 2. Varying either one would shift the frequency. In the improved control, the condenser is varied. At first glance, it would seem easier to

Fig. 2. A conventional control circuit and the resultant response curves. See text.



vary R_1 , but as the amount of boost is also dependent upon R_2 , and as there is a fairly limited range over which R_2 can be varied, the capacity variation is the better method.

The chart, Table 1, lists the condensers and the corresponding transition frequencies used by the author. If the reader wishes to experiment with other transition frequencies, the proper value of capacity in microfarads can be found by dividing 7.25 by the frequency desired. Similarly, with a given condenser, the resultant transition frequency in cycles is equal to 7.25 divided by the capacity of the condenser, in microfarads.

The transition frequency is that at which the reactance of the condenser equals R_1 . The reactance of the condenser, X_c , equals $1/2\pi fC$, where f is in cycles and C is in farads. Transposing, C equals $1/2\pi X_c f$. Substituting 22,000 ohms for X_c , and multiplying by 10^6 , C equals $100/13.8f$. Solving the equation produces C equal to $7.25/f$.

Similarly, the frequency in cycles equals $7.25/C$, C being in microfarads.

It should be noted that ordinary paper condensers can be expected to be as much as 20% off from their rated value. This will affect the response curves obtained in the completed amplifier, as will be seen from the curves of Fig. 4. However, it has been found satisfactory to select condensers from the chart or by formula. The reason is that the exact frequency itself is not too important as long as there is a wide enough choice of frequencies available. After the amplifier is completed and tests indicate that one or two curves need correction, then a change can be made in the responsible condenser. If the transition frequency is too low, a condenser of lower capacity is substituted and if too high, a larger condenser can be used.

R_2 should be selected by testing with an accurate ohmmeter to be sure that it is as close as possible to 22,000 ohms. If it is not close to this value, all of the transition frequencies will be off even though the condensers used are accurate. The regular commercial tolerances are satisfactory for the other resistors and condensers in the circuit as they have little effect upon the over-all frequency response.

In order to provide an adjustable treble boost, R_1 and C_{11} are included in the circuit of Fig. 3 and a 500,000-ohm pot is used for R_1 . When the arm of R_1 is rotated to the end of the pot that is connected to R_2 and R_1 , its counterclockwise position, C_{11} is effectively out of the circuit. This is the setting for flat high-frequency response. When the arm is rotated to the end connected to R_2 , the higher frequencies are bypassed around the attenuating network R_2 and R_1 and fed directly to the grid of the tube. This produces the maximum treble boost. As shown in the curves of Fig. 4, the high transition frequency is about 3000 cycles at full treble boost. This transition frequency can be

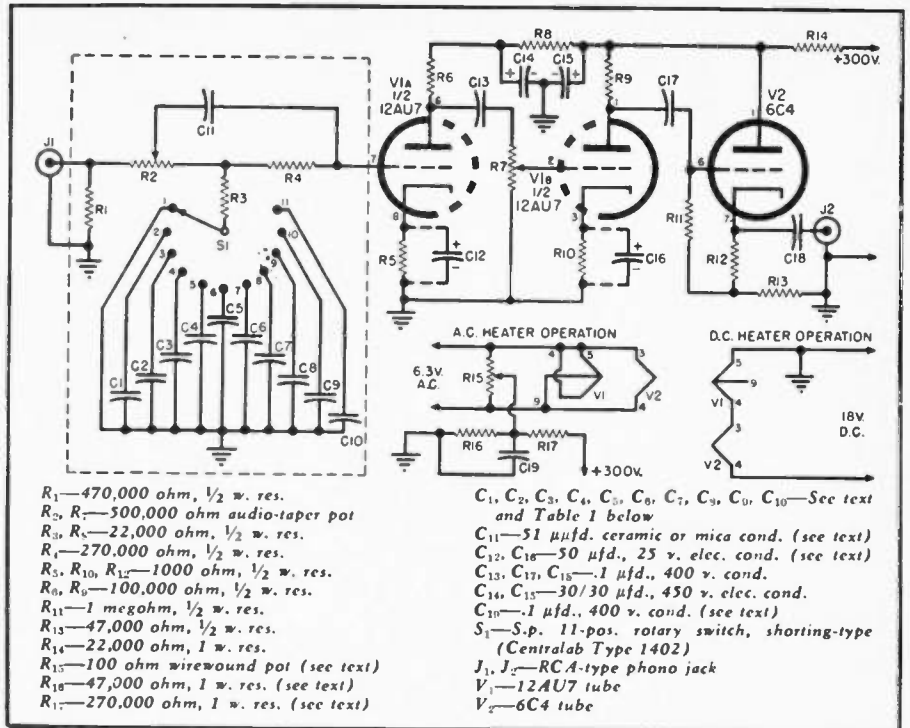


Fig. 3. Schematic of control and amplifier. Section in dotted portion can be added to an existing amplifier, if desired, or built in conjunction with amplifier diagrammed.

raised, if desired, by using a smaller condenser in place of the 51 μ fd. specified for C_{11} , or lowered by using a larger value.

In order to provide a good signal-to-noise ratio with a minimum of distortion, the tone control circuits are connected to the grid of the first tube of the amplifier and the gain control is placed between the first and second stages. As the input impedance of the amplifier is not less than about 150,000 ohms at any frequency, irrespective of the tone control settings, it may be fed from either a triode amplifier or cathode-follower stage of a tuner or phono preamp.

The input signal level should not be greater than about 1 volt. If the tuner or phono preamp used furnishes an output greater than this, a gain control should be used on the input source to reduce its output to the proper value. For minimum distortion, the input level should not be greater than necessary to provide sufficient output to drive the power amplifier used in

SW. POS.	CAPACITY (μ fd.)	TRANSITION FREQ.
1	Short	Flat
2	.2	36
3	.15	48
4	.1	73
5	.068	106
6	.05	145
7	.035*	207
8	.022	330
9	.018	400
10	.015	480
11	.01	725

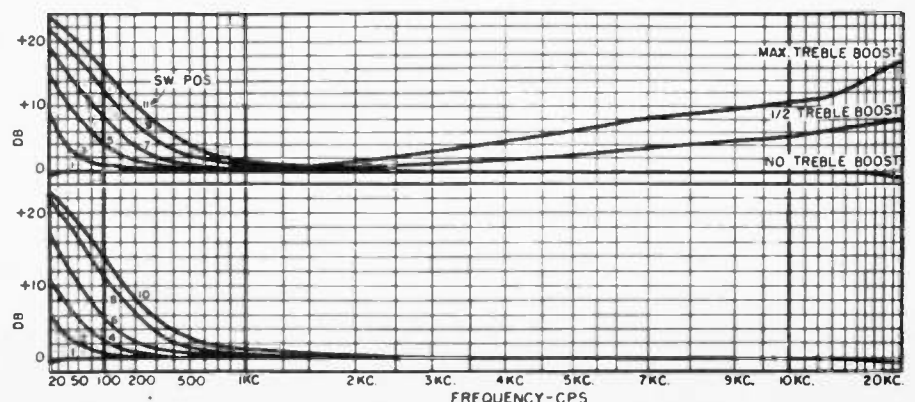
*.035 μ fd. made up of .03 μ fd. and .005 μ fd. in parallel.

Table 1. Values of C_1 through C_{10} and the resultant transition frequencies. The switch positions correspond to those shown in Fig. 3 and the resulting curves shown in Fig. 4.

the system to full volume. As the over-all gain of this amplifier is about 12 db, a 1-volt input will furnish 4 volts output.

The use of a cathode follower in the output stage provides an output impedance of about 650 ohms so that a hundred feet or so of shielded cable can be connected between this amplifier and the power amplifier of the system without appreciable loss of

Fig. 4. The response curves of the improved bass-boost control. See Table 1.



high-frequency response. If this amplifier is to be mounted within one or two feet of the power amplifier, the cathode follower, V_2 , can be omitted from the circuit. The output could then be taken from C_{15} , resulting in fewer parts and a smaller amplifier.

The bass-boost control consists of a *Centralab* type 1402, which is an 11-position, single-section, shorting-type switch, small enough to fit into a two-inch-deep chassis. A shorting-type switch is necessary in order to prevent clicks when the control is turned. The original screws were removed from the switch and two 6-32 round-head machine screws, two inches long, substituted for them. Two one-inch spacers were used to mount a *Cinch-Jones* type 2004 4-point terminal strip on the rear of the switch. The condensers were then mounted between the switch terminals and the terminal strip which is connected to ground.

This bass and treble boost control can be added to the input of an existing amplifier if desired, instead of building the complete amplifier shown here. The parts included within the dotted lines should be wired into the amplifier in the same manner as shown in the circuit of Fig. 3. If the treble boost control is not required, R_2 can be changed to a 470,000-ohm, $\frac{1}{2}$ watt

resistor and R_1 and C_{11} eliminated from the circuit. The grid of the tube in that case, should be connected to the junction of R_2 and R_1 .

As the control introduces a loss of 23 db at 1000 cycles, this much additional voltage gain must be available if it is incorporated into an existing amplifier. If this gain is not available then it is advisable to build the complete amplifier shown in Fig. 3.

Fig. 1 shows the underside of the amplifier as built by the author. As a 5" x 7" x 2" chassis could be easily fitted into the cabinet used, there was no need to make the amplifier compact. The treble-boost control, R_2 , is mounted in the center of the front drop of the chassis, and the volume control, R_3 , is mounted to the right. The placement of the other parts can be readily determined from the photograph.

In the installation for which this amplifier was built, 18 volts of d.c. was available for heater supply, so the heaters of the 12AU7 and the 6C4 were connected in series. The use of d.c. made it possible to eliminate cathode bypass condensers C_{12} and C_{16} and to run unshielded leads in the chassis. The polarity of the heater supply is not important and either side can be connected to ground. If the

heaters are to be operated on a.c., it will be advisable to include C_{12} and C_{16} in the circuit and to shield the leads from the input jack, J_1 , to R_2 and those from R_2 to R_1 , in order to reduce hum to a minimum. The heaters, of course, should be connected in parallel.

R_{15} acts as a hum-balancing pot when a.c. is used. If the 6.3-volt winding of the transformer in the supply to be used with this amplifier is connected to ground or to a bias voltage, this connection must be removed to prevent R_{15} from being shorted out. R_{16} and R_{17} make up a voltage divider across the 300-volt supply to furnish a 45-volt positive bias on the heaters. C_{15} is a bypass condenser to keep the heaters at ground potential at audio frequencies.

A 5" x 7" bottom plate should be used to complete the shielding of the chassis.

The plate power requirements are 6 milliamperes at 300 volts. This can be obtained from the supply for the power amplifier and should be well filtered.

Although the conventional bass-boost control is simpler in design, this improved unit, by its performance, amply compensates the builder for the few additional parts required.

-30-

A FEEDBACK LOUDNESS CONTROL

A low output impedance as well as high- and low-frequency compensation characterize this compact control.

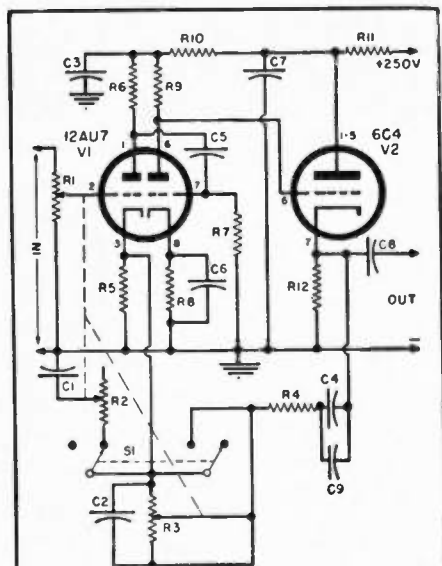
By RAY C. WILLIAMS

Biophysics Dept., Medical College of Va.

LOUDNESS controls which compensate for physiological characteristics of the human ear, as shown by the Fletcher-Munson curves, are considered a "must" by most high-fidelity enthusiasts.

Such a control should have the following features: simplicity, compensation for both high and low frequencies, constant mid-frequency gain when compensation is switched in or out, continuous variability, and low output impedance. A loudness control which incorporates all of these features is shown in Fig. 1.

The basic circuit consists of a two-stage resistance-coupled amplifier with cathode-follower output. A feedback loop consisting of C_2 , R_3 , R_1 , and C_{11} , is connected between the cathode-follower output and the cathode of the input stage. C_1 is a 20 μ f., 150 v. electrolytic shunted by a .01 μ f. ceramic condenser and is used for d.c. blocking only. C_2 - R_3 give low-frequency compensation and when shunted by C_1 - R_2 compensates for the high frequencies. The control element is a three-gang



- R_1 *—500,000 ohm linear-taper pot
- R_2 *—3000 ohm linear-taper pot
- R_3 *—250,000 ohm audio-taper pot
- R_4 —3300 ohm, $\frac{1}{2}$ w. res.
- R_5 —1000 ohm, $\frac{1}{2}$ w. wirewound res.
- R_6 , R_7 —47,000 ohm, $\frac{1}{2}$ w. res.
- R_8 —470,000 ohm, $\frac{1}{2}$ w. res.
- R_9 —1000 ohm, $\frac{1}{2}$ w. res.
- R_{10} , R_{11} —4700 ohm, $\frac{1}{2}$ w. res.
- R_{12} —27,000 ohm, $\frac{1}{2}$ w. res.
- C_1 , C_2 —.05 μ f., 400 v. cond.
- C_3 , C_4 —20 μ f., 450 v. elec. cond.
- C_5 —20 μ f., 150 v. elec. cond.
- C_6 —.1 μ f., 600 v. cond.
- C_7 —50 μ f., 25 v. elec. cond.
- C_8 —.25 μ f., 400 v. cond.
- C_9 —.01 μ f. ceramic cond. (shunt for C_1)
- S_1 —D.p.d.t. switch
- V_1 —12AU7 tube
- V_2 —6C4 tube

* All three pots ganged into a single unit. See text.

Fig. 1. Diagram of feedback loudness control.

potentiometer and is made of the IRC type "Q" and multisections having the values indicated in the parts list.

When the control element is at the maximum clockwise position, C_2 is shorted by R_2 making the feedback loop resistive. Shunt condenser C_1 is in series with R_2 and has little or no effect, the result being a two-stage amplifier having a flat frequency response. When the control element is turned counterclockwise, the reactance of C_2 becomes increasingly large at frequencies downward from 1500 cycles. C_1 becomes an increasingly effective shunt at frequencies upward from 1500 cycles. The reduction of the feedback fraction at the low and high frequency ends, with the mid-frequency fraction remaining constant, results in a marked increase in gain at the low and high frequency ends of the spectrum. Obviously, varying degrees of compensation will be obtained when the control element is turned between these two extremes. Switch S_1 simply shorts C_2 - R_2 and opens the circuit of the shunt condenser C_1 to make the feedback loop resistive, resulting in a flat amplifier.

In the maximum counterclockwise position the gain at 1500 cps is 14 db; at 30 cps is 38 db.; and at 15 kc. is 26 db. In the maximum clockwise position the gain from 30 to 15,000 cps is 14 db.

This loudness control can be incorporated into new or existing equipment or built as a separate unit. All grounds should be tied to a common point and the plate supply voltage should be well filtered.

-30-

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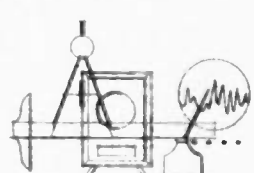
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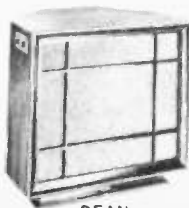


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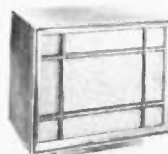
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Nothing before has been offered to equal it in the annals of the high fidelity market. The first complete audio amplifier incorporating compensated preamplifier, tone controls, power amplifier, and power supply all built upon a "printed circuit" plate. **LOOK AT THESE SPECIFICATIONS!** **POWER OUTPUT:** 12 watts, 24 watts peak at 1% harmonic distortion; 5 watts at 1% intermodulation distortion; 3 db power rating, 30 cycles to 20,000 cycles. **FREQUENCY RESPONSE** 20-20,000 cycles \pm 1 db. **INPUTS:** Radio, TV-Sound and Tape inputs: 1 megohm impedance, Phonograph inputs: 27,000 ohms for magnetic and dynamic cartridges. Built-in equalizer for use of ceramic, FM (Capacitance) and crystal pick-ups; adjustable equalization effective on all types of pick-ups. **SENSITIVITY** (Gain Factors): 6 millivolts at magnetic-phonograph input required for 12 watts output. 0.3 volts required at Tuner, Tape, TV inputs for 12 watts output. 42 db gain between magnetic-phonograph input and tape take-off jack. 10 db gain between Radio, Tape, TV inputs and tape take-off jack. **OUTPUT IMPEDANCES:** 4, 8 and 16 ohms. **DAMPING FACTOR:** 15. Source impedance much lower than internal resistance of loudspeaker. **FEEDBACK:** Four negative feedback loops result in virtual elimination of distortion; 20 db feedback around voltage and power amplifier stages. **QUIETNESS:** Noise plus hum 75 db below maximum output on Radio, TV, Tape input. 70 db below maximum output on phonograph input. (54 db below effective program level with 10 millivolts input.) **TAPE TAKE-OFF JACK:** Provides output of sufficient level to drive any tape recorder. Independent of volume control setting. **POWER OUTLET:** 110 volts for tuner, record player, etc., controlled by on-off switch. **CONTROLS:** SELECTOR (and TUNING control). 7 positions. Radio, TV, Tape and 4 phonograph positions for the following Turnover characteristics: AES, NARTB*, 800, LP. **ROLL-OFF:** 4 positions of high-frequency record compensation: LP (16), NARTB* (13.7), AES (12), RIAA. 78 (0). Figures in parentheses indicate db roll-off at 10,000 cycles. *New NARTB is identical with RIAA. **TA-Victor:** New Orthophonic, and new AES. **BASS:** Continuous control; 16 db boost to 16 db cut at 50 cycles; marked "flat" position. **TREBLE:** Continuous control; 12 db boost to 15 db cut at 10,000 cycles; marked "flat" position. **LOUDNESS (VOLUME):** With ON-OFF switch, compensated for loudness (Fletcher-Munson) curves. **TUBES:** Six in all doing the work of nine. No difficult-to-locate or special types used. 3-12AX7, 2-6AQ5, 1-6X4. **DIMENSIONS:** Only 4 1/4" high, 10 3/8" long and 8" deep from front panel to clearance over jacks on back panel. Shpg. Wt. 12 lbs.

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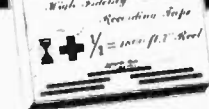
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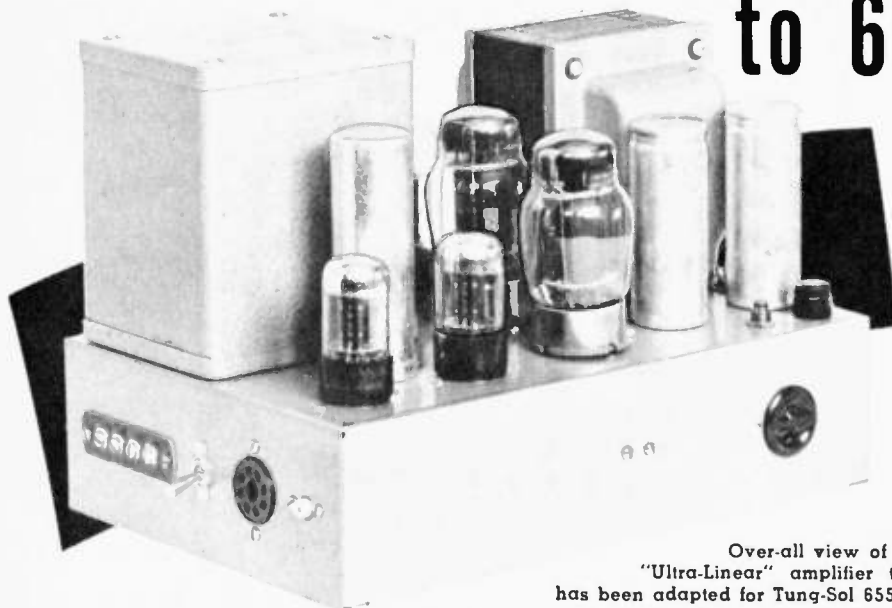
HI-FI ANNUAL & AUDIO HANDBOOK

CHAPTER 3

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Adapting the "Ultra-Linear" Williamson to 6550 Operation



Over-all view of the "Ultra-Linear" amplifier that has been adapted for Tung-Sol 6550's.

By **HERBERT I. KEROES**
Acro Products Company

A new high-power output tube, the 6550, will find many applications in converting present-day amplifiers and in new equipment. Up to 100 watts push-pull can be had.

SINCE its introduction many years ago, the *Williamson* amplifier has undergone a few design changes to further improve its performance. As originally described by Williamson, the amplifier was a 15-watt unit designed for low distortion, uniform output, and small phase shift over the entire audio range.¹ Since the original conception of the *Williamson* amplifier, American manufacturers have jumped on the bandwagon and today one will find many variations of the original circuit. Performance-wise there is wide variation among the different units made in this country. One of the circuit improvements made by American manufacturers came with the application of "Ultra-Linear" operation to the output tubes, a mode of operation which doubled output power and further reduced distortion.² This amplifier has been widely accepted by audiophiles with the result that there are about twenty commercial amplifiers on the market today which incorporate this design feature.

The application of "Ultra-Linear" operation to the *Williamson*-type amplifier increased the output power to 30 watts using the same type of output tubes operating at the same voltages. When this circuit was first introduced it was immediately noted that the new combination provided better sound, even at the low volume levels which the original amplifier could handle. This phenomenon has resulted in a new evaluation of the power requirements of an amplifier as a part of an audio

system and, in general, it has been observed that in amplifiers of analogous design, the unit of greatest capacity will sound best.

The attainment of high power in audio amplifiers has become relatively easy and inexpensive due to two factors, the increased efficiency of the "Ultra-Linear" output circuit and the introduction of new output tubes with greater power handling capabilities. One recently introduced tube, the *Tung-Sol* 6550, is particularly adaptable to output stages of the "Ultra-Linear" type and can be used to advantage in the "Ultra-Linear" *Williamson* circuit to provide an amplifier of 60-watt capacity having an intermodulation content at maximum output of 6/10th of one per-cent. This amplifier differs only in a small degree in dimensions and number of circuit elements from its predecessors, and many *Williamson*-type amplifiers can easily be modified to take advantage of the improved performance.

Amplifier Circuit

An examination of the circuit diagram reveals the basic *Williamson* circuitry of the first three stages. The first two, the input voltage amplifier and direct-coupled cathodyne phase inverter, are familiar and unchanged even with regard to tube type, the 6SN7. The driver stage also remains a 6SN7, with but one change. Individual cathode resistors have been added to provide a slight amount of local feedback in order to improve the

feedback phase characteristics and increase the stability margin of the amplifier.

The output stage is coupled to the driver through a resistance capacity network which provides conventional *RC* coupling at signal frequencies and an attenuated direct coupling at subsonic frequencies. This again introduces an improved low-frequency phase characteristic which adds to the stability margin of the amplifier. The use of this combined *RC* and direct coupling is made possible by the choice of fixed bias operation of the output tubes whereby the required negative bias is obtained from a separate bias supply. The fixed bias supply consists of T_3 , a 6.3 volt, 1 amp. filament transformer; a 50 ma. selenium rectifier SR_1 ; resistor R_{20} ; and electrolytic capacitors, C_{12} and C_{13} .

In order to reduce hum to a minimum in preamplifiers that are to be powered from the main amplifier, a positive bias has been applied to the heater line through resistors R_{27} , R_{28} , and capacitor, C_8 . If a separately powered preamplifier is to be used, this network can be eliminated, together with the hum balancing potentiometer R_{29} , and the centertap of the 6.3 volt winding on the power transformer T_2 can be grounded. "B plus" voltage for operation of the preamplifier can be taken either from point X or Y depending on the preamp to be used.

"Ultra-Linear" Output Stage

The "Ultra-Linear" type of output stage is characterized by output tubes of the tetrode type with the screens of the tubes connected to taps equally positioned about the centertap of the output transformer. The operation of the stage can most readily be understood by the following considerations: first, if the screen of an output tube is connected to the plate, the tube functions as a triode, and the plate characteristic curves are concave upward. Secondly, if the screen is connected to "B plus," the tube operates as a tetrode, and the plate characteristic curves are concave downward. If, however, the screen is connected to a tap on the primary of the output transformer, a type of operation is obtained midway between triode and tetrode. Depending upon the type of output tube used, the tap can be chosen to result in an almost linear set of plate

characteristic curves, and this mode of operation has been termed "Ultra-Linear." It has been determined experimentally that the best operating point for the 6550 is with the tap located at 40% of the primary turns.

"Ultra-Linear" operation of an output stage has sometimes been described as the application of negative feedback to the screen grids of the output tubes. If this concept is used to explain the operation, it should be noted that the feedback is of the power type rather than the more usual voltage or current feedback, and power is supplied to the screen grids over the operating cycle. It can be demonstrated mathematically that when power feedback is applied to the screen grid of a tube, the linearity of the plate characteristic curves can be improved over and above the amount normally to be expected by a consideration of voltage feedback only."

The output transformer is an *Acrosound* TO-330. This transformer is ideally suited for the 6550 tubes, providing the correct impedance match for maximum power and lowest distortion, and primary taps located at 40% of the total winding. The frequency response of the TO-330 is flat ± 1 db from 10 cps to over 100 kc., thereby providing the necessary low phase shift over the audio range for best feedback stability and faithful transient response. The halves of the primary winding are tightly coupled to make available a full 60 watts of output over the entire audio range 20 cps to 20 kc. Although the nominal rating of the transformer is 50 watts at 20 cps, no difficulty was experienced in obtaining full undistorted output at the low frequency extreme. Too much emphasis cannot be placed on the fact that an amplifier cannot be better than its output transformer irrespective of the circuit used. This component serves many functions in a feedback amplifier as well as providing an impedance match between the output stage and the speaker. However, all of the necessary conditions can be met by a propitious choice of design and this unit can, in fact, be improved in certain performance categories, for example, bandwidth over and above the circuit with which it is associated.

Feedback Stability

In a feedback amplifier it is always desirable to maintain a maximum amount of feedback stability in order to assure complete stability under all conditions of output power level and output load. The degree of stability of a feedback amplifier is generally rated in terms of stability margin, meaning the amount of additional feedback in db that can be added before the amplifier becomes unstable and oscillates. This design figure is usually taken under conditions of rated resistive output load. However, loudspeakers are not constant resistance devices, but present to the amplifier an impedance containing a large reactive component over a good portion of their operating range. Moreover, in the band outside

Power Output	60 watts @ 1000 cps; within ± 5 db of 1 kc. level @ 60 watts over range 20 cps to 30 kc.
Frequency Response	± 1 db @ 1 watt, 2 cps to 220 kc.
Intermodulation Distortion (60 and 3000 cps mixed 4:1, equiv. sine-wave power)	10 watts—.07%; 20 watts—.10%; 30 watts—.15%; 40 watts—.25%; 50 watts—.40%; 60 watts—.60%
Square-Wave Response	Rise time on 20 kc.—2 microseconds; overshoot on 20 kc.—none observed; ripple on 20 kc.—approx. 1%; droop on 20 cps—5%
Hum and Noise	80 db below rated output
Nominal Feedback	20 db
Feedback Stability Margin	10 db
Damping Factor	15
Sensitivity	1.3 volts r.m.s. for full output

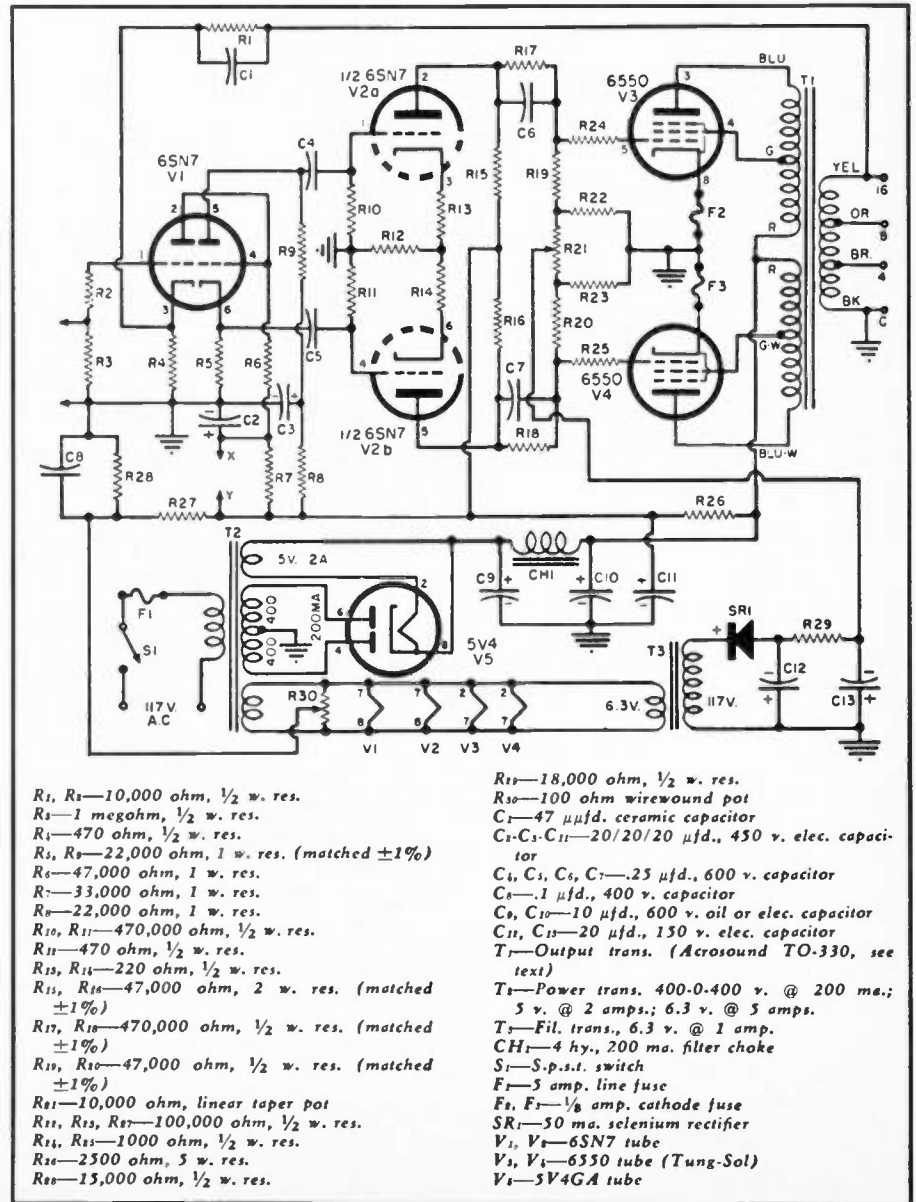
Table 1. Performance characteristics of the converted "Ultra-Linear" amp'ifier.

of the range of the speaker, the impedance may be almost completely reactive. It is desirable, therefore, to have a stability margin of 6 db or more to assure complete stability.

The stability characteristics of a feedback amplifier are associated with the bandwidth and phase shift characteristics of the amplifier circuit and output transformer, and there are several choices available to the designer

to increase the stability of a given amplifier. First, bandwidth may be traded for stability. In this procedure loss networks are added to shape the amplifier response curve so that the response of the amplifier falls off by the amount of feedback plus the stability margin before the phase of the feedback voltage becomes regenerative. Secondly, gain within the useful band may be traded for stability. In this

Complete schematic diagram of the 60-watt version of "Ultra-Linear" Williamson.



- R₁, R₂—10,000 ohm, 1/2 w. res.
- R₃—1 megohm, 1/2 w. res.
- R₄—470 ohm, 1/2 w. res.
- R₅, R₆—22,000 ohm, 1 w. res. (matched $\pm 1\%$)
- R₇—47,000 ohm, 1 w. res.
- R₈—33,000 ohm, 1 w. res.
- R₉—22,000 ohm, 1 w. res.
- R₁₀, R₁₁—470,000 ohm, 1/2 w. res.
- R₁₂—470 ohm, 1/2 w. res.
- R₁₃, R₁₄—220 ohm, 1/2 w. res.
- R₁₅, R₁₆—47,000 ohm, 2 w. res. (matched $\pm 1\%$)
- R₁₇, R₁₈—470,000 ohm, 1/2 w. res. (matched $\pm 1\%$)
- R₁₉, R₂₀—47,000 ohm, 1/2 w. res. (matched $\pm 1\%$)
- R₂₁, R₂₂—10,000 ohm, linear taper pot
- R₂₃, R₂₄, R₂₅—100,000 ohm, 1/2 w. res.
- R₂₆, R₂₇—1000 ohm, 1/2 w. res.
- R₂₈—2500 ohm, 5 w. res.
- R₂₉—15,000 ohm, 1/2 w. res.

- R₂₉—18,000 ohm, 1/2 w. res.
- R₃₀—100 ohm wirewound pot
- C₁—47 μ fd. ceramic capacitor
- C₂, C₃, C₄, C₅, C₆, C₇—20/20/20 μ fd., 450 v. elec. capacitor
- C₈, C₉, C₁₀, C₁₁—25 μ fd., 600 v. capacitor
- C₁₀—1 μ fd., 400 v. capacitor
- C₈, C₁₀—10 μ fd., 600 v. oil or elec. capacitor
- C₁₁, C₁₀—20 μ fd., 150 v. elec. capacitor
- T₁—Output trans. (*Acrosound* TO-330, see text)
- T₂—Power trans. 400-0-400 v. @ 200 ma.; 5 v. @ 2 amps.; 6.3 v. @ 5 amps.
- T₃—Fil. trans., 6.3 v. @ 1 amp.
- CH₁—4 hy., 200 ma. filter choke
- S₁—S.p.s.t. switch
- F₁—5 amp. line fuse
- F₂, F₃—1/8 amp. cathode fuse
- SR₁—50 ma. selenium rectifier
- V₁, V₂—6SN7 tube
- V₃, V₄—6550 tube (*Tung-Sol*)
- V₅—5V4GA tube

procedure local feedback may be added to stages within the amplifier in order to reduce their contribution to the phase shift. Thirdly, the bandwidth of the stages may be extended by the use of certain design techniques, and the phase shift correspondingly reduced.

The first method is subject to the criticism that it restricts the amplifier band reducing the rise time with regard to square wave response and, in this manner, affects the fidelity of transient reproduction. The first and third methods may be combined; the bandwidth increased and then loss networks added.

An appreciable increase in the bandwidth of the amplifier described has been achieved by the use of the TO-330 transformer. The response of the amplifier, with feedback, is flat to over 200 kc. An adequate stability margin of 10 db has been maintained by the use of methods two and three. A small amount of degeneration has been added to the driver stage by the inclusion of individual cathode resistors. The sub-sonic bandwidth has been extended and shaped by the addition of the 470,000 ohm coupling resistors R_{12} and R_{13} . With these added resistors the bias developed on the grids of the output tubes is partially dependent on the voltage developed at the plates of the driver tubes, and a plate current balancing control has been added to the output stage. The procedure of balancing plate current has been facilitated by individually fusing the output tube cathodes. To check plate current the fuse is removed. The fuse clip serves as a convenient tie point for the connection of a milliammeter.

Construction of the Amplifier

The amplifier can be constructed on a chassis 8" x 12" x 3". A careful arrangement of parts permits direct point-to-point wiring of the stages and a short, direct feedback connection between the output transformer and the first stage. The axiom for wiring amplifier stages is to have leads as short

and direct as possible. It is desirable to twist filament leads; also leads to the power switch and preamp power connector. A neater job will usually result if filament, switch, and power supply circuits are wired first, then "B plus" circuits, then signal circuits less the coupling capacitors. The coupling capacitors are added last, and since these are generally large, they may be looped over the space from stage-to-stage. The coupling capacitors to the output stage can terminate on pin No. 6 of the 6550 sockets, since this pin is not a tube connection and can be used as a tie point.

Care should be taken when wiring the output transformer to see that the proper color coding is observed for the primary leads. Make certain that the tracer leads are connected to the output tube that is energized from the cathode of the phase inverter. If these leads are incorrectly connected the amplifier will motorboat when it is turned on. Correct phasing can be restored by either reversing the transformer leads or by reversing the connections of the coupling capacitors at the phase inverter section of the first tube.

The total cathode current per output tube will run about 75 ma. with a plate supply voltage of 425 volts and a grid bias voltage of minus 48 volts. If the cathode current differs considerably from this figure, it may be advisable to change the value of R_{20} until normal bias and plate current is obtained.

Performance

The measured performance figures of the amplifier are given in Table 1. The frequency response at low output levels is flat from 2 cps to 220 kc. Maximum power of 60 watts is delivered at all frequencies between 20 cps and 30 kc. The square wave response at 20 cps shows 5% droop, and at 20 kc. the square wave is clean with no overshoot and a rise time of 2 microseconds. The

intermodulation figures are particularly good, being only 0.15% at 30 watts and 0.6% at 60 watts, for a combination frequency of 60 and 3000 cps, mixed 4:1.

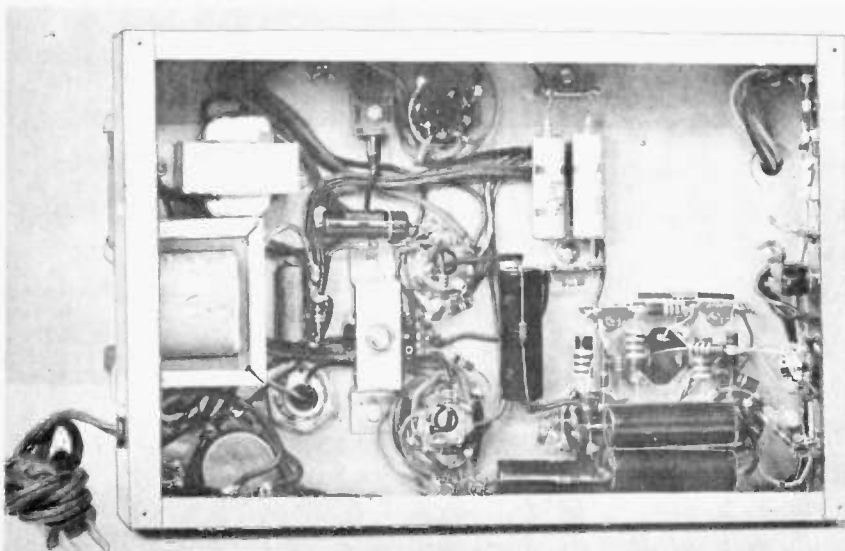
Much has been written about amplifier testing and on the interpretation of test results. However, the deeper one goes into the field of amplifier design, the more apparent it becomes that the best test instrument is the human ear with music supplying the signal source. Unfortunately, the ear cannot supply a numerical rating of merit, but only a comparison of "better" or "not as good." The amplifier described in this article has been subjected to comparative listening tests with both the older "Ultra-Linear" Williamson using KT-66's and with other good amplifiers. Listeners were generally agreed that this amplifier had many points of superiority.

The relative importance of the power amplifier in a high-fidelity system has always been a controversial subject. There are those who maintain that a low power amplifier of 5 watts or so is adequate for good reproduction and qualify this by the indisputable statement that the average sound power required for good room volume is no greater than this figure. Others state that a moderately good power amplifier is a much more perfect device than other elements of the reproducing system, in particular phono pickups and speakers. Both of these schools of thought fail to recognize some basic facts relating to requirements imposed upon the power amplifier. In the first case, although average room volume may require only a few watts, peak powers may exceed the average by 10 times or more, and it is the fidelity with which these peaks are reproduced that contribute to the feeling of presence. With regard to the second point, it is true that there is still room for improvement in pickups and speakers, however, any additional contribution to intermodulation distortion in the power amplifier makes itself felt by the generation of new combination tones which further change the identity of a musical instrument, and produce the effect of blurring the sound. This latter effect can be easily demonstrated in a comparison test by playing a poor recording with lots of surface noise. The noise will be there in each instance, but will be much less objectionable with an amplifier of lower distortion. Finally, one must recognize that a condition of interaction exists between the speaker and the power amplifier. A high power amplifier of low and constant internal impedance exerts better control over the speaker characteristics at high peak powers.

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3.: Wireless Engineer (Editorial), August, 1955.

Underchassis view of amplifier. Note the point-to-point wiring of circuit.



A 35-WATT "INFINITE FEEDBACK" AUDIO AMPLIFIER

By CHARLES P. BOEGLI
Cincinnati Research Company

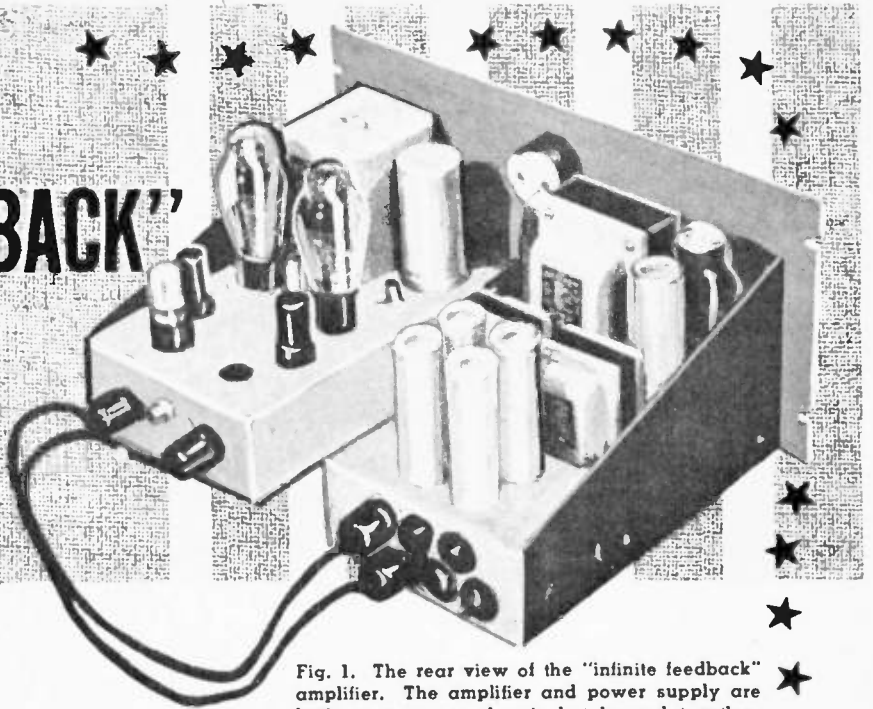


Fig. 1. The rear view of the "infinite feedback" amplifier. The amplifier and power supply are built on separate chassis but housed together.

A rather novel circuit—it incorporates, in a single compact design, both positive and negative feedback.

DURING the last six years advances in high-quality sound reproduction have been most marked. Phonograph pickups now available at modest cost track with forces that would have seemed impossibly low only a short time ago, and have responses wider and smoother than those obtainable from the most expensive units of earlier days. Speakers and their enclosures are being developed at almost as rapid a rate.

As far as amplifiers are concerned, several recent noteworthy circuits have frequency responses much wider and distortion levels much lower than were afforded even by laboratory amplifiers of a decade ago. The excellent performance of these circuits is generally obtained by relatively large amounts of negative feedback over at least three stages. Instability is usually avoided by various novel means, such as the direct coupling of several stages within the loop. The resulting circuits often need to be constructed with accurately calibrated components and sometimes require some adjustment for best performance; these extremes are necessary for reasons that will become evident later in this article.

In any amplifier employing large amounts of negative feedback over several stages, the frequency response of that part within the loop must be very carefully controlled. Almost without exception, the final circuits display over-all frequency responses much wider than are required for top-quality reproduction. The *object* is to attain low distortion and low output impedance—the *penalty* is the wide frequency response. As a matter of fact, an entire reproducer system does not need a flat response below 16 cps or over 30,000 cps but the distortion must be quite low and the output impedance must be small—the latter especially if the speaker is one of low efficiency.

Actually, the frequency response and output impedance of an amplifier are not independent when the unit is loaded with a speaker. An amplifier of high output impedance and flat response with a resistance load will show a very uneven response, particularly in the bass region, when connected to a speaker.

The lower limit to the output impedance of the amplifier would seem to be a negative resistance equal to the d.c. resistance of the speaker voice coil in series with the speaker lines and the output transformer secondary. Even with this negative resistance the speaker cone could conceivably make some undesirable excursions because the speaker is not 100% efficient in converting electrical to mechanical energy and *vice versa*. In practice, such large negative impedances are difficult to realize with an amplifier of low distortion and good stability. Granting that they might be obtained, they might nevertheless be undesirable, for the large error-correcting voltage appearing within the amplifier in case the cone made undesirable movements would undoubtedly overload the amplifier (unless it had enormous reserve power) and cause considerable harmon-

ic distortion. A lower limit to the output impedance seems necessary, but the magnitude of this limit depends upon many factors, among them the power-handling capacities of the amplifier and the efficiency of the speaker.

The amplifier covered in this article was originally intended to have approximately zero output impedance along with negligible distortion and adequate frequency response. These goals have been accomplished with a simple and trouble-free circuit, requiring no precision components or delicate adjustments, and making use of both negative and positive feedback. The principle is well known but the manner of utilizing it in this amplifier is considered to be novel and simpler than that of previously-designed circuits.

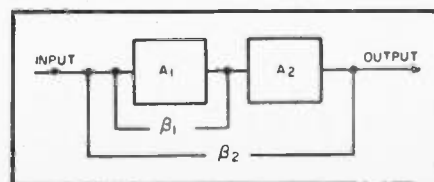
The Amplifier

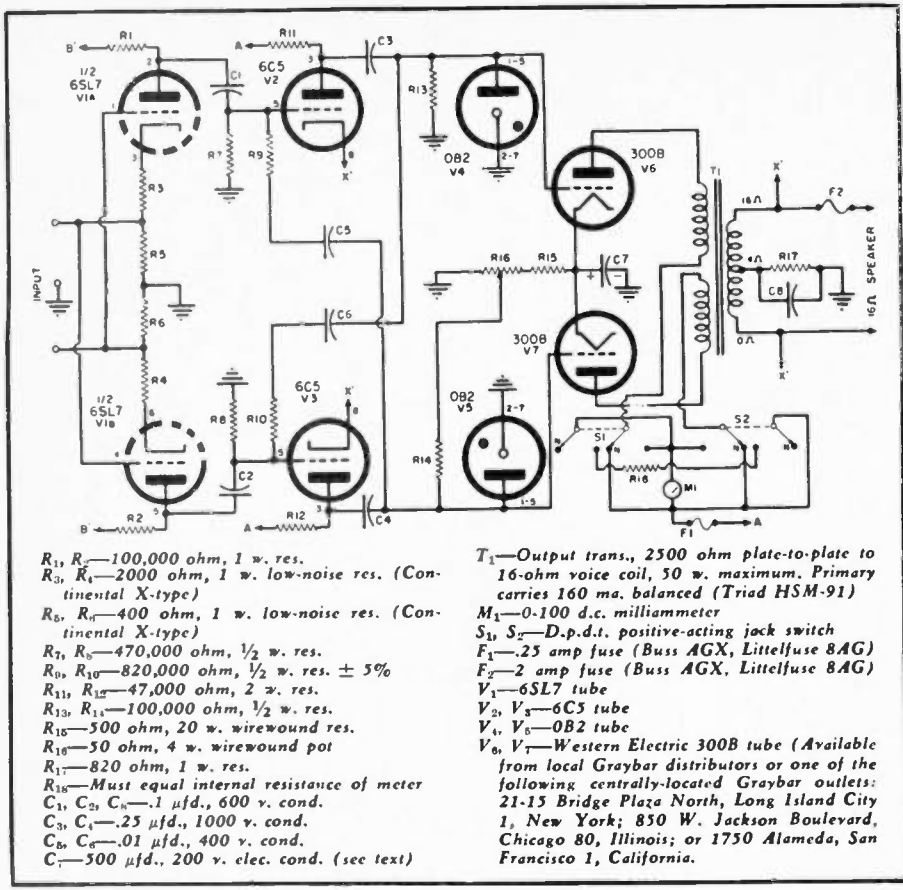
The reduction of distortion effected by negative feedback in any amplifier is, to a first approximation, directly proportional to the gain reduction¹:

$$\frac{A'}{A} = \frac{1}{1 - \beta A} \approx \frac{D'}{D} \dots (1)$$

where β is the feedback factor (negative for negative feedback), A and D are the amplification and distortion of the amplifier without feedback, and A' and D' are the same characteristics with feedback. Distortion reduction is thus dependent not only upon the feedback factor but also on the original amplification A . The output transformer of an audio amplifier acts as a step-down unit as far as voltage is concerned; if it is to be part of the

Fig. 2. Block diagram of amplifier with two feedback loops. See discussion in article.





tortion in the output stage is removed and unwanted speaker excursions are eliminated to an extent dependent solely upon the characteristics of the speaker. If $A_1\beta_1$ is slightly greater than 1.0 then any disturbances to the output are actually overcorrected. While the speaker damping is improved the distortion becomes worse. When $A_1\beta_1 = 2.0$ the distortion is equal to that resulting if the positive feedback were removed entirely.

An amplifier of this type will be unstable (it will oscillate) when the denominator of equation (2) becomes equal to or less than zero, that is, when $A_1\beta_1 + A_1A_2\beta_2$ becomes equal to or greater than 1.0. The stability of the amplifier increases as the quantity $A_1\beta_1 + A_1A_2\beta_2$ becomes smaller. Maximum stability is thus achieved by using as much negative feedback as is conveniently possible. At very low or very high frequencies where, as a result of phase shifts in the negative feedback loop, instability may tend to exist, the positive feedback can be reduced and shifted in phase to keep the sum $A_1\beta_1 + A_1A_2\beta_2$ less than 1.0. It should be noted, however, that with sufficient negative feedback the midrange positive feedback can be greater than unity without instability, which means that within certain limits a perfectly stable negative output impedance can be obtained. In the amplifier to be described the quantity $A_1\beta_1$ is made slightly greater than 1.0 and 100% negative feedback is employed. This combination insures minimum distortion and maximum stability, together with an output impedance near zero and an over-all voltage gain near unity. The power gain, however, is quite large.

Since in this type of circuit instability is avoided primarily by regulating the positive feedback, the response of the entire circuit within the negative feedback loop need not be controlled very closely, and the final amplifier may display an extremely low output impedance without necessitating excessively wide frequency response.

Aside from reasons of stability there is a significant advantage in using 100% negative feedback, and this is that the response of the entire amplifier is almost completely independent of the response of that part within the loop. Rather drastic measures may then be employed to eliminate any tendency toward oscillation without effect on the over-all frequency response.

The low output impedance is, of course, limited by the power handling capability of the output stage, which is not affected by the negative feedback. Only a limited amount of power can be drawn by very small load impedances without overloading the output stage, whereupon distortion sets in. The amplifier is thus unable to correct for errors in the output voltage when these errors are large and the load impedance is small, which means that overdriving the speaker will generally result in considerable distortion or other undesirable effects.

feedback loop a worthwhile reduction in distortion can be effected only by including a fairly large number of amplifier stages within the same loop. Avoiding oscillation in multistage amplifiers with over-all feedback, however, poses a very difficult problem, which is solved in such circuits as the Williamson by using direct coupling between at least two of the stages to eliminate part of the phase shift caused by ordinary RC interstage coupling.

The amplifier to be described accomplishes the same result with just two stages, namely, driver and power output. By means of positive feedback the gain of the driver stage is increased approximately to infinity; that is, if the negative feedback were removed the driver would oscillate. This principle has certainly been used before² but generally in the design of cheap amplifiers. Its application to high-quality amplifiers seems largely to have been overlooked.

The gain of an amplifier of this type is:

$$A = \frac{A_1A_2}{1 - A_1\beta_1 - A_1A_2\beta_2} \dots \dots (2)$$

where the symbols A and β represent the amplification and feedback factors shown in Fig. 2. If the driver stage is caused to oscillate, then $A_1\beta_1$ is equal to 1.0 and the equation simplifies to

$$A = - \frac{1}{\beta_2}$$

For negative feedback β_2 is negative so that the over-all amplification is

the reciprocal of the negative feedback factor. In a single-loop amplifier this result could be obtained only with an infinite gain within the loop. The feedback would then be infinite (in terms of decibels) and for that reason the present amplifier is called the "infinite feedback" amplifier.

The output impedance is: $Z_o =$

$$Z_L \frac{1}{\left(1 + \frac{Z_L}{Z_{p2}}\right) \left(1 - \frac{A_1A_2\beta_2}{1 - A_1\beta_1}\right) - \left(1 - A_1\beta_1\right)} \dots \dots (3)$$

where Z_o is the observed output impedance, Z_L is the load impedance, and Z_{p2} is the output impedance without feedback. If $A_1\beta_1 = 1.0$ then $Z_o = 0$; or the amplifier has zero output impedance. The rate of change of Z_o as $A_1\beta_1$ is varied slightly from 1.0, however, depends upon β_2 , the amount of negative feedback, and will be smaller as the negative feedback becomes greater.

If the output of such an amplifier is subjected to some voltage disturbance δ (which might arise, for instance, from an unwanted oscillation of the speaker cone or the introduction of distortion in the output stage), then a correcting output of magnitude

$$E_o = \frac{A_1A_2\beta_2}{1 - A_1A_2\beta_2 - A_1\beta_1} \delta \dots \dots (4)$$

also appears at the output terminals. When $A_1\beta_1 = 1.0$, $E_o = -\delta$, which means that the correcting voltage is equal in magnitude but opposite in sign to the disturbance, or that all dis-

Circuit Design

The theoretical reduction of distortion is seldom attained in practice, so every possible step must be taken to improve the quality of the amplifier prior to the introduction of feedback. In this particular version the output tubes are 300B's in straight class-A push-pull operation with 460 volts on the plates. The drivers are 6C5's with a plate supply of approximately 550 volts. To develop full amplifier output each 6C5 must deliver a signal of about 100 volts peak to the grid of the following 300B; the gain of the 6C5 is about 14 so the peak grid signal at the 6C5 is about 100/14 or 7.1 volts.

Some speculation may arise as to why such expensive tubes as 300B's have been used in this amplifier (they cost approximately \$9.00 apiece). It was the object of this work to provide an amplifier of around 30 watts output and 300B's are the only triodes capable of supplying this output in the desirable class-A operation. No doubt some available power tetrodes or pentodes could have been used, connected as triodes, but it has been the writer's observation that results are never as satisfactory as when triodes are used. Part of the problem probably arises from the difficulty of balancing two tetrodes when they have been so connected and making each tube do its share of the work. 6C5's were selected as drivers because they are capable of delivering the required output voltage with very low distortion even in the absence of negative feedback; in this respect they seem to be slightly superior to the more common 6J5's.

Push-pull negative feedback is taken from the secondary of the output transformer to the cathodes of the 6C5's. The entire secondary of the transformer is thus at a d.c. potential of about 8 volts above ground, so the speaker lines must not be grounded at any point. Positive feedback from the plate of each 6C5 to the grid of the other 6C5 is adjusted so the driver stage is oscillating weakly when the negative feedback is removed. A low-cut filter comprised of R_0C_0 in the positive feedback network prevents oscillation at very low frequencies where the over-all negative feedback loop gain begins to fall off and a high-cut filter made up of R_0C_0 and R_0C_1 (where R_0 is the output resistance of the 6C5, C_0 is the capacity at the grid of the 300B, and C_1 is the total capacity at the grid of the 6C5) performs the same functions at high frequencies. Such simple precautions are insufficient unless an output transformer of very high quality is employed, and the sizes of the components in this loop are largely dependent upon the characteristics of the output transformer. Because push-pull negative feedback is taken from two ends of the voice-coil winding of the transformer, the capacities to ground at the two ends must be about equal.

This circuit was constructed with a standard output transformer having 4, 8, and 16-ohm taps on the secondary.

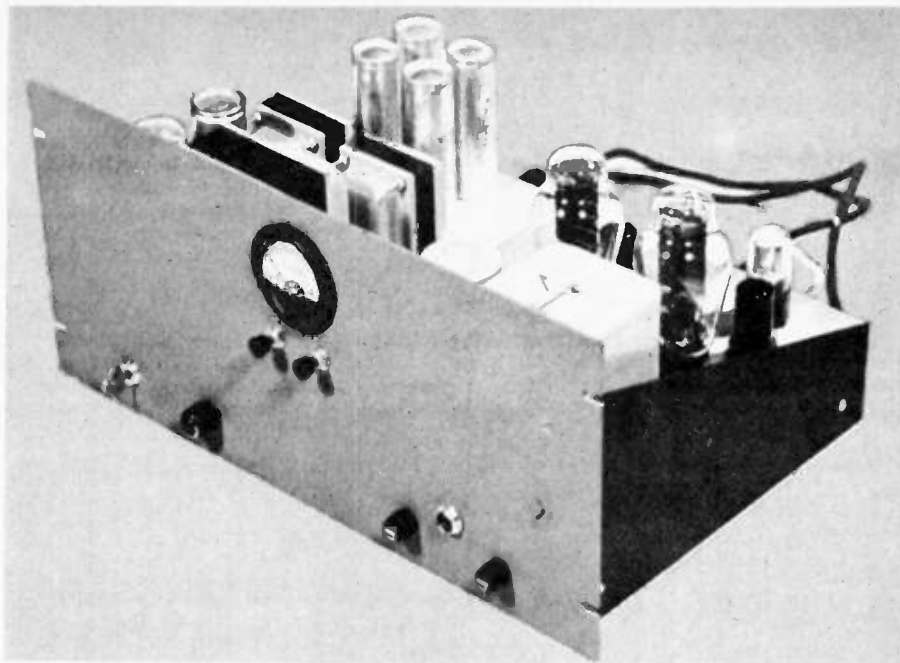


Fig. 4. Front panel view of 35-watt amplifier. The meter is permanently in circuit.

The 4-ohm tap was used as a center-tap and a balanced output was taken from the 0 and 16-ohm taps. This arrangement has been quite adequate, since the d.c. resistances and the output voltages each side of the "center tap" were very well balanced with the transformer used. An 8-ohm speaker, however, is necessarily connected in an unbalanced manner, which may be less satisfactory than when a 16-ohm speaker is attached. For this reason it would be desirable to have a transformer with 4, 8, and 16-ohm balanced secondary windings; at present such a transformer is not commercially available.

The inverter is an essential part of the amplifier because it prevents vari-

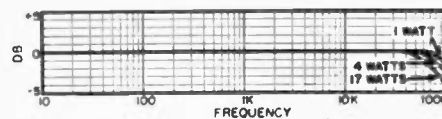


Fig. 5. Response of "infinite feedback" unit.

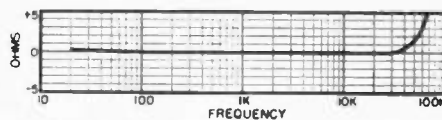
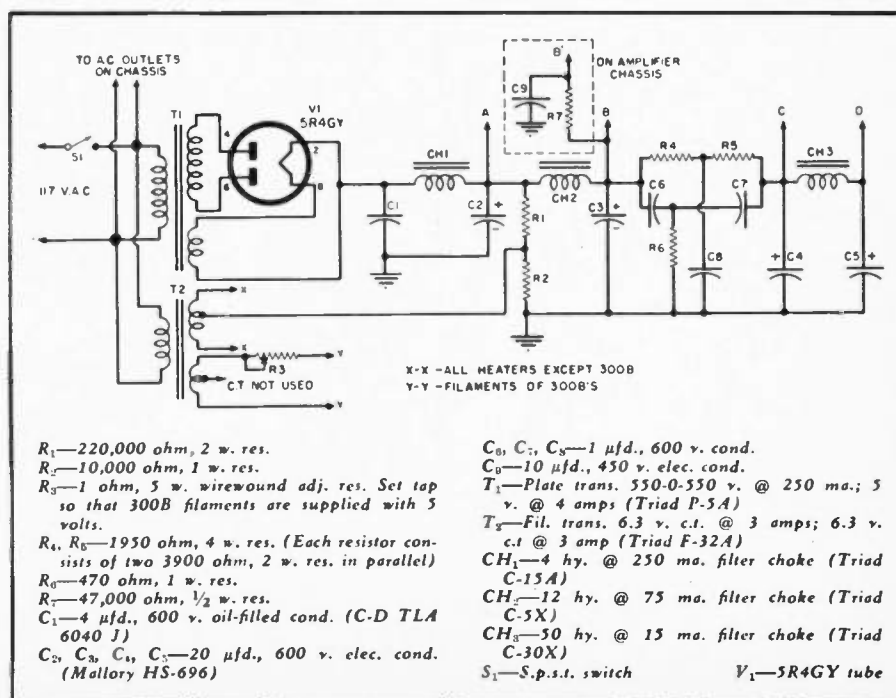


Fig. 6. Output resistance of the amplifier at .78 volt into a speaker of 16 ohms.

ous input sources from influencing the positive feedback. The circuit is the writer's simplification of the cross-coupled amplifier developed by Van

Fig. 7. Power supply diagram. Connections C and D are used for preamp power supply.



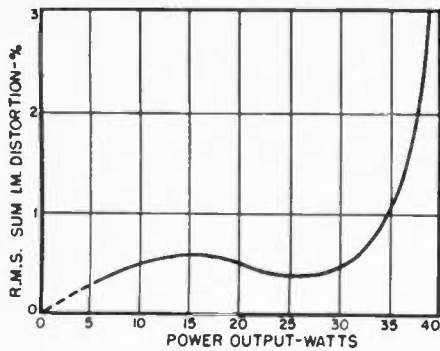


Fig. 8. Intermodulation distortion of the "infinite feedback" amplifier. The resistive load is 16 ohms and input is balanced.

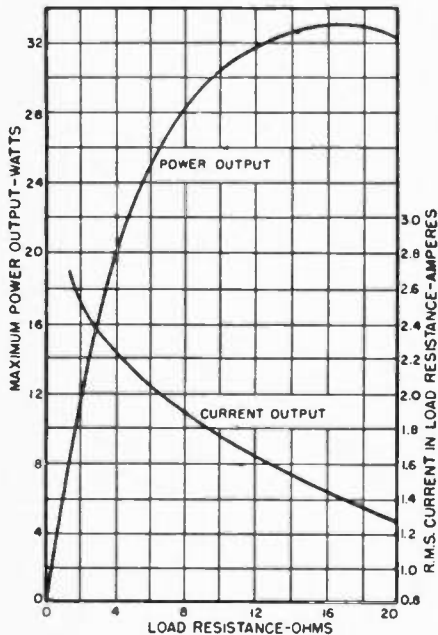


Fig. 9. The calculated maximum power and current delivered by the amplifier into various load resistances. Refer to text.

Scoyoc³ and performs essentially the same function as a line-to-push-pull grid transformer. It has been fully described in a previous article.⁴

The circuit of the complete amplifier is shown in Fig. 3; the parts list describes the components that were used in the experimental model, with

a few exceptions. Only one generally unavailable part has been used in the experimental construction, namely, the 500- μ fd., 200-volt bypass on the 300B cathodes. This condenser may of course be eliminated but it is desirable to use as large a condenser here as possible. A 300- μ fd., 150-volt unit, which is commercially available, would be quite satisfactory.

The circuit diagram of the power supply designed for the experimental model is shown in Fig. 7. It is built on a 7" x 12" x 3" chassis which is a twin to the amplifier chassis. Fig. 1 is an over-all view of the completed amplifier and power supply.

Performance

Fig. 5 shows the frequency responses at various power outputs into a 16-ohm resistive load. The response, particularly at high frequencies, is dependent upon the power that must be supplied. It may be remarked at this point that this amplifier falls outside the class of minimum-phase structures; that is, the phase curve cannot be computed from the frequency-response curve. The response curves show that 3 db less than full power is delivered ± 0 db over a range of 16 to 40,000 cps; a tribute to the high quality of the output transformer. Fig. 6 shows the output resistance of the amplifier on the 16-ohm tap as related to the frequency.

The r.m.s. sum intermodulation distortion for frequencies of 60 and 3000 cps (4:1) at different power levels is shown in Fig. 8. Below the point at which the grids of the 300B's are starting to be driven positive, the intermodulation distortion of the entire amplifier is substantially the same as that of the inverter, from which it might be concluded that the power-amplifier stages are practically distortionless. The intermodulation distortion is less than 2% for power levels less than 37.5 watts.

The r.m.s. sum intermodulation distortion is almost always greater than the harmonic distortion, and is a fairly good measure of the cleanness of the output. For advertising purposes some

manufacturers have published figures for various other types of IM distortion measurements (such as the level of the first-order difference tone only) which are much lower than those for r.m.s. sum IM distortion and must not be confused or compared with them.

The output tubes in this amplifier are operated very near their maximum ratings. For this reason a plate-current meter should be provided and the plate currents balanced carefully. The single balancing control designed into the amplifier, when rotated, increases the plate current of one tube at the same time it decreases that of the other. It is sometimes necessary to interchange the positions of the 300B's to arrive at a balance. The zero-signal plate currents of the output tubes are approximately 94 ma. per tube. If the output leads of the amplifier are shorted together the negative feedback is removed and the oscillation of the drivers then tends to drive the grids of the output tubes well into the positive region. 300B's are not intended for operation in this region and will quickly be destroyed by this eventuality. The same type of destruction will occur if the output is effectively short-circuited such as by severely overdriving the speaker. A number of protective devices have been employed to prevent this type of destruction. Not only the plate supply but also the speaker line is fused and 0B2 voltage-regulator tubes are placed across the 300B grids. Of these three precautions the 0B2's are the most effective.

The power-handling capabilities of the output stage are not influenced by negative feedback, which simply attempts to keep the output voltage constant irrespective of changes in the load resistance. Fig. 9 shows the theoretical maximum output power this amplifier can deliver into various load resistances and also the maximum current that will flow in the load. When the output is shorted the current rises to a very high value, and the quick-acting fuse in the output will blow.

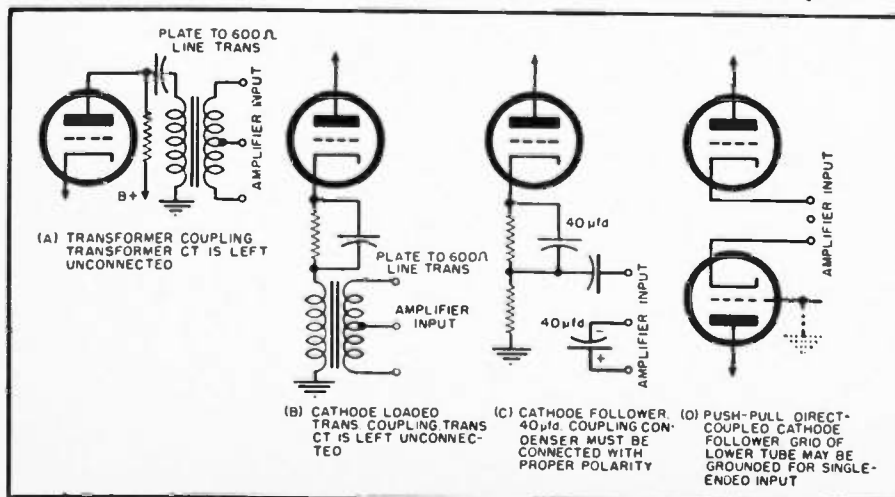
The unit requires .43 volt balanced or unbalanced input to be driven to full output. The noise level of the experimental unit measured .005 volt, or 74 db below 35 watts, with input shorted. The damping factor of the amplifier is, of course, infinity.

Because of its low input impedance the amplifier cannot be driven by many of the preamplifiers now in use without special precautions. Fig. 10 illustrates several schemes by which existing preamplifiers may be employed or new units designed.

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Fig. 10. Several methods of connecting preamps to the "infinite feedback" amplifier.



"ULTRA-LINEAR" OPERATION OF 6V6 TUBES

By **DAVID HAFLER**
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Fig. 1. The Grommes 100BA power amplifier after conversion to "Ultra-Linear" operation. See diagram of Fig. 4.

One of the best designs in recent years covering an audio amplifier using 6V6 tubes. The author, in this case, has converted a Grommes unit to illustrate his design idea.

EVER since the introduction of "Ultra-Linear" circuitry,¹ there has been a steadily growing interest in amplifiers utilizing this type of output stage coupling. The basic arrangement has become popular in ardent audiophile circles and has also found commercial and industrial applications where extremely low distortion is required.

Essentially, the "Ultra-Linear" circuit is illustrated in Fig. 2. The screens of beam power output tubes are connected to taps on the primary of the output transformer; or if it is desired to operate the screens at a different a.c. potential than the plates, to a tertiary winding on the output transformer. Either arrangement requires a transformer with the correct ratio of screen load to plate load if optimum results are to be obtained, and a mismatch will lead to inefficiency and/or increased distortion.

The "Ultra-Linear" arrangement has been mistakenly referred to as a feedback circuit. This is not correct since negative feedback would produce a reduction in gain which does not occur with the "Ultra-Linear" circuit. It would be just as incorrect to refer to a triode as a tetrode with feedback as it is to analyze the "Ultra-Linear" circuit as a feedback circuit. Instead it must be considered as a new and different type of tube structure which is neither triode nor tetrode.

The circuit provides some of the advantages of both triodes and tetrodes, and it overcomes some of the disadvantages of each of these types. For example, it is more efficient and provides more power output than triodes. Its capabilities in this respect parallel the capabilities of tetrodes. However, it has low internal impedance, almost as low as triodes and about one-tenth that of tetrodes; this provides good

loudspeaker damping. Lastly, and most important of all, it has a more linear input-output relationship at most power levels than either triodes or tetrodes which means that its distortion is lower than other methods of operation. This alone justifies the use of the circuit in those cases where low distortion is the guiding criterion.

The "Ultra-Linear" circuit has achieved popularity in deluxe amplifier arrangements such as conversion of the Williamson circuit.² It has been widely used with tubes of the KT66, 807, and 5881 type for circuits in the 20 to 30 watt power bracket—for circuits of truly outstanding characteristics suitable for the most critical usage. Naturally, 20 or 30 watts is a lot of power for living room use—just as 200 horsepower is a lot of power for a deluxe automobile. However, there are definite advantages to high powered amplifiers which are operated at a fraction of their potential output just as there are definite advantages to high powered cars which are run at a fraction of their capabilities.

Nevertheless, not all of us want, or can afford, 200 horsepower cars; and not all of us feel the need for, or wish to spend the money for, amplifiers of 20 or more watts power rating. Many audiophiles and music lovers are very happy with amplifiers in the 10 to 15 watt power bracket. The popularity of this range is demonstrated by the sales success of thousands of Williamson-type amplifiers as well as tens of thousands of lower cost amplifiers using 6V6 tubes providing 10 to 15 watts of power output. Undoubtedly, the greatest number of amplifiers in home use utilize the type 6V6 tube in one of several popular circuit arrangements, all of which have essentially similar performance characteristics.

The possibilities of using the "Ultra-

Linear" arrangement with 6V6 tubes in medium-powered amplifiers has been investigated carefully. It has been found that the tube is well suited for this mode of operation since its dynamic input-output characteristic can be linearized by proper selection of a tapping point for screen connection.

The characteristics of the 6V6 are not at all similar to the 6L6 family, and the connection arrangement which is optimum for 6V6's is quite different from that which can be used with the large tube types. As a tetrode, the 6V6 permits 10 to 15 watts of output depending on plate supply voltage and bias. These ratings are based on the point where clipping of a sine wave becomes visible—which happens when the grids start to go positive, and the driving source cannot furnish power to the tubes.

If the same tubes are triode connected (by strapping the screen to the plate), power output, using the same criteria, is reduced to 2½ to 3½ watts. When the "Ultra-Linear" connection is used, the power output depends on the position of the screen taps. If a 50% tap is used, power is reduced to about one-half of the tetrode capability. If a greater than 50% tap is used, power is reduced toward the triode limitations. At a tapping point of about 24%, power output is within 90% of the tetrode condition, and distortion at all levels up to maximum is minimized. This point, therefore, has been selected as the optimum operating point for "Ultra-Linear" use.

It would be possible to take an even lower tapping point and obtain slightly more power output than the tetrode connection, but the distortion at low levels and the internal impedance both begin to increase as the tap is brought closer to the zero per-cent point which

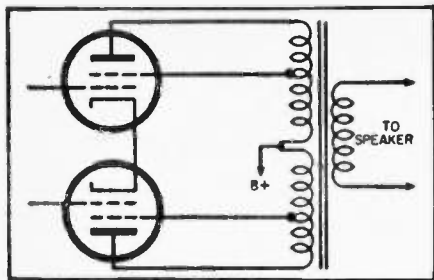


Fig. 2. Basic "Ultra-Linear" arrangement.

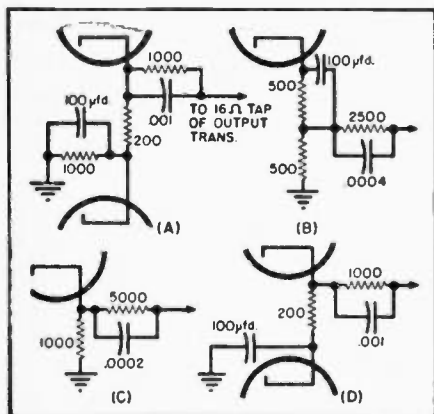


Fig. 3. Various feedback arrangements from voice coil of the output to the cathode of an early stage. See text for discussion.

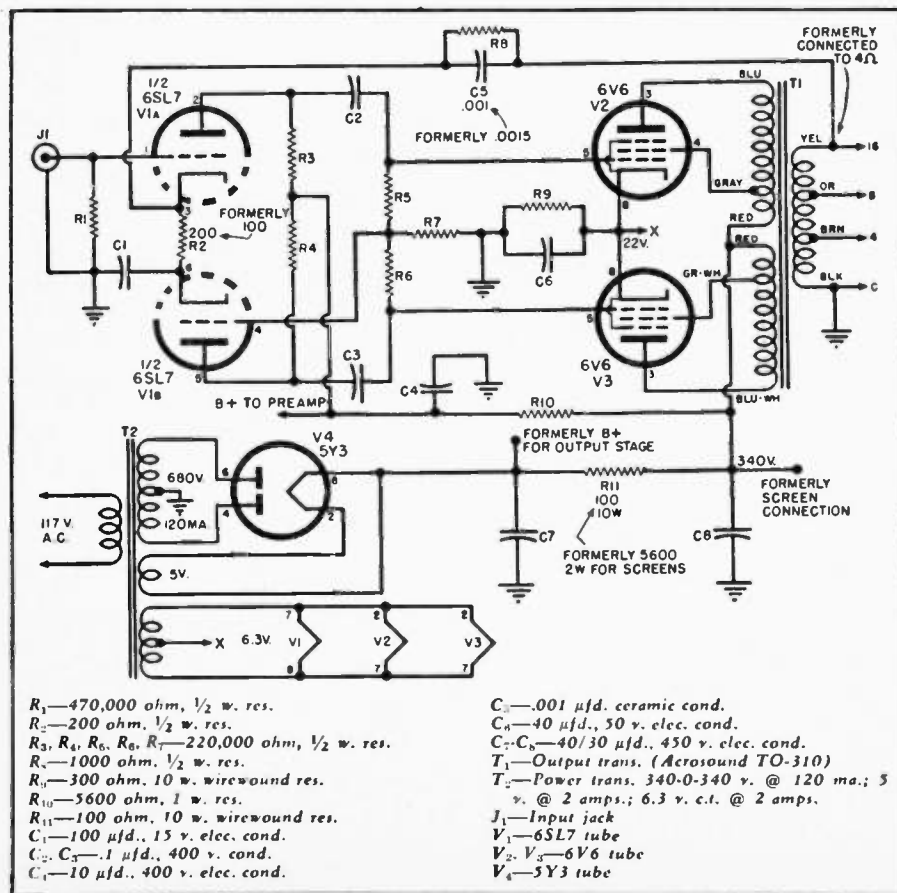
is coincident with conventional tetrode connection.

Thus the "Ultra-Linear" operating point has been set at a compromise level in which the factors of maxi-

imum power output, distortion at various levels, and internal impedance have all been weighed against each other. It must also be mentioned that listening tests at various tapping points with no feedback around the amplifier validate this selection of the tapping point. This was done without feedback on the assumption that the best amplifier without feedback would also be the best after the application of feedback. In these listening tests, the triodes fell behind because they could not handle the power (after all, 3 watts is insufficient for musical peaks), the tetrodes were somewhat screechy and boomy (too much internal impedance for satisfactory speaker damping), and the 24% point sounded natural and smooth even without connection of the amplifier feedback loop.

Use of the "Ultra-Linear" circuit involves utilization of an output transformer with the correctly placed taps. A special transformer, the *Acrosound TO-310*, has been designed specifically for this application; and its parameters were selected so that it would not limit the ultimate capabilities inherent in the circuit. For example, its bandwidth has been set at ± 1 db from 10 cps to 100 kc. so as to provide the low phase shift and good transient performance desired in the most critical applications. Similarly, its distortion characteristics complement those of the "Ultra-Linear" circuit and permit low distortion at both high and low levels from 20 cps to over 20 kc.

Fig. 4. The Grommes 10³A power amplifier converted to "Ultra-Linear" operation.



- R₁—470,000 ohm, 1/2 w. res.
- R₂—200 ohm, 1/2 w. res.
- R₃, R₄, R₆, R₈, R₇—220,000 ohm, 1/2 w. res.
- R₅—1000 ohm, 1/2 w. res.
- R₉—300 ohm, 10 w. wirewound res.
- R₁₀—5600 ohm, 1 w. res.
- R₁₁—100 ohm, 10 w. wirewound res.
- C₁—100 μfd., 15 v. elec. cond.
- C₂, C₃—1 μfd., 400 v. cond.
- C₄—10 μfd., 400 v. elec. cond.
- C₅—0.001 μfd. ceramic cond.
- C₆—40 μfd., 50 v. elec. cond.
- C₇, C₈—40/30 μfd., 450 v. elec. cond.
- T₁—Output trans. (*Acrosound TO-310*)
- T₂—Power trans. 340-0-340 v. @ 120 ma.; 5 v. @ 2 amps.; 6.3 v. c.t. @ 2 amps.
- J₁—Input jack
- V₁—6SL7 tube
- V₂, V₃—6V6 tube
- V₄—5Y3 tube

Circuit Considerations

There are many 6V6 circuits which have become popular, but by far the most commonly used is that in which a twin triode phase inverter is used to drive a pair of 6V6's; and feedback is carried from the output winding of the output transformer to the cathode of one of the triode sections. This basic configuration is simple, practical, economical, and adequate. The a.c. grid-to-grid voltage requirements of the 6V6 output stage are not stringent, and the phase inverter supplies ample drive without the need for an intermediate push-pull stage such as is used in the Williamson-type circuit. Since there are only two stages, the problems of utilizing feedback are simplified (as there is less phase shift in the circuit), and the designer can use less elaborate circuitry and components while preserving a satisfactory margin of stability.

Generally the phase inverter tube is a high mu triode such as the 6SL7 or 12AX7 in order to obtain as much gain as possible within the two stages. Actually, except for gain considerations, the specific type of inverter is of comparatively little consequence—circuit performance is determined almost completely by the mode of operation of the output tubes with respect to bias, supply voltage, and impedance match; the quality of the output transformer; and the proportion of feedback. The voltage amplifier stage contributes relatively little, as compared to the contribution of the output stage, to the over-all quality of the amplifier.

Conversion of these circuits to "Ultra-Linear" operation can be done by substituting an output transformer which has properly placed taps for connection to the 6V6 screens. Generally, this substitution will make an immediate decrease in distortion.

If the original amplifier used a screen dropping resistance, this is removed for "Ultra-Linear" operation; and the screens are connected to the tapping points on the primary of the output transformer. It is important to observe polarity and to connect the screen to the same primary side of the transformer as that from which the plate is energized. Otherwise an oscillatory condition will be provoked. Similarly, polarity must be observed between upper and lower output tubes, or the feedback from the secondary side of the transformer may be in the incorrect phase and cause regeneration.

When the screen resistor of the original circuit has been removed, the screen bypass condenser must also be disconnected. This can be readily put to good use by paralleling it across one of the filter condensers of the power supply for extra filtering and lowered power supply impedance.

The only other changes which need be made involve the feedback resistor and feedback compensating condenser which shunts this resistor (or in some circuits bypasses it to ground). The ratio of series resistor to shunt re-

sistor in the feedback path determines both the total gain in the circuit and the proportion of feedback. For example, with a 6SL7 phase inverter and feedback from the 16-ohm tap of the Acrosound TO-310 transformer, the power amplifier will have 17 db of feedback and require a maximum input signal of 3 volts to drive it to full output when the ratio of feedback to cathode resistance is 5 to 1. If the ratio is changed to 7.5 to 1, the amplifier will be driven with a 2 volt input, but the feedback is cut down by 3 to 4 decibels. Similarly, a 12AX7 has about 50% more gain than a 6SL7. If this tube is used with a 7.5 to 1 ratio of resistance, the amplifier can be driven to full output with 2 volts of signal while still maintaining 17 db of feedback. In the original construction, it is recommended that the 12AX7 be used so as to obtain this increased sensitivity. However, in converting an existing amplifier, the constructor can leave the 6SL7 tube in the circuit and can adjust for the required sensitivity by varying the feedback resistor. If necessary, he can sacrifice a portion of the feedback in order to maintain sufficient gain for the pre-amplifier stages which are being used.

In many commercial amplifiers, the power amplifier section must be sufficiently sensitive to be driven by 1 volt of input because of the relatively low gain of the earlier stages. If this is the case, it is necessary to diminish the feedback (by increasing the feedback resistor). However, the most modern preamp designs are intended to supply about a two volt input such as is found on Williamson-type amplifiers. Any of these preamps will handle the converted 6V6 amplifier and still permit 14 or more db of feedback. This is sufficient feedback to reduce distortion, hum, noise, and internal impedance to low values suitable for top quality applications. Thus the more common front-end arrangements will serve with the "Ultra-Linear" 6V6 amplifier while preserving an adequate proportion of feedback. When the 12AX7 is used, the designer has an additional 3 or 4 db of latitude in his choice of gain *versus* proportion of feedback.

In some amplifiers which are of the public-address type rather than the high-fidelity type, inadequate feedback is used which is limited to 6 db or less. Conversion of these amplifiers with the increased feedback which results from a 5 to 1 resistor proportion will produce insufficient gain. In those cases, there must be either a sacrifice of feedback or the addition of more gain in the early stages. However, in these amplifiers the original quality is generally so poor that the substitution of the "Ultra-Linear" output arrangement will make a decided improvement in performance even if only 6 db of feedback is used. The *relative* improvement in a low grade amplifier is even greater than is achieved by converting a fairly good amplifier which has a high proportion of feedback.

When feedback in excess of 12 db

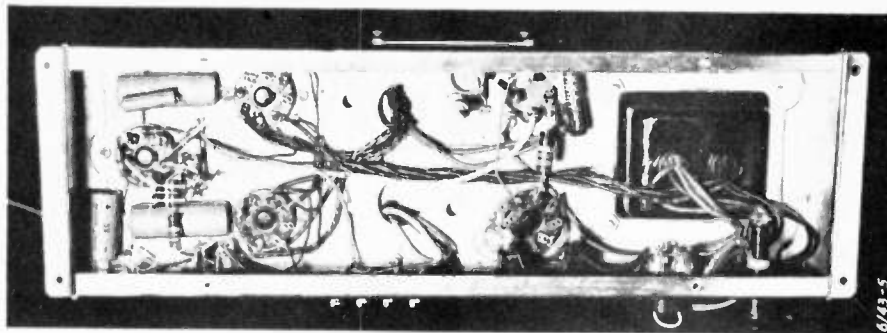


Fig. 5. Bottom view of converted Grommes amplifier showing new output transformer.

is used, there is some possibility that the amplifier response will peak in the ultrasonic region even though the response without feedback is flat over a very wide range. This peaking can be eliminated with a consequent improvement in transient response, by adding a network to change the phase of the feedback voltage in the peaking region. One simple arrangement is to add a small condenser across the feedback resistor. A suitable condenser value in the type of circuit under discussion is one which makes the product of the feedback resistor in ohms and the condenser in microfarads equal to unity. Several typical circuits using a 5 to 1 resistor proportion are illustrated in Fig. 3. In these arrangements, the feedback connection is brought to the cathode or pair of cathodes of the phase inverter stage. All of the arrangements have the same proportion of feedback and the identical phase correction.

Circuit Conversion

These conversion considerations are exemplified in the conversion of the Grommes 100BA amplifier, Fig. 1, the circuit of which is shown converted in Fig. 4. This amplifier is typical of many which come both with and without preamps in the \$40 to \$60 price bracket.

The converted amplifier has extraordinary specifications for its size and price. In fact its specs read amazingly like those of a conventional triode Williamson amplifier. Frequency response is flat $\pm .5$ db from 20 cps to over 100 kc. at a 1 watt level. (By increasing the size of the cathode condenser of the 6SL7 the low end response can be made flat to below 5 cps.) At 10 watts, response is flat ± 1 db from 20 cps to over 60 kc., and clean waveform is preserved from 20 cps to 30 kc. even at this high a level.

The transient response as evaluated

by square waves is shown in Fig. 6. There is a minimum of transient distortion and phase shift at these two extremes of the audio band.

Intermodulation distortion is extremely low. It runs about .1% at 1 watt, rises to .4% at 8 watts, and to .5% at 10 watts. It is still below 1% at 11 watts. These tests were made with 40 and 7000 cps mixed 4 to 1 and are based on equivalent sine-wave output. This is the conventional method of rating which is used for practically all commercial amplifier equipment.

The quality of a low cost 6V6 amplifier is normally not up to the top high-fidelity standards which have been set by the Williamson-type amplifiers produced in recent years. However, it is now possible, by using the "Ultra-Linear" circuit arrangement and a top quality output transformer, to convert these run-of-the-mill amplifiers into ones whose quality is comparable with the best obtainable in the 10 to 15 watt power range. For many people this power range is ample for all home requirements.

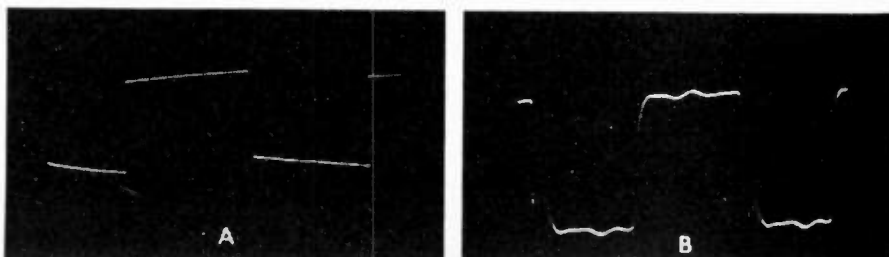
Careful listening tests have borne out the justification for the "Ultra-Linear" conversion. Particularly in the low frequency range there is substantial improvement. The solidness and clarity of the heavy bass passages is a revelation when one contrasts old and new amplifiers. The silkiness and smoothness of the treble range also stand out in a side-by-side comparison. In short, the improvement in measured characteristics is confirmed and substantiated by a corresponding improvement in listenability.

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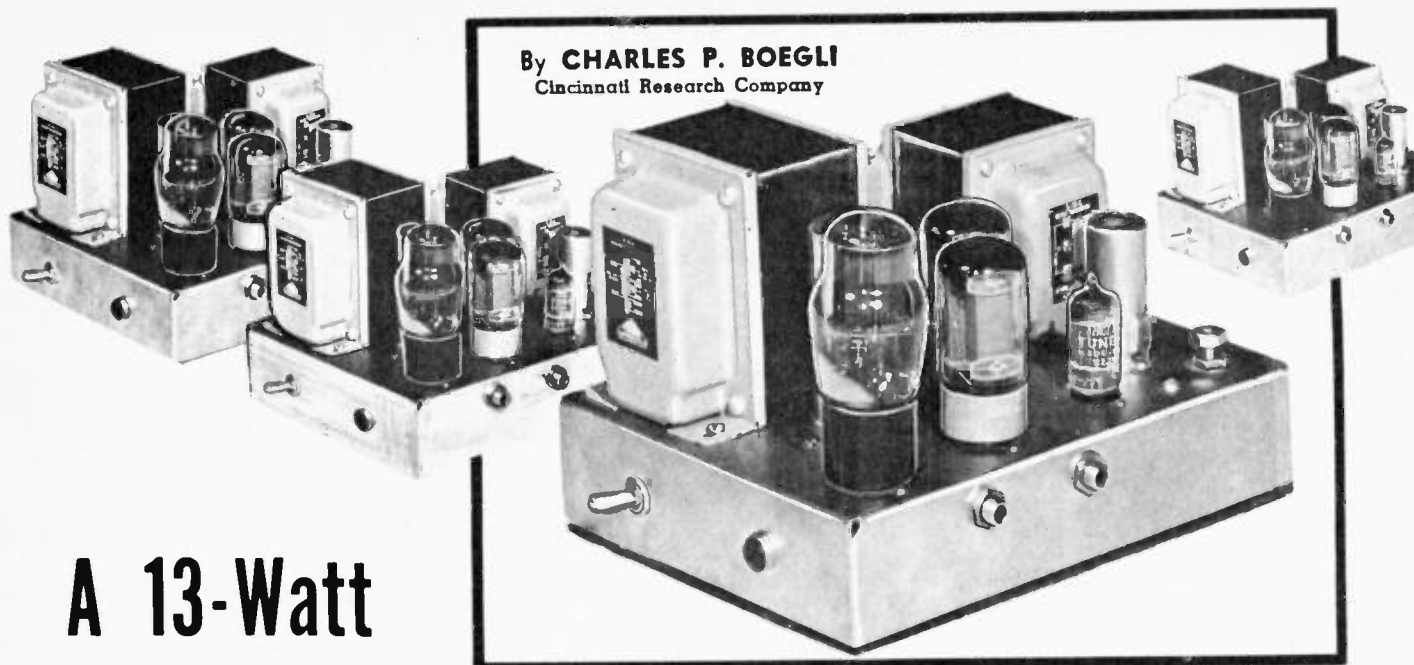
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Fig. 6. Square-wave performance (A) at 20 cps and (B) at 20 kc. See text for details.



By CHARLES P. BOEGLI
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A 13-Watt

All-Triode "Infinite Feedback" Amplifier

Construction details on a well-designed, all-triode unit which is right size for the average home audio system.

IN ONE of the author's previous articles¹ he described a 35-watt amplifier with a novel combination of features, namely 100% negative feedback around the output transformer and the output and driver stages, together with sufficient positive feedback around the driver stage to cause it to oscillate in the absence of the negative feedback. That article presented the mathematical basis of the design and showed that the arrangement can lead to a very stable amplifier having extremely low distortion and approximately zero output impedance. These principles were applied to the design of an amplifier using class-A push-pull 300B tubes in the output stage and the unit was found to develop 37 watts at 2% (r.m.s. sum) IM distortion, or 35 watts at a distortion limit of 1%.

Two features of this amplifier make it somewhat unsuited for average home use. The output power of 37 watts is far in excess of that required in most homes, and the low input impedance of the amplifier necessitates a special final stage in the preamplifier with which it is to be used. Need seems to exist for a smaller amplifier which, while retaining the same circuit features, will have a maximum output on the order of ten watts, sufficient for home systems utilizing all but the most inefficient speaker systems. That the input impedance should be sufficiently high to permit the use of most available preamplifiers is a further requirement.

These objectives have been accom-

plished in the circuit to be described here. At the same time, because every part of the circuit has been designed with a view toward economy of construction, the new amplifier can be made at a cost somewhat less than comparable *Williamson* units. That no sacrifice in performance has been made to economy should be strongly emphasized; nevertheless, the final circuit does not contain a single unnecessary component.

Design

The circuit diagram of the complete amplifier is shown in Fig. 1. For the output stage, tubes of the class including 1614, 5881, KT-66, and 350B were selected. Not only are such tubes widely available but their characteristics are so similar that they may be used interchangeably with no modification of the remainder of the circuit. All of them have a maximum plate dissipation on the order of 26 watts when connected as triodes (with the exception of the 350B, for which the dissipation is 34 watts). The greatest power is developed by operating them near their maximum ratings. A control is incorporated into the stage for balancing the plate currents of the output tubes, but this adjustment has a very small effect on hum level.

The output transformer is a *Triad S-35A*, a reasonably-priced component of exceptional characteristics. In this circuit the full output power of 13 watts is available from less than 16 to over 30,000 cps. The entire circuit

has been designed around this transformer; preliminary calculations² seem to indicate that a wider frequency response is of no benefit whatever in audio amplifiers. As in the earlier 35-watt amplifier, the secondary of the output transformer is connected in balanced fashion, with the 0 and 16-ohm taps attached to the 16-ohm speaker and the 4-ohm tap grounded through the driver bias resistor paralleled with a small bypass capacitor. The speaker lines are at a small positive d.c. potential and must not be grounded to the chassis in any manner.

Various output stages were tested in the experimental work, including straight pentode and "Ultra-Linear" arrangements. Both these circuits increased the available power output to about 18 watts. Both appeared to have the common property that while the distortion level was quite low if the amplifier was connected to a load of the correct impedance, it rose objectionably as the load impedance was reduced below the correct value. This tendency was not observed with the triode connection; a reduced load resistance lowered the maximum power output but the distortion at lower levels increased only slightly. The "Ultra-Linear" output stage requires an output transformer considerably more expensive than the one indicated in this article and the nominal increase of output power does not seem to warrant this extra cost.

The output stage is thus quite conventional; the bypass capacitor across the cathode resistor common to the output tubes has been eliminated in the interest of economy with no measurable adverse effect upon performance.

A neon bulb shunting the grids of the output tubes limits the input voltage and prevents destruction of the tubes in the event the speaker leads are accidentally shorted together. (Shorting the speaker leads together effectively removes the negative feedback and permits the drivers to oscillate.) Tubes like the 5881 are considerably more resistant to this type of abuse than the 300B's used in the larger amplifier but they are nonetheless ruined in less than a minute by shorting the speaker lines in the absence of the neon-bulb limiter.

The push-pull driver is a single 12AU7, around which sufficient positive feedback is fed to produce oscillation in the absence of the negative feedback. The cathodes of the 12AU7 are connected directly to the secondary of the output transformer, providing 100% negative feedback, preventing oscillation of the drivers, and indeed, resulting in an extremely stable amplifier. Output tubes of the heater-cathode type bring up several problems not encountered with the filament type used in the larger amplifier. When early models of the present unit were turned on, the driver tubes heated up much more rapidly than the output tubes and oscillation occurred for a few moments, until the output tubes "caught up." This oscillation was heard as a loud "whoop" in the speaker—frightening to many listeners. A great deal of work went into the de-

sign of the positive-feedback loop to eliminate this effect, and it never occurs in the latest circuit.

The input stage, which is also the inverter, is a dual triode with a large common cathode resistor. Signal is fed into one grid and the other grid is grounded through a capacitor. The outputs from the two plates are very closely balanced and have practically identical internal resistances. The circuit is well balanced and, except for the grid connections, completely symmetrical. 60-cps hum created by the large heater-to-cathode voltage, and 120-cps hum from the power supply appears in-phase at the two plates and is canceled at the transformer. Also, because it is quite degenerative, the input stage introduces a negligible amount of distortion.

Modification for push-pull input is easily made, and consists of reconnecting the first stage so that the two inputs are fed to the two grids. This means that for push-pull input the grid shown grounded in the circuit diagram is removed from ground and connected to the other side of the input. A dual gain control is then required.

During the experimental work on early versions of this amplifier, changing the input tube was found to have a profound effect on the distortion. The resistances inserted between the 12AX7 plates and the driver grids overcome this tendency and allow re-

placement of the 12AX7 with no special precautions as to selection. They also permit this tube to be lightly loaded while at the same time a comparatively low resistance is presented to the 12AU7 grids, improving the high-frequency response. A grounded tube shield will eliminate any tendency of the first stage to pick up hum, but the shield has been found unnecessary in most cases.

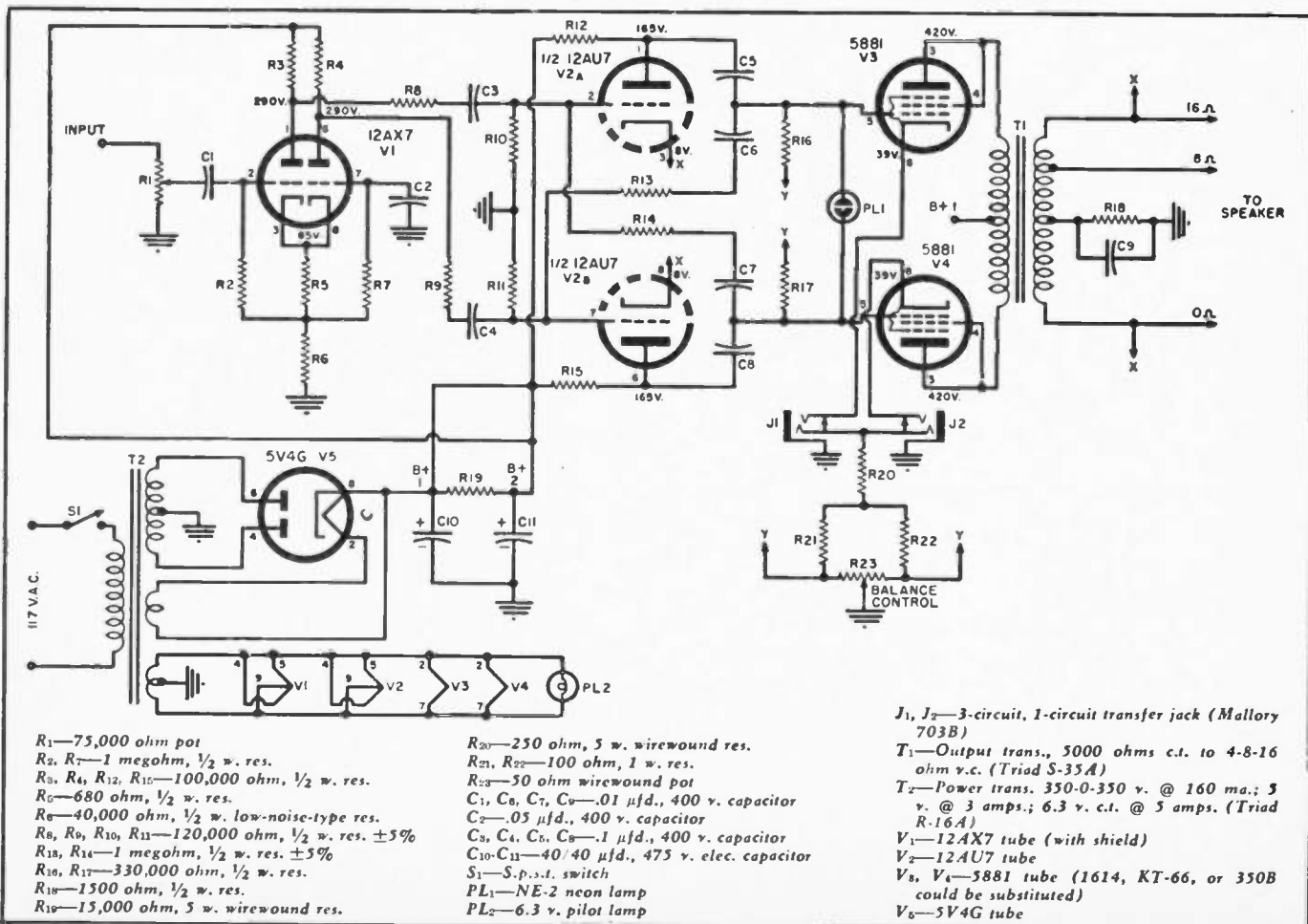
A 75,000-ohm potentiometer is employed as a gain control in the amplifier's input. This resistance is not too low for the great majority of preamplifiers if they have an output capacitor no smaller than 0.25 μ fd. The use of a gain control larger than 100,000 ohms may result in an increase in hum and a loss of high frequencies at certain settings, since the input capacitance of the inverter stage is appreciable.

Because of the symmetry of the amplifier, the power supply can use a minimum of filtering without increasing the hum level. The large amount of feedback in the amplifier helps keep the hum level quite low even with unbalanced output tubes, although badly unbalanced tubes decrease the stability at low frequencies and in extreme cases can cause motorboating.

Construction

The entire amplifier and power supply can be mounted on a 7 by 9 by 2-

Fig. 1. Schematic of 13-watt amplifier. Parts should not be substituted in construction.



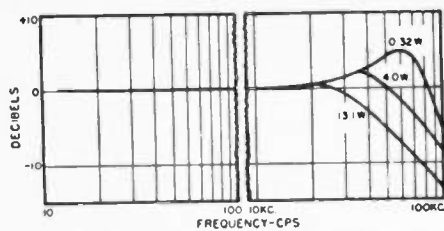


Fig. 2. Frequency response of the 13-watt "infinite feedback" amplifier. See text.

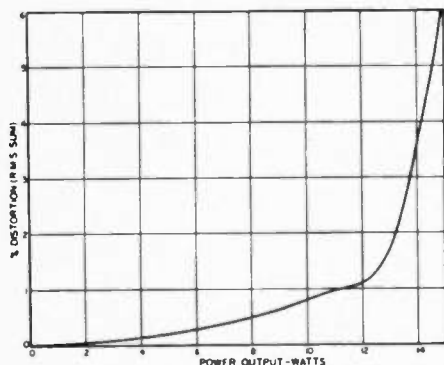


Fig. 3. Intermodulation distortion of the amplifier, 60 and 7000 cps, 4 to 1 ratio.

inch chassis if the layout is done carefully. The necessary compactness is achieved by mounting components directly on the output-tube sockets while using a terminal strip for the first two stages. The photographs show the completed amplifier and an underchassis view which clearly illustrates how the terminal strip is mounted. The terminal strip should be assembled by attaching the resistors first, followed by the capacitors.

Coupling capacitors of the highest quality should be used in constructing the amplifier. The writer has made over a dozen units, and in every case where excessive hum or distortion was encountered the trouble could be traced to a leaky capacitor which was throwing the two halves of the amplifier out of balance. Except for the few 5% resistors indicated on the diagram, no precision or specially selected or matched components are required to assure satisfactory performance.

The constructor must be cautioned not to change the values of any com-

ponents in the circuit or to use tubes other than those specified. Changes very often have an unexpected result; for instance, decreasing the size of the coupling capacitors between the input and driver stages actually causes a rise in the bass response and may lead to motorboating.

Figs. 2 and 3 show the frequency response into a 16-ohm resistive load and the r.m.s. sum intermodulation distortion at various output levels. An input of 0.5 volt is sufficient to drive the amplifier to full output. The unit shows an output resistance of approximately zero ohms over the range of audible frequencies, which forces the most refractory speaker to behave docilely. Although the response is quite flat listeners are often impressed with what seems to be greater bass response than that attainable with other flat amplifiers, and this can be attributed to the high damping factor (by the usual definition, the damping factor is infinite). The improvement is especially noticeable with woofers of low efficiency.

The amplifier is very stable and shows no tendency to motorboat or, unless overdriven, to oscillate at any frequency or output level. A capacitance of 0.1 μ fd. shunted across the speaker terminals does not cause any oscillation or other evidence of instability; this is far in excess of the capacitance presented by any speaker system. The entire amplifier can be wired in a few hours' time and no special precautions as far as lead dress, shielding, or bus bar ground are necessary if the layout of the experimental model is followed. The noise level of several chassis with the inputs shorted varied somewhat with the tubes used but a value greater than 2 mv. = 77 db below 13 watts was never encountered. In all the experimental amplifiers built by the author ground connections were made to any convenient chassis point.

Summary

Since the appearance of the article describing the 35-watt unit several commercial amplifiers have been developed featuring "variable damping

factor." The articles describing these amplifiers^{3, 4} draw various conclusions as to the damping factor desirable for results of the highest quality.

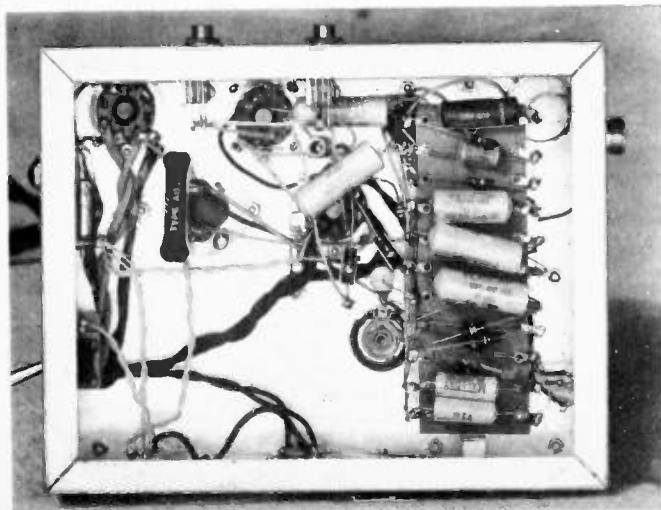
If, instead of the 1-megohm fixed positive-feedback resistors in the amplifier described here, a dual potentiometer was employed, the output resistance of the amplifier can be controlled within small limits. When the amount of positive feedback is reduced the output resistance increases up to a maximum corresponding to a damping factor of 4; if the amount of positive feedback is increased the output resistance becomes negative. As the latter occurs, however, the stability of the amplifier becomes much worse, and a damping factor of -1.0 can certainly not be obtained.

When the positive feedback is varied in either direction from the optimum (sufficient to make the driver oscillate in the absence of negative feedback) the distortion of the amplifier at a given output level increases. The increase is gradual while the positive feedback is being lowered. On the other hand it rises rapidly as the positive feedback is increased beyond the optimum value. For this reason, the use of such a control to obtain negative damping factors is not to be encouraged. *No circuit has yet been devised* which will, at the same time, produce a large negative damping factor and low amplifier distortion⁵. This objection does not apply, of course, to amplifiers in which a control is provided to vary the output resistance between positive limits.

An amplifier of the type described here, unfortunately, will not make a five-dollar speaker in a cardboard box perform like a two-hundred dollar assembly in a folded horn. It will try, of course, and sometimes the effort becomes so strenuous that oscillation occurs. Such oscillations have been touched off by overloading the amplifier during efforts to obtain resounding bass from tiny speaker enclosures. Given the amplifier, the remedy in such cases lies in either of two directions: a better speaker system should be installed or a filter should be inserted before the amplifier to remove the low frequencies which the system cannot reproduce anyway. Used in conjunction with a speaker of comparable quality, the amplifier is capable of results which have, to date, been thoroughly pleasing to well over a hundred persons and displeasing to only two or three.

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Under chassis view of the 13-watt amplifier. Layout and lead dress is non-critical but use of terminal strip simplifies construction.

Control of

AMPLIFIER DAMPING FACTOR



By **DAVID HAFLER**
Acro Products Company

THE damping factor of an amplifier is defined as the ratio of nominal load impedance to actual output impedance. The nominal load impedance is the value of impedance in which the amplifier should be terminated for normal operation. For example, the 16-ohm output on an amplifier should normally be terminated in a 16-ohm speaker or other load. Other values of load impedance will generally degrade performance. However, the actual output impedance of the amplifier is unrelated to this nominal impedance and the 16-ohm output tap of an amplifier could represent any measured impedance from -16 ohms to +160 ohms.

It is simple to measure the output impedance of an amplifier. A signal voltage is introduced at the input and the output voltage is measured with no load on the amplifier (keeping the signal level below the overload point). Then a variable resistor is put across the output and varied until the output voltage has dropped to one-half of its unloaded value. The measured d.c. resistance of the variable resistor which drops the output to half voltage is equal to the output impedance of the amplifier. If the voltage rises when the load is introduced, the output impedance is negative; and a 2 to 1 change in voltage gives the resistor value which is equal to the negative impedance.

Obviously, a zero output impedance cannot be measured by this approach. Therefore, a zero impedance is determined as the condition where the connection of any value of load produces no change in output voltage.

Since the damping factor of an amplifier is equal to the nominal load impedance divided by the output impedance, the damping factor can be changed by controlling the output impedance. Thus, varying the output impedance gives variable damping. The damping factor can be made unity by making the output impedance equal to the load impedance. It can be made infinite by making the output impedance zero. Likewise, it can be made negative by making the output impedance negative.

Is variable damping a "must"? Here are some pros and cons on this currently "hot" audiophile topic.

Until recently the damping factor of an amplifier was an incidental resultant of the design. Triodes without feedback had damping factors in the range of 2 to 4. Tetrode amplifiers had damping factors of 1 to 10 (depending on the amount of feedback used). More recent designs using triodes with feedback or "Ultra-Linear" stages with feedback have had damping factors ranging from 10 to 30. It was generally felt that higher damping factors were more effective than lower ones, but design was aimed more at obtaining low distortion and similar attributes than at achieving a specific degree of damping.

Now, however, the latest fad in amplifier design is to provide means of controlling the damping factor through control of the amplifier's output impedance. Variable damping is appearing on more and more commercial amplifiers, and the advertising claims for it herald it as a tremendous advance and an absolute necessity for the audio enthusiast. Amazingly, these claims are inconsistent since some recommend high damping factors, others lower ones; and even the negative damping factor is extolled. It is well worth while examining the reasons for variable damping, the means by which it is done, and its results. In this way, perhaps, the role of variable damping in amplifier design will be better understood.

Why Variable Damping?

Even though variable damping is a feature of amplifier design, its function has nothing to do with amplifier performance. Variable damping is introduced for the purpose of obtaining better loudspeaker performance. It is widely appreciated that the performance of a loudspeaker is influenced by the impedance of the source from which it is driven. Variable damping

makes it possible to optimize the source for any given loudspeaker.

Unfortunately, it is difficult to determine what comprises the proper source impedance for a loudspeaker. There are three basic schools of thought on this subject, and their opinions are incompatible and contradictory.

School A claims that a speaker should be *critically damped*. Depending on the speaker system being used, this is generally attained when the speaker is almost matched to the amplifier and the damping factor is approximately 1 or 2. A range of variable damping from 1 to 10 would take care of almost all systems if critical damping were the only consideration.

School B claims that the speaker should be matched in impedance at frequency extremes. Most loudspeakers exhibit a substantial rise in impedance at low and high frequencies. If a constant voltage amplifier, one with a zero output impedance, were used, the power into the speaker would decrease (because it takes increased voltage to maintain constant power across an increased impedance). Conversely, a high impedance source, which would match the speaker impedance at high and low frequencies, would make for flatter power output. It is necessary to get output impedances as high as 10 times the nominal impedance (damping factor of .1) to follow the practices of this school.

School C believes in the need for an infinite damping factor, or at least as high a damping factor as possible, obtained by a source impedance which approaches zero. The reasoning behind this school of thought is that a zero impedance will short circuit the back e.m.f. due to spurious speaker motions and thus produce cone motions more closely following the amplifier output. This of course would provide less distortion and superior transient response

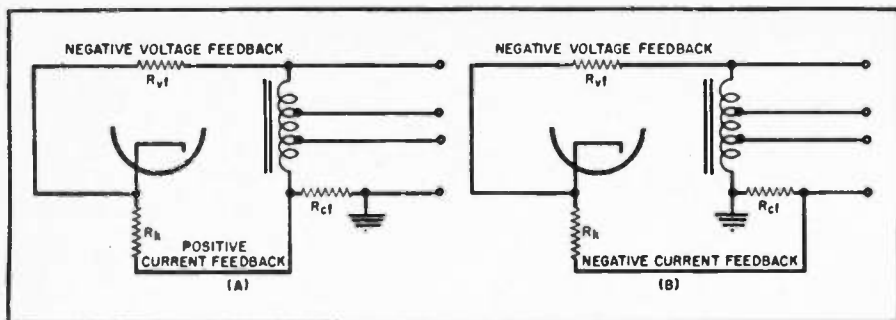


Fig. 1. The use of current feedback for damping factor control. Refer to article.

as well as making the output of the amplifier independent of impedance variations in the speaker.

A subgroup of "School C" believes in carrying the output impedance into the negative region to the point where the d.c. resistance of the speaker voice coil is cancelled out. In this way the total circuit impedance, including amplifier and speaker, is approximately zero; and the speaker cone is rigidly coupled to the amplifier. This represents the ultimate in damping, past which one cannot go.

How It Is Done

Variable damping is accomplished through the manipulation of feedback around the output stage. Normally, a high grade power amplifier has negative voltage feedback which lowers its output impedance. It is also possible to increase the output impedance by using positive voltage feedback, but this is basically an unstable mode of operation. It is practical, however, to use current feedback; and the effect of current feedback on output impedance is inverse to that of voltage feedback—positive current feedback decreases output impedance, while negative current feedback increases it. It is useful, therefore, to combine voltage and current feedback to obtain a wide range of impedance control.

Fig. 1 illustrates how voltage and current feedback can be combined to obtain any desired output impedance and damping factor. In Fig. 1A negative voltage feedback is combined with positive current feedback to lower the output impedance and to increase the damping factor. In Fig. 1B the combination of negative voltage and negative current feedback increases the output impedance and reduces the damping factor.

In each case, R_k is the cathode resistor of the stage to which feedback is taken. R_{vf} and R_k form a voltage divider which controls the proportion of negative voltage feedback. R_{cf} is a resistor in series with the load. The current through the load and through R_{cf} produces a voltage across R_{cf} which is fed back to furnish current proportional feedback. R_{cf} must be made small or too much of the load power will be dissipated in it. Because it is small, it must be introduced in series with R_k or else its shunting effect would change operating conditions of the stage biased by R_k .

The larger R_{cf} is, the more current

feedback there is. Also, changes in the load will produce current changes in R_{cf} and change in the current feedback. Therefore, such changes as shifting to a speaker of different output impedance brings about a change in output impedance and a corresponding change in damping factor, because of the change in the ratio of voltage and current feedback.

For those who are interested in experimenting with variable damping, it can readily be added to an "Ultra-Linear" Williamson-type circuit by using a .5 ohm rheostat for R_{cf} . This can consist of a 1-ohm resistor and a 1-ohm rheostat or potentiometer in parallel. If a wirewound control without a parallel resistor is used, poor contact at some points of rotation of the slider arm would make the effective resistance increase and cause big changes in current feedback. This effect is minimized by having a fixed resistor in parallel.

The circuit gives an approximate range of control of output impedances (on the 16-ohm nominal output) from -12 ohms to +1 ohm if the current feedback is positive and +1 ohm to +15 ohms if the current feedback is negative. The total possible damping factor variation is from about -1.3 to +1 and including infinity in this range. If a loudspeaker load is connected to the amplifier, its impedance variations might cause even more current feedback, thus extending the range of control. Unfortunately, large proportions of current feedback may cause instability and oscillations. The experimenter is warned that a wide-band a.c. v.t.v.m. or scope should be kept connected across the amplifier output when adjusting the damping in order to avoid instability which could damage the speaker system should too much current feedback be used. In particular, the use of positive current feedback can easily lead to instability irrespective of the amount of voltage feedback. Negative current feedback adds to the total negative feedback; and if instability is a problem, a reduction of the negative voltage feedback can be made (by doubling the value of R_{cf} , for example) to keep the total feedback within the range of satisfactory stability. Many circuits use ganged controls to vary both R_{cf} and R_{vf} simultaneously so as not to change the total amount of negative feedback. For the purpose of this article it was felt that such variants are of minor

pertinence; and, therefore, they are not discussed. Of far greater importance are the end results of using current feedback to vary the damping of the amplifier.

Effects of Variable Damping

The use of current feedback for damping factor control influences the performance of both amplifier and loudspeaker. The effect on amplifier performance is generally ignored in presentation of information on variable damping because the effects on speaker performance are more obvious. However, some mention of what happens to amplifier performance is justified since we are interested in the over-all amplifier-speaker combination rather than one alone.

1. *Amplifier performance:* Irrespective of whether feedback is of the voltage or current proportional type, it influences the amplifier distortion. Therefore, the addition of positive current feedback to an amplifier will increase its distortion; while adding negative current feedback will reduce the amplifier distortion.

As mentioned, the use of positive current feedback will lead to instability if the output impedance is made too negative. Instability can also arise when too much negative current feedback is added to the amplifier. These problems appear superficially unimportant because the amplifier can always be checked for stability before it is put into service. Unfortunately, however, there is no certainty that laboratory stability will mean stability under home listening conditions.

The reason for this is that the current feedback varies when the load impedance is changed. Connection of a loudspeaker will give a different proportion of current feedback than will be obtained with a resistor. Connection of a multiple speaker system with crossover network will cause drastic changes in feedback at the crossover frequency where impedance changes always occur.

Even if variations in load impedance do not cause instability, they cause changes in frequency response. Obviously, when the feedback changes, the gain changes; and if this is a different effect at different frequencies (as happens on complex loads), then there is frequency discrimination.

2. *Loudspeaker performance:* When current through the load is fed back through the amplifier, any non-linearities in load current are applied again to the load as part of the driving signal. Thus if a speaker has non-linear voice coil excursion, a non-linear driving signal will be applied to the speaker when current feedback is being used. This signal may either correct for the original non-linearity or it may add to it, depending on phase relationships. It has been claimed that positive current feedback provides a phase relationship which reduces loudspeaker distortion by this type of cancellation of some of the distortion components. However, as shown in Table 1, there

is no clear-cut reduction in distortion as the damping is increased, nor is there much difference in distortion when the damping is decreased with current feedback. Apparently, the effect of variable damping on distortion is dependent on the type of speaker used, its baffling, and similar variables which make it difficult to generalize.

One experiment which can be readily attempted with limited equipment indicates that under some conditions positive current feedback *increases* speaker distortion while negative current feedback *decreases* it. When a signal is fed into the loudspeaker, smother its output by putting a heavy cardboard across the orifice of the baffle. This places an air load on the cone which changes the linearity of the voice coil motion. A corrective signal should be in such phase that the amplifier delivers more output and pushes the speaker harder to overcome the smothering. Either more positive feedback or less negative feedback would furnish the correct compensating signal.

At most frequencies where this experiment is tried, the speaker impedance increases, the current through R_{ef} is decreased, and the current feedback is decreased. Positive current feedback causes a reduction in gain and does not correct for the smothering effect; while negative current feedback causes the speaker to be driven harder, thus correcting the effect. At the bass resonance frequency, however, the speaker impedance is decreased by the extra air load; and the correction effect is reversed. Therefore, this particular type of speaker non-linearity is affected differently by different types of current feedback depending on the frequencies at which testing is done. It is probably possible to pick frequencies and test conditions which can tip the scales in any direction desired by the experimenter.

Although the effects of different damping factors on speaker distortion are not conclusive, the effects on frequency response are quite certain. The response of the speaker-amplifier combination increases with increases in impedance when the damping factor is low (and source impedance is high), and decreases with increases in impedance when the damping factor is high (and source impedance low). The response follows the impedance curve with low damping and is inverse to the impedance curve with high damping. Which is the more desirable response curve?

Evidently, if speakers are designed to operate with a low-impedance source, this is the best condition to use. If the speaker manufacturer sets his response specifications by observing a fixed voltage across the voice coil at various frequencies, the amplifier with zero source impedance (infinite damping factor) would duplicate the manufacturer's test conditions. In this case a higher source impedance would cause response peaks at impedance peaks, such as the bass resonant

Frequency	Amplifier Impedance —5 ohms D.F. = —3.2	Amplifier Impedance 0 ohms D.F. = ∞	Amplifier Impedance 50 ohms D.F. = .32
50 cps	15 %	20 %	18 %
100 cps	8.2%	5.2%	4.8%
500 cps	.8%	2.3%	3.0%
5000 cps	2.0%	1.8%	1.9%

Table 1. Loudspeaker harmonic distortion for various damping factors. Measurements made at absolute sound pressure at 500 cps of 94 db. Sound pressures at other frequencies are obtainable by reference to the response curves shown in Fig. 2 below.

frequency; while a negative source impedance would cause a loss in response at impedance peaks. The correct response curve for a speaker will be obtained only if the speaker is operated as intended by its manufacturer.

Fig. 2 shows the response curve of a 12-inch loudspeaker in the medium price category (near \$30.00). These curves, taken with various damping factors, show that the frequency response is intimately related to the source impedance. With high damping factors (low source impedance), there is a definite loss of bass and treble response. Experiments were also carried out with better quality speakers, and it appears that the effects of different damping factors are diminished with better grades of loudspeakers.

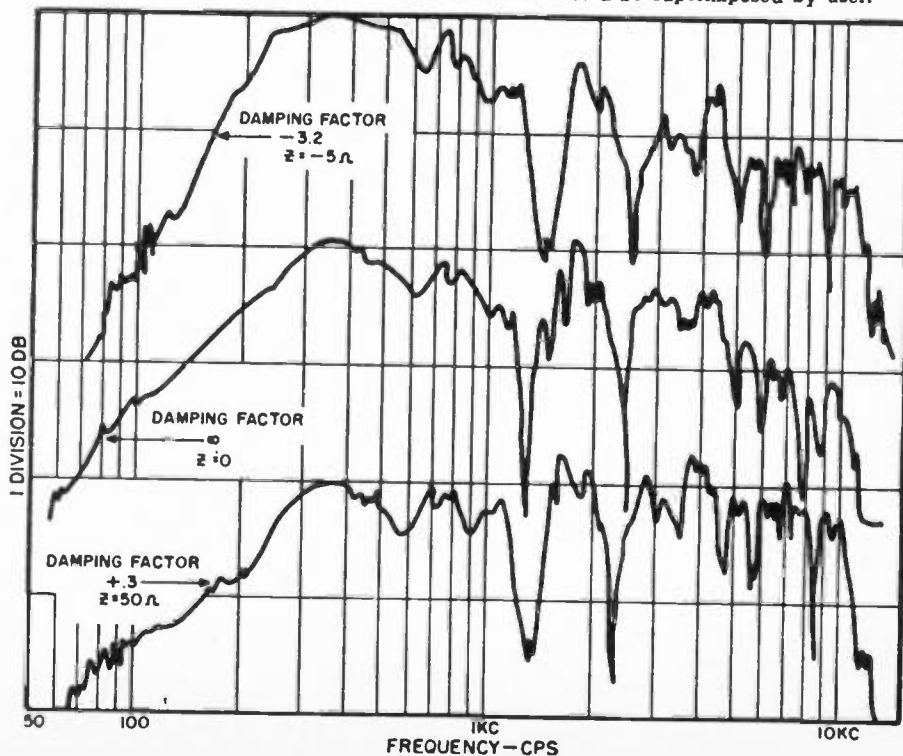
The higher the quality of the loudspeaker system (including baffle), the smoother and less variable is the impedance characteristic of the system. With less impedance variation in the speaker system, there will be less changes in frequency response as the damping is changed.

The same situation holds for speaker damping. Better grades of speakers with more efficient and larger magnet structures will generally be critically damped with a damping factor in the

range of 1 to 4. Low cost, inefficient speakers may have so much d.c. resistance that a negative source impedance is necessary to bring the total circuit impedance in the range where the back e.m.f. generated by the voice coil is effectively short circuited. Therefore, except with the poorest types of systems, moderately low source impedances will supply sufficient damping to nullify overshoot and boom which are spuriously generated by underdamped systems. Damping factor control over the range of 1 to 10 would cover the possibility of obtaining critical damping of most better grade speaker systems. Increases in damping factor past the condition of critical damping will have practically no effect on the damping of the system. The impedance contributed by the amplifier after passing a damping factor of 10 is so small compared to that contributed by the speaker that damping is unaffected by further reduction of amplifier source impedance.

Thus, after eliminating poor grade speakers, it appears that any damping factor of 10 or more will serve to provide satisfactory speaker damping. However, speaker response will depend on the damping factor used. The best source impedance for a particular

Fig. 2. Frequency response with variation in damping factor. For a more accurate comparison of these responses, these three curves should be superimposed by user.



speaker must be specified by the manufacturer of that speaker, as the means for proper adjustment of frequency response through variable damping are beyond the capacity of any but a well equipped acoustics laboratory.

The author has questioned several speaker manufacturers and has found them amazingly (and amusingly) reluctant to comment on how their speakers should be operated. The consensus seems to be that modern speakers are expected to work properly with amplifiers of low positive source impedance (damping factors of 10 or more), and the manufacturers' specifications are set from a constant voltage source—the equivalent of a zero impedance source. Any departures from this conventional standard should be mentioned by the manufacturer in his directions for using the speaker.

Conclusion

Weighing the various pros and cons of variable damping, the author finds it difficult to justify variable damping except in the limited case of low grade speakers or situations where the speaker manufacturer intended a specific source impedance other than close to zero.

Speaker distortion may be affected

for better or for worse through variable damping. Frequency response will generally be most suitable for a low source impedance. Therefore, a fixed, low source impedance, such as is normally obtained from amplifiers with appreciable negative voltage feedback, will provide close to optimum performance as well as insuring that the speaker is critically damped (or slightly overdamped).

On the other hand, the use of current feedback to provide variable damping introduces new problems which tend to degrade listening quality. When a multiple speaker system is used with a crossover network, the impedance-sensitivity of current feedback will cause frequency variations which cause rough and ragged reproduction at the crossover frequencies. If no crossover is used in a multiple system, the impedance variations in one part of the system cause current feedback changes which are reflected in changes in signal level to other units in the system. This again accentuates raggedness of frequency response.

The most serious drawback to the use of variable current feedback lies in the dangers of instability. This is particularly true where the feedback is positive. In order to minimize this difficulty, some designs introduce fil-

ters to confine the positive feedback to a limited frequency spectrum. However, this causes frequency unbalance similar to tone control action.

Listening tests under reasonably well controlled conditions indicate that, as theoretically expected, high output impedances lead to boomy and screechy sound quality, while negative impedances lead to a loss of bass. Extremes in either direction lead to veiled and indistinct sound quality. Undoubtedly, other experiments using different equipment could lead to different conclusions, but so far the author has found nothing to justify variable damping, while many factors indicate that it is undesirable.

EDITOR'S NOTE: One thought that should not be overlooked is that many manufacturers have been including some form of variable damping in their new amplifiers at no additional cost to the consumer. Many individuals may find it quite interesting to experiment with this feature. In all cases, it can be cut out of the circuit if not desired.

The measurements shown in Table 1 and Fig. 2 were made by Mr. Bruce DePalma, of M.I.T. Mr. DePalma also contributed many ideas on the subject of variable damping during the course of the discussions and tests on which this article are based.

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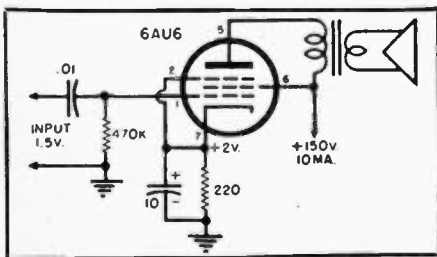
HIGH-EFFICIENCY, LOW-POWER AUDIO STAGE

By LAWRENCE FLEMING

MOBILE receivers and intercoms are among the items that require only $\frac{1}{3}$ -watt or so of audio output to the speaker, but where high gain and low power drain are important. A quarter of a watt of audio is actually enough for a living room, except for peaks, where the listening is at ordinary conversational levels.

Regular power pentodes like the 6AQ5 require a lot of plate power, and

Fig. 1. The high-efficiency, sensitive audio output stage for intercoms and mobile gear. Output is .36 watt with 1.5 volts input and only 10 milliamperes power drain.



their power sensitivity is not particularly high because they were designed primarily for high output. The most sensitive power pentode, the 6BK5, gets its high gain by using a large cathode, and the heater alone draws 7.5 watts. To get high sensitivity at low power input, what is needed is a tube having high transconductance per ma. of plate current, and a reasonable-sized heater. Voltage amplifier pentodes, such as the 6AU6, fill these requirements.

It turns out that a 6AU6 makes a junior-sized output tube that is more sensitive than any except the 6BK5, and at a fraction of the power drain on the supply. It has, moreover, a much higher power sensitivity per watt of drain from the supply than any of the audio power pentodes.

Fig. 1 shows a typical circuit. The load impedance should be about 20,000 ohms. Changing it to 15,000 or 25,000 ohms will not make much difference. A good output transformer is one rated for battery-type power tubes, such as the Stancor A3327 or Knight 62G94.

If the plate supply voltage is higher than 150, use a dropping resistor in se-

ries with the screen of the 6AU6 and bypass the screen to the cathode with a 10 μ fd., 150-volt electrolytic, otherwise the screen will overheat. For a 300-volt supply, the correct resistance is 56,000 ohms and for a 250-volt supply, 33,000 ohms. For all supply voltages the 220-ohm cathode resistor is best, and the plate and screen currents are 7 and 3 ma. respectively.

At 1.5 volts audio input, the 6AU6 so used will put out 360 milliwatts of audio. In comparison, a 6AS5 at the same 1.5 volt signal supplies only 140 milliwatts and a 6AK6 only 65 milliwatts. With a 300-volt supply and the necessary screen dropping resistor, 6AU6 output is about 500 milliwatts at overload.

While many other small pentode types such as the 6CB6 and 6AU6 have even higher transconductance per ma., they are not satisfactory for class A power service because their screen current rises too high during large signals. In general the operating conditions for maximum power output in class A are low grid bias together with the highest screen voltage permissible within the rating of the tube.

-30-

HI-FI AMPLIFIER

By N. H. CROWHURST

WHEN you ask a man working on an audio amplifier what seems to be the difficulty, nine times out of ten the trouble turns out to be some form of instability. Even people with a better-than-average understanding of audio can be stumped by some of the perverse things that happen in this category. For this reason the author makes no apology for introducing another discussion of this subject.

The basic cause of all kinds of instability is positive feedback. It is important to bear this in mind, because it can often help in locating the exact spot where the instability occurs or in determining the cure.

Usually the instability occurs at either end of the audio band. When it occurs at the low frequency end it results in what is commonly termed "motorboating." Instability at the high-frequency end can result in high-frequency oscillation, usually at a frequency beyond the audio range. This will have the effect of blocking the signal, giving it an unnatural sound. This is caused by the high-frequency oscillation fully loading the amplifier. Such loading is not obvious because it is inaudible. However, when the audio signal comes along, its application, on top of the oscillation of the amplifier, temporarily reduces the amplitude of oscillation and allows some of the audio to get through.

Another form of instability occurs when the high-frequency condition is not quite sufficient to cause continuous oscillation but approaches it and sets off a damped oscillation at certain points in the audio waveform. This form of instability is called "parasitic oscillation."

Another form of instability that can occur gives rise to a terrific screech from the amplifier and this is due to positive feedback in the audio range which causes the amplifier to oscillate wildly at a random frequency. No doubt most of our readers are quite familiar with these forms of trouble. The problem is to track them down and stop them.

Motorboating

Motorboating is due to positive feedback at a low frequency. Usually the feedback is such that it does not produce a smooth sinusoidal waveform at the frequency of oscillation, but gives rise to a series of plops, from which

Instability

Should your audio equipment develop some form of instability—here are the answers on possible sources of trouble and their remedies.

the name "motorboating" was derived. It may be that the "B+" voltage at a certain point in the supply system gradually rises, or gradually falls, until a certain point is reached. When this occurs some low-frequency triggering action suddenly changes the current drain produced by the amplifier on the supply circuit, so as to return the voltage to the point from which it started, whence it slowly rises or falls, as the case may be, until the triggering action repeats itself. Although this complicates the shape of the waveform somewhat, the principle is still the same in that it can be regarded, broadly speaking, as a low-frequency positive feedback.

Sometimes it is helpful to first determine which way the pulses are going and find out just where such a triggering action could occur, by anal-

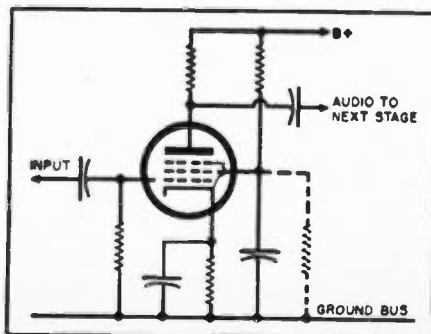
ogy with a multivibrator circuit. It is always helpful in tracking down the trouble to apply a d.c. voltmeter to various points in the circuit and watch for changes in voltage coincident with the plops.

When the point at which the change of voltage is greater than at other points is determined, this is a good place to start looking for the positive feedback region. It may be that there is insufficient decoupling to a screen or plate supply of some stage, according to the particular circuit used.

For example take the circuit of Fig. 1. The center of trouble may be this pentode stage. Although it would not oscillate by itself, the time constants of the screen decoupling capacitor and the plate supply decoupling capacitor may be such that, in combination with the rest of the amplifier, a low-frequency pulse can be fed around the circuit. Making the previously-mentioned voltage checks may show that the greatest change in supply voltage when the plop occurs appears at the screen of this tube. This indicates that the time constant of this circuit should be changed in order to avoid this condition.

If the low-frequency response of the amplifier is better than it need be, the use of smaller coupling capacitors might clear the trouble, but if the full low-frequency response is required, it may be possible to remedy the trouble by increasing the decoupling values in this particular part of the circuit, or it may be possible to make the circuit stable merely by changing the relative values of the resistors and capacitors in the decoupling circuit.

Fig. 1. This high-gain stage circuit can develop instability, either motorboating or h.f. blocking. The cure rests in adjusting the plate and screen feed resistor values or altering the screen feed to a voltage divider system (shown dotted).



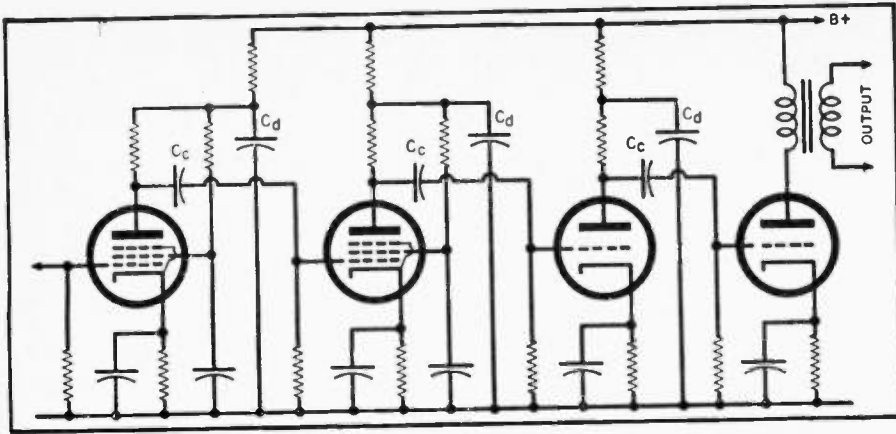


Fig. 2. This kind of high-gain audio circuit can cause motorboating troubles. Attention to the values of C_c and C_d can sometimes cure this particular circuit fault.

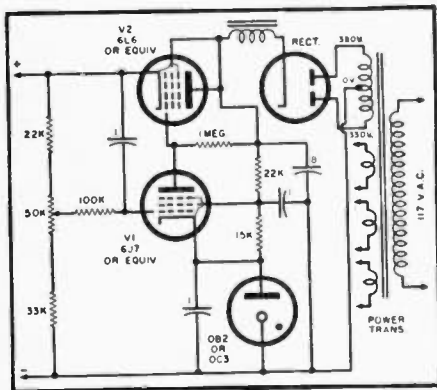


Fig. 3. Where other methods do not help, a stabilized power supply providing low internal-resistance "B+" supply, using this circuit, will usually effect a cure.

Sometimes it is advantageous to use a smaller value of screen decoupling capacitor or perhaps a lower value of screen dropping resistor together with a resistor in shunt with the capacitor in order to provide voltage-divider feed to the screen instead of series feed. These are just suggestions that have proved helpful in curing this particular kind of instability and which can readily be tried.

Sometimes, however, checking with a voltmeter seems to show that the fluctuation occurs all around the circuit and only to a slight degree. This means that the positive feedback is probably taking place throughout the amplifier as a complete entity and can-

not be blamed on one particular tube or set of components.

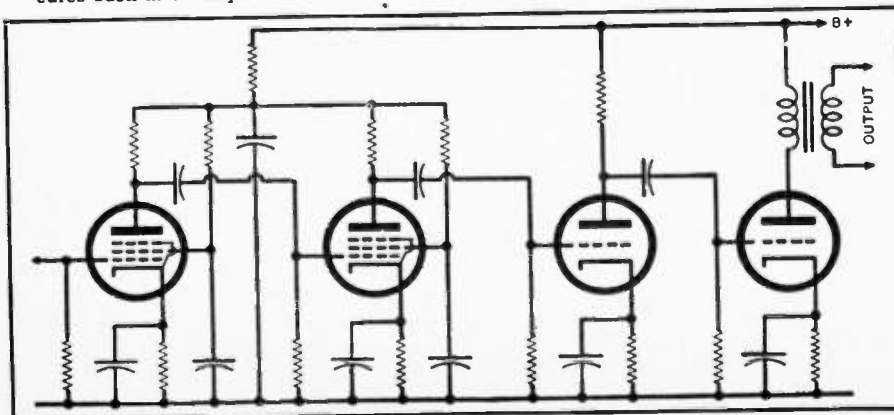
Fig. 2 shows a typical circuit which is subject to over-all motorboating. In this case the simplest measure would be to reduce the values of coupling capacitors C_c or else to increase the values of decoupling capacitors C_d .

However, sometimes it happens that either of these changes merely alters the frequency of motorboating, but does not stop it. This often happens where there is a large amount of gain in the amplifier. Here the only remedy is to pay attention to the impedance of the "B+" supply unit because the motorboating is probably due to the coupling impedance provided by the supply unit.

Decoupling of a slightly different type, as shown in Fig. 4, can sometimes achieve satisfactory results without reducing the supply impedance. The important point in rearranging decoupling, over a multi-stage amplifier, is that two consecutive stages cannot, by themselves, introduce instability. The instability always occurs over three or more stages. For this reason decoupling of stages in pairs conserves parts and offers more stability than providing separate decoupling to each stage along the line.

If this step is not successful, a stabilized power supply utilizing the circuit of Fig. 3 or a similar arrangement can prove helpful. This uses two tubes to reduce the source impedance

Fig. 4. Changing the method of decoupling used in Fig. 2 as shown here, sometimes cures such instability. This method also has the advantage of conserving components.



of the supply. In fact, the source impedance can be reduced very close to zero by careful adjustment of circuit values.

Adjusting the values of the resistors controlling the screen voltage of V_1 will change the source impedance of the supply and can even make it negative. V_1 should be a high slope pentode, such as a 6J7, while V_2 should be a triode-connected tube capable of passing the required "B+" drain. A 6L6 is good up to about 150 ma.

High-Frequency Blocking

When the amplifier oscillates at a high frequency, this oscillation, like the low-frequency variety, can occur either in a single tube or in the amplifier as a whole. If the amplifier as a whole is oscillating, then removal of any tube in the amplifier will stop the oscillation. If the oscillation is occurring in part of the amplifier, then removal of tubes not contributing to the oscillation will not stop the oscillation. This is a simple check that can be made both for motorboating and oscillation troubles.

The only point to be remembered is that, particularly in motorboating, such a check is not necessarily conclusive. For example, removing a tube may alter the operating conditions of the remaining tubes, due to the change in current drain, so as to give a misleading result. One of two things may happen: removing the tube may allow the oscillation to continue, although the tube was contributing to the oscillation previously. This can happen because removing the tube changes the circuit voltages and another form of oscillation arises when the tube is removed (it should be noticed in this case that the frequency of oscillation is changed by removing the tube). Alternately, removal of the tube may stop the oscillation although the tube was not contributing to it previously—again because the operating conditions of the oscillating tubes are changed by the removal of this tube.

If scope examination reveals that the oscillation is occurring throughout the amplifier, there are two things to look for: (1) the placement of hot leads so high-potential leads from the output of the amplifier can feed back to high-impedance leads near the input of the amplifier; (2) bad ground returns, which introduce common ground resistance into successive stages of the amplifier. If the amplifier is already built and rearrangement of the layout is out of the question, the former trouble can sometimes be corrected by use of shielded leads, especially where such leads are fairly long. Fig. 5 shows one method of ground wiring that avoids instability and other troubles due to bad ground wiring.

If high-frequency oscillation can be traced to a single stage then the cure can best be approached by considering this stage as a single-stage oscillator similar to some kind of r.f. oscillator.

A beam-tetrode, high-gain stage can sometimes go into oscillation because

it acts as a negative resistance. This often occurs if the plate voltage falls considerably below the screen voltage. Changing the value of the screen dropping resistor is usually the cure for this kind of oscillation, so that the tube resumes stable operating conditions. Too high a plate load resistor, in relation to the screen dropping resistor, can cause oscillation.

Usually this kind of oscillation will start up without much warning even when the amplifier may have been operating previously under quite stable conditions. This is because the condition of oscillation itself reduces the plate voltage below the screen voltage and therefore the operating conditions change when oscillation starts. If it is not expedient to reduce the value of the plate coupling resistor, the value of the screen dropping resistor should be increased, or else the screen dropping resistor should be changed to a voltage divider, as mentioned in the section on motorboating.

Sometimes output stages can be responsible for oscillation. In its commonest form this occurs as a kind of oscillation similar to tuned-grid/tuned-plate oscillation in radio applications. The similarity exists in that the grid circuit is tuned by the interelectrode capacitance resonating with the leads and possibly the leakage inductance of the driver transformer. Miller-effect feedback causes oscillation due to the fact that the plate-load impedance at this frequency is basically inductive and hence the circuit behaves similar to a tuned-plate/tuned-grid oscillator at r.f.

One remedy is to damp the r.f. resonant circuits causing the oscillation. Plate blocking resistors will help swamp the inductance of the plate-load impedance at the oscillation frequency, while grid blocking resistors will help damp the resonant circuit in the grid. The method of connection is shown at Fig. 6A.

If the power stage is designed to draw grid current, i.e., it is of the B_2 or AB_2 variety, then a series resistor in the grid will severely restrict the output power available. In this case a shunt resistor is necessary, as shown in Fig. 6B. Whichever method of blocking connection is employed, it is essential that the resistor be connected as close to the tube pins as possible, because the oscillation is occurring at high frequency and the inductance of the tube leads is part of the oscillator circuit, so the resistance must be inserted where it will successfully swamp this lead inductance.

Parasitic Oscillation

This is the final kind of oscillation under the heading of instability. It is sometimes difficult to diagnose, although it is quite easy to see as soon as the output is presented on an oscilloscope. However, the amplifier may perform quite well under normal test conditions; there will be no oscillation in the absence of an audio signal or the frequency response appears to be quite normal or perhaps a measurement of

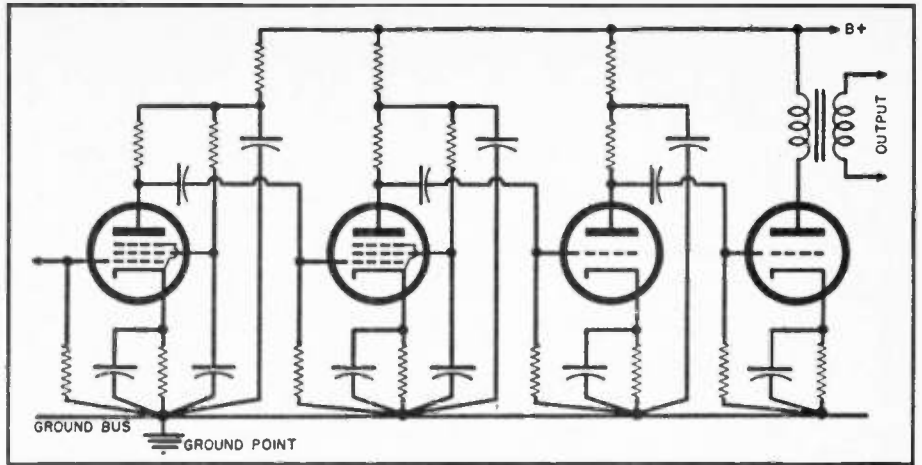


Fig. 5. Poor ground return wiring can cause either type of instability. This schematic shows the method of making returns, using the circuit of Fig. 2, to avoid the ground return as a possible cause of instability in the audio amplifier.

harmonic distortion shows satisfactory figures. But when audio is passed through the amplifier it is evident that something is wrong.

It produces results similar to fairly severe type intermodulation distortion. Looking at the output from a sine-wave input on an oscilloscope screen, the waveform is similar to that shown at Fig. 7. This is due to the fact that at some high frequency the amplifier is approaching a condition of instability but it does not actually oscillate, because the positive feedback is not quite sufficient to maintain continuous oscillation. However, if the amplifier operates under a condition where plate current stops abruptly, as in the case of class AB, or where the output runs into grid current at some point in the waveform and this grid current stops abruptly, then this parasitic oscillation may appear on parts of the audio waveform. The remedy is similar to that for continuous oscillation using grid—and plate—(and if pentodes are used, screen)—blocking resistors.

The cause of this trouble and the reason why it is hard to trace lie in the fact that with a single sine-wave input, its effect may be not noticeable, except on an oscilloscope screen. The frequency of parasitic oscillation may be too high for the harmonic distortion measuring equipment to register, and it is also too high to affect the movement of the loudspeaker diaphragm, and hence the output sounds quite normal. However, when several frequencies are passed through the amplifier simultaneously, a high-amplitude low frequency will set off this parasitic oscillation which will have a tendency to lose, momentarily, higher frequencies present on the same waveform, because of the saturating effect of the high-frequency oscillation. This is what causes the effect similar to intermodulation.

Tracking down various forms of instability in audio equipment can be a very frustrating experience. Fortunately, audio enthusiasts are not easily daunted. So stay with it, take a little time to figure out the possibilities, and remember, "Faint heart never won fair music."

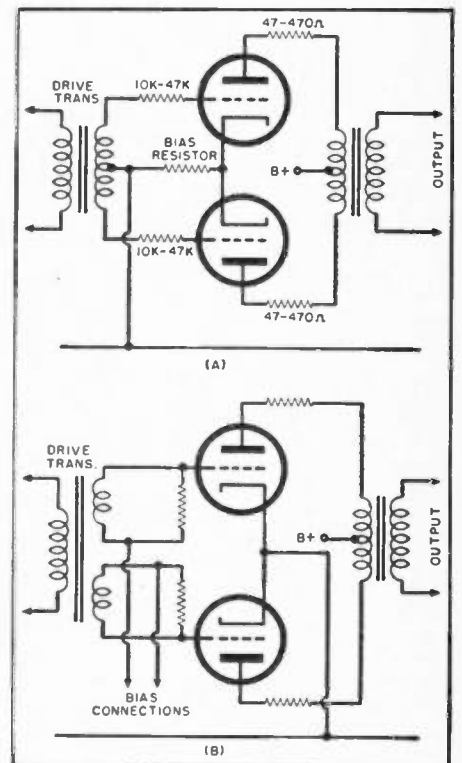


Fig. 6. (A) Push-pull output stages can sometimes go into high-frequency oscillation. Grid and plate blocking resistors, connected close to the tube pins, will stop this. In pentode or tetrode stages, such blocking resistors may also be required. (B) Alternative connections necessary for output stages running into AB_2 or B_2 , to avoid stage loading by the grid current.

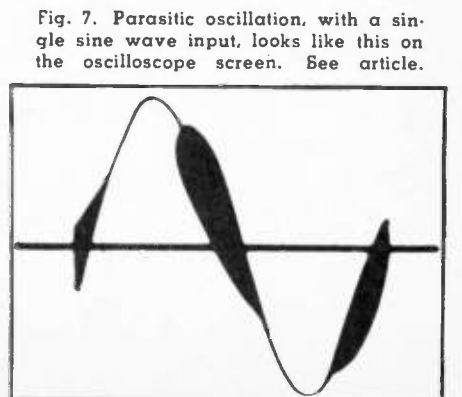


Fig. 7. Parasitic oscillation, with a single sine wave input, looks like this on the oscilloscope screen. See article.

A Cathode-Follower Amplifier



Over-all view of the author's 3-watt cathode-follower amplifier. It is built on a chassis measuring 5" x 7".

By RALPH C. JOHNSTON

Details on a novel all push-pull amplifier with cathode follower output, using a low-cost output transformer.

THE audio experimenter may have noticed that most of the popular circuits use a relatively expensive output transformer. These transformers have high primary inductance, low leakage inductance, and low distributed capacity, and may represent one of the most expensive items in the high-fidelity amplifier. However, it is possible to extend the low-frequency response of an inexpensive output transformer by using it in a cathode-follower circuit. This is because of the large amount of negative feedback introduced in a cathode follower. Another feature of the circuit is its good damping. The output impedance of the amplifier is so low, that the damping is limited principally by the d.c. resistance of the output winding, a fraction of an ohm.

When most people think of a trans-

former cathode-follower amplifier, they think of an inefficient circuit using half a dozen 6L6's in push-pull parallel and a transmitter-sized power supply. This is what is required if power on the order of 10 or 15 watts is to be obtained. As the output power goes up, the factors of driving voltage, power supply size, and current rating of the output stage are compounded. The author believes he has reached a good compromise at 3 watts. A single 12BH7 dual triode is used as an output tube.

In quest of more power, a pair of 12B4's was considered. After the circuit was designed, it was found that a 500-volt supply was needed to get enough driving voltage, and that only two more watts of power were obtained.

Since the amplifier was designed for

a phonograph, it was found convenient to eliminate the phase inverter, and take balanced output directly from the cartridge. Power supply hum and extraneous pickup are effectively eliminated since these effects are balanced and tend to cancel in the output stage.

Two output transformers were tried in the circuit. One was a *Merit* A-2936, which is a 10-watt replacement transformer selling for less than \$2.00 net. The other was a *Peerless* S-510-F, a 10-watt transformer having a response ± 1 db, 20-30,000 cps.

Fig. 3 shows the simplified equivalent circuits for a transformer in the plate and in the cathode circuit. Analysis of the low-frequency circuit is fairly simple. When the reactance of the primary of the transformer becomes low enough, it loads the circuit and the response drops off. The cathode follower's lower output impedance allows the primary reactance of the transformer, and thus the frequency, to go much lower before the loading effect of the transformer becomes apparent. When numerical values are substituted in the equivalent circuit, it is found that the low-frequency response is extended about 10 times.

Analysis of the equivalent circuit for high frequencies becomes complicated because the various distributed capacities and leakage inductances are difficult to determine. It was found experimentally that the high-frequency response was attenuated somewhat when using the *Merit* transformer in a cathode-follower circuit. The high-frequency response of the *Peerless* transformer, on the other hand, was hardly affected.

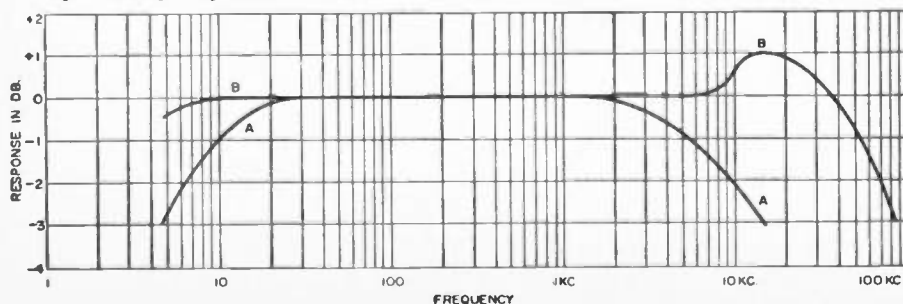
Direct coupling is used throughout the amplifier except between V_2 and V_3 . The cathodes of V_2 and V_1 are run at the same voltage as the plates of V_1 and V_2 respectively. This arrangement eliminates four coupling capacitors and four grid resistors. It also helps the low-frequency response and the stability of the feedback loop at these frequencies.

Since this is an all push-pull circuit, it is important that it be balanced. The constructor should balance the components of the two halves of the circuit as well as he can with the equipment available. However, there are certain features about the circuit which tend to correct any unbalance. R_8 and R_{17} are unbypassed resistors common to both halves of the circuit and provide phase inverter action to correct unbalance. Negative feedback amounting to 15 db is used around the three voltage amplifier stages. This broadens the frequency range and lowers the distortion which is present in high level driver stages.

The output stage consists of a 12BH7 used as a push-pull cathode follower. Resistor R_{18} is used to provide the correct grid bias, allow direct coupling, and to reduce the plate voltage so that the plate dissipation rating will not be exceeded.

Since the cathodes of V_2 and V_1 run around 100 volts above ground, there was the danger of heater-cathode leak-

Fig. 1. Frequency response of amplifier using (A) Merit and (B) Peerless transformers.



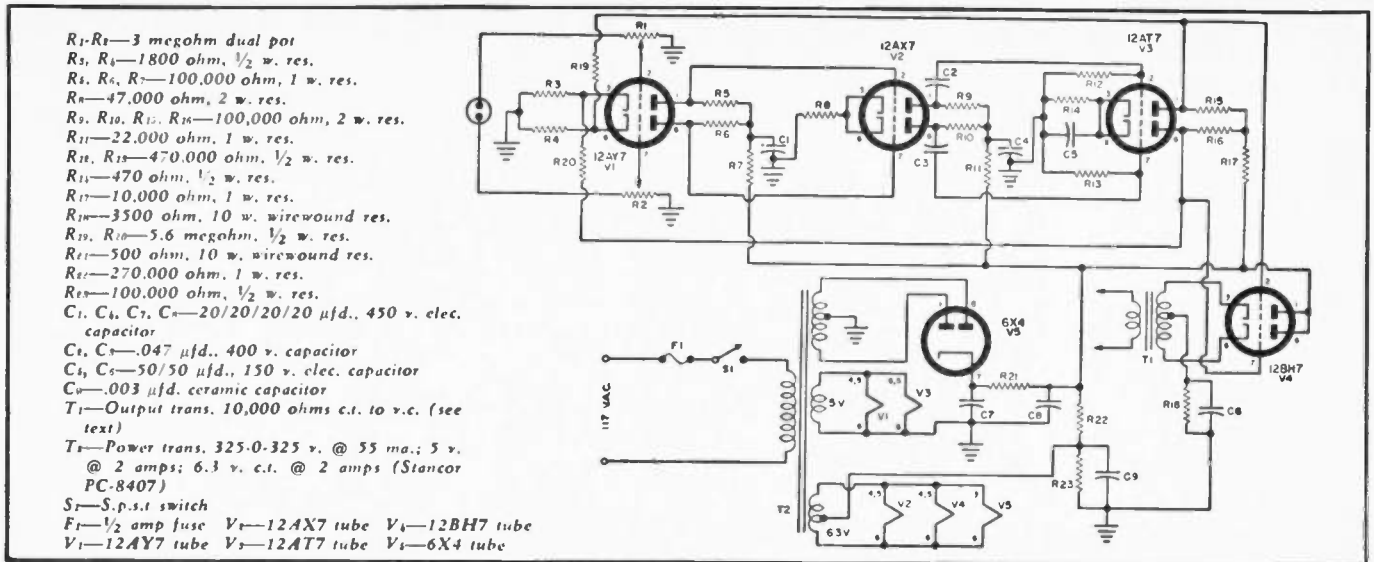


Fig. 2. Schematic of cathode-follower amplifier. An inexpensive output transformer and single dual-triode output tube are used.

Fig. 3. Simplified transformer equivalent circuits. See discussion in the article.

age or breakdown. The ideal solution would be to run the heaters of these tubes at 100 volts above ground, and use a second 6.3 volt winding for the remaining tubes. No suitable transformer was found so V_1 and V_3 were run off of the 5-volt winding, with no apparent ill effects.

Almost any transformer of the stated impedance will work in the circuit. To obtain good high-frequency response, a transformer having low leakage inductance should be used. The primary inductance is not too important because the cathode follower circuit provides good low-frequency response with small values of primary inductance.

To take full advantage of the amplifier, it is recommended that a wide range ceramic cartridge such as the *Electro-Voice* Model 84 be used. It gives results comparable to a magnetic cartridge and needs no preamplifier or compensation.

The amplifier was constructed on a 5 by 7 inch chassis. The *Peerless* transformer would require a slightly larger chassis.

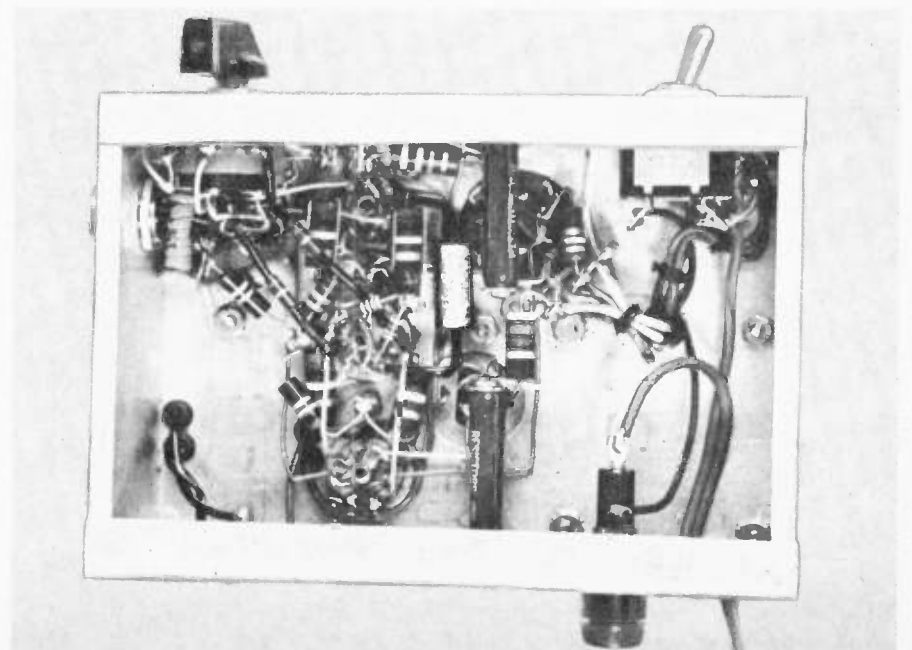
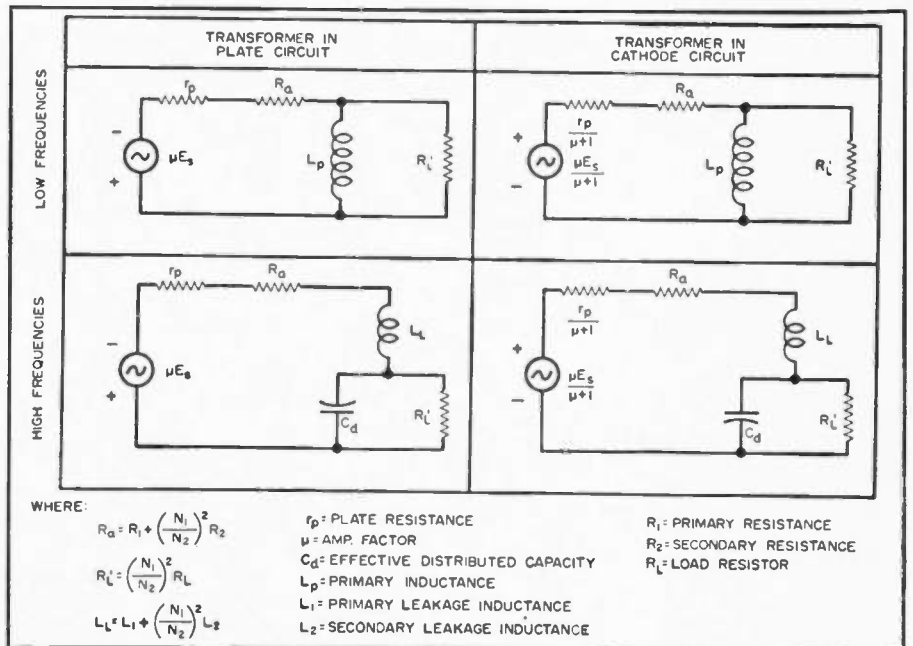
Two-conductor shielded wire should be used between the amplifier and pickup. A shielded lead similar to the one now in the changer arm was placed in the arm to give a balanced lead.

The amplifier is being used with a *G-E* S-1201A speaker mounted in a home-built "Super Horn". The power output is more than adequate since one must shout to be heard when full power is being used. Visitors are amazed to hear such fidelity and volume level from the pint sized output transformer and 12BH7.

REFERENCE

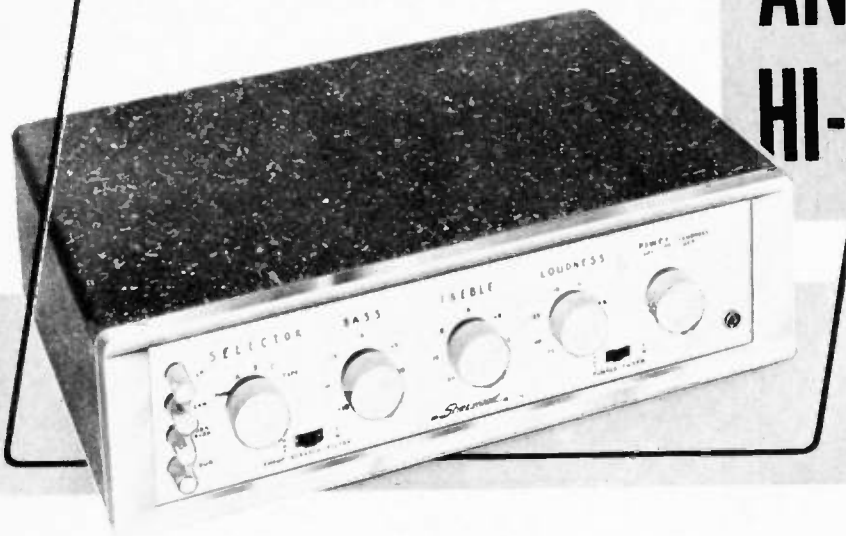
1. Gately, E. & Benham, T. A.: "Super Horn—A Folded Horn Enclosure," *RADIO & TELEVISION NEWS*, September, 1953.

Under chassis view. A larger base can be used if construction seems too crowded.



Front view of the Model S-1000 amplifier-preamp-control system unit by Sherwood.

AN INTEGRATED HI-FI AMPLIFIER



By
EDW. S. MILLER

Gen. Mgr. and Chief Engr.
Sherwood Electronic Laboratories

Design data on a compact unit combining a control unit, preamplifier, and power amplifier in a single housing.

IN RECENT months more and more designers have been turning their talents to the problem of meeting the demand represented by the general market. As a result, several amplifiers have been developed which combine both a phono preamplifier and control system in an integrated, self-cabineted unit.

The unit to be described, the *Sherwood* Model S-1000, provides these features plus a complete group of operating refinements which the less technically minded music lover will appreciate.

Recent tests by large consumer organizations and experts show that under typical home-listening conditions, essentially no difference can be discerned among different brands of *good* basic high-fidelity power amplifiers (except in cases where the amplifiers affected speaker operation because of variations in damping).

On the other hand, the same studies showed that differences *can* be heard among most tone-control designs and other controls which affect frequency response in some way. In addition, annoyances such as switch clicks, hum,

rumble, hiss, etc., were objectionable to even the most untrained listener.

Guided by these findings, the S-1000 amplifier was designed to meet the following requirements: (1) low noise and hum level at least as good as that of an amplifier employing d.c. operated heaters; (2) no switch clicks or other disturbances accompanying any control operation whatsoever; (3) accurate push-button-operated record equalizer selector; (4) cathode-follower recording output preceding the tone controls and the loudness control. Provisions must be made to eliminate possible recorder feedback when playing back from the recorder; (5) calibrated tone controls having accurate flat settings and low distortion, especially at the intermediate settings of the treble control where conventional controls exhibit frequency response curves with a shelf-like shape; (6) a loudness control that operates without compensation at a centered or "12 o'clock" setting and thus provides a reserve of flat gain; (7) a front-panel switch to remove loudness compensation and restore conventional flat volume control action with no change in level; (8) a speaker damping switch to provide a choice of high or low and, especially, negative damping; (9) front-panel operated rumble and scratch filters with 12 db./octave frequency slopes; and (10) a good basic 20-watt amplifier which has

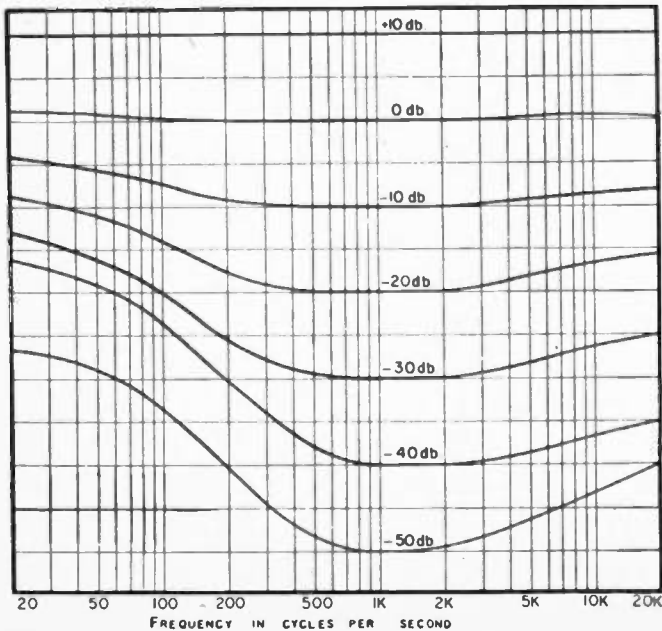
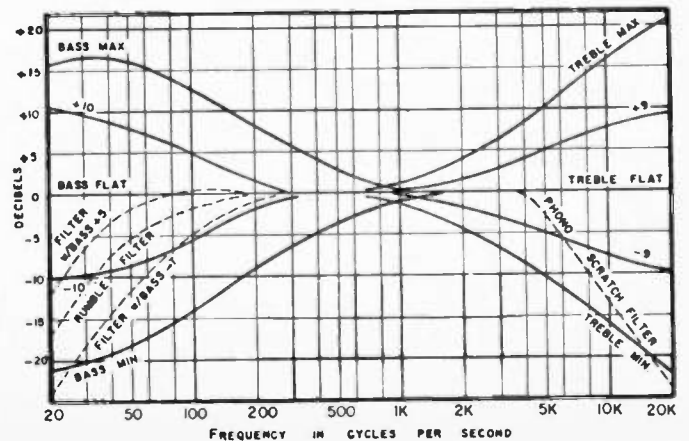


Fig. 1. Compensation curves obtained with the Centrallab Model A control incorporated in the Sherwood Model S-1000 amplifier. Special taps at 19 and 40 per-cent rotation introduce loudness compensation at a more counterclockwise position. See text.

Fig. 2. How the bass control's characteristics combine with the rumble filter's characteristic to give varying frequency turnovers to the sharp low-frequency filtering. For a full discussion of this feature and its effect, refer to article.



less than 1½% intermodulation distortion at 15 watts. (The frequency response at this level must be flat from 30 cps to 15 kc.)

The circuit that evolved is shown in Fig. 3.

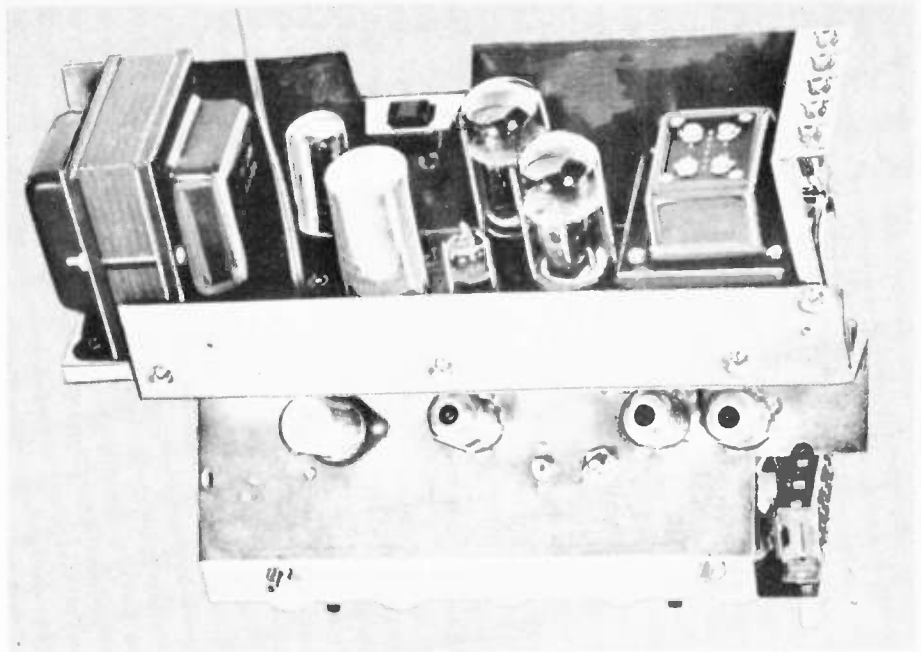
A feedback, bypassed-cathode type phono preamplifier circuit is used with the low-noise, low-microphonic British Z729 input tube. Record compensation is simplified for the music lover by the use of a four-button push type switch. Each button gives exact equalization to 30 cps for LP, RIAA (new AES), London, or European recordings.

By careful heater layout, hum is kept below a -60 db or 1½ μv. level (at input grid). This is equivalent to results obtained with d.c. operated heaters.

A s.p.d.t. switch, in conjunction with C_{in} and R_{in} (across the phono input), is used as a scratch filter with magnetic cartridges. This circuit has a 12 db per octave roll-off and a 5 kc. turnover when used with a G-E pickup.

To avoid the possibility of overloading a recorder by improper tone control adjustment, the recorder output in the circuit is located ahead of the loudness control and the tone controls. If the recorder output is used with the selector switch in the "Phono" position, the push-button record equalizers and scratch filter may be used. Hence the recorded material, when played back, will have the proper equalization.

With the selector switch in the "Tape" position—for playback—the recorder output is automatically disconnected. The reason for this is that



Rear and bottom views of the S-1000 amplifier chassis. Dimensions are 14" x 10½" x 4".

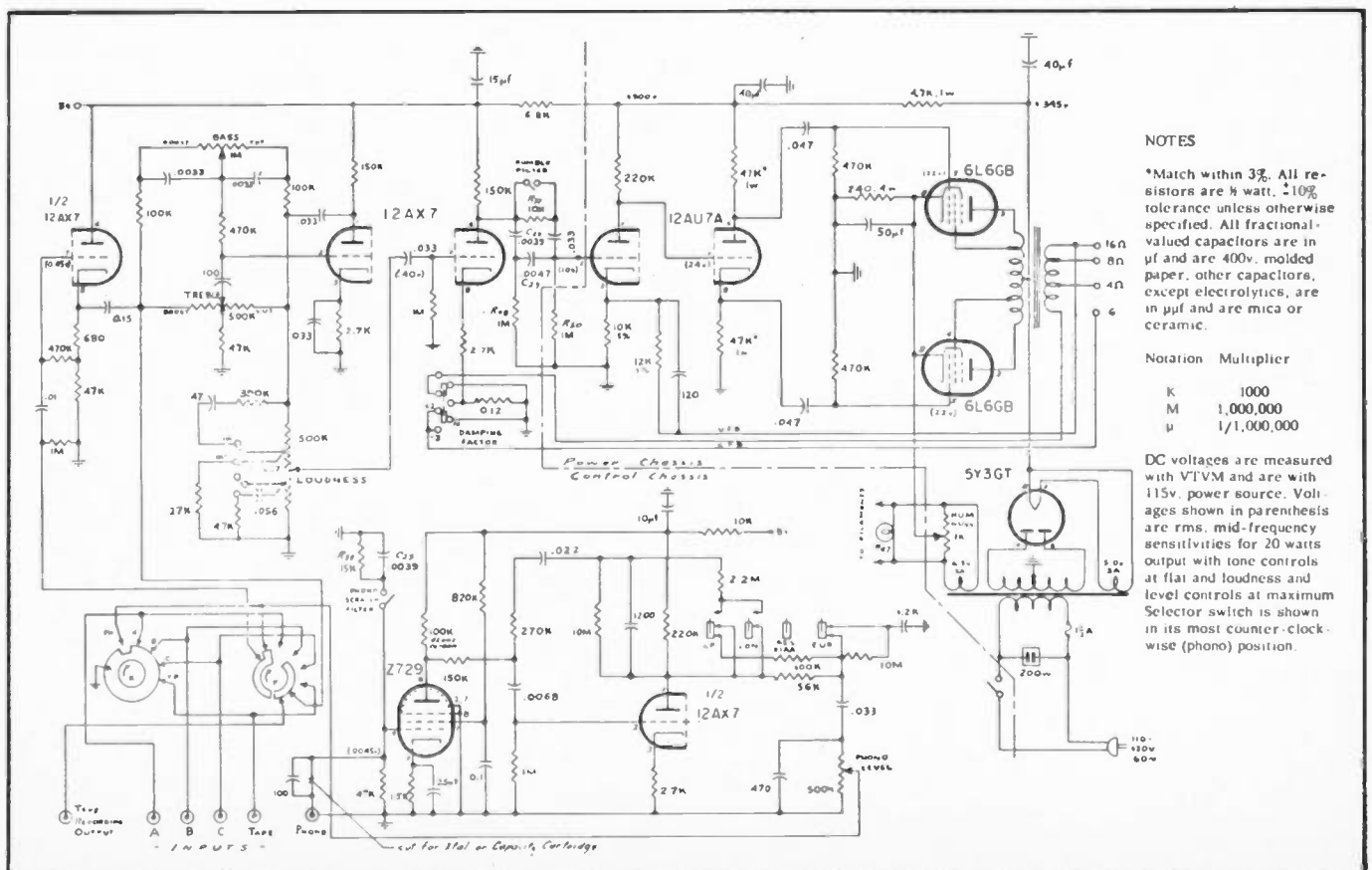
many recorders use interrelated recording and playback amplifier circuitry so that during playback, a feedback situation can develop. Usually, the recorder's input is fed from an amplifier's recording output, which in turn, is connected internally to the amplifier input being fed from the recorder output. To break this feedback loop, it is necessary to remove the recording cable from one of the jacks; or, as is done in the S-1000, the recording out-

put must be automatically disconnected by the selector switch in the tape position.

Monitoring while recording remains entirely flexible since the loudness and tone control functions are independent of the signal to the recorder.

Even tone controls have various degrees of refinement. Some of the disadvantages of the popular combination bass-treble losser circuit are: (1) poor flat setting using normal tolerance

Fig. 3. Schematic of the Sherwood Model S-1000 amplifier. The Z729 tube is a low-noise, low-microphonic British input tube.



NOTES

*Match within 3%. All resistors are ½ watt, ±10% tolerance unless otherwise specified. All fractional-valued capacitors are in μf and are 400v. molded paper, other capacitors, except electrolytics, are in μf and are mica or ceramic.

Notation Multiplier

K	1000
M	1,000,000
μ	1/1,000,000

DC voltages are measured with VTVM and are with 115v. power source. Voltages shown in parentheses are rms. mid-frequency sensitivities for 20 watts output with tone controls at flat and loudness and level controls at maximum. Selector switch is shown in its most counter-clockwise (phono) position.

parts, (2) distortion difficulties because of high driver-tube voltage levels, and (3) intermediate treble curves have shelving actions.

Some of these difficulties can be overcome by using similar circuitry in a feedback configuration. Here, however, problems of instability are introduced with the shelving action still evident in one form or another.

A recent circuit with unity gain² eliminates the first two difficulties mentioned. The problem of shelving can be avoided by proper choice of a resistor inserted between the treble control tap and ground. The *Sherwood* design uses 47,000 ohms.

The response with the controls in the flat positions is $\pm 1/2$ db or better. The controls' rotational positions for this response, in addition to being marked, are well defined due to a flat portion in the resistance element curve at the middle of the controls' range.

Following the tone controls is the loudness control. Its exact circuit position was determined as the best compromise between a too-high level position, which is free from hum pickup problems but subject to early stage overloading, and a too-low level position with the difficulties of hum and noise amplification.

Although loudness controls with continuously variable compensation based on Fletcher-Munson curves^{3,4} are quite familiar to the audio engineer, the less-informed high-fidelity user frequently condemns their operation, requesting that a loudness compensation "in-out" switch be provided to restore the action to that of a foolproof volume control. This user's difficulty arises from his usage habits with the conventional volume controls found on commercial radio and television sets. Here, the normal listening level is almost always obtained with the volume control near the "12 o'clock" or center position. A similar position on a typical loudness control results in as much as 15 db bass compensation, which is only desirable for low-level listening. Therefore, the user having his loudness control adjusted at 12 o'clock for normal listening, immediately complains of excessive bass compensation and loses faith in loudness controls. On the other hand, a flat or uncompensated level is usually available only near the full clockwise position. Controls thus adjusted, lack that extra reserve of gain which enables the user to show off his powerful amplifier's performance without disturbing preset level adjustments.

The loudness-control designer, desirous of overcoming these difficulties, is immediately confronted with a manufacturing problem. In order to have the loudness compensation introduced at a more counterclockwise position, he requires the control to be supplied with taps lower than the standard 37% and 62% rotation. At this time, no

control has been made available in the highly mass-produced $1\frac{1}{16}$ in. diameter size with other than these standard taps. *Centralab*, however, now supplies its premium Model A ($1\frac{1}{16}$ in. dia.) control with special taps at 19% and 40% rotation. In addition, the advantages of its low-noise, long-life construction, made possible by a non-rubbing contact design, are "extra" benefits in the amplifier design.

The compensation curves obtained with this control are shown in Fig. 1. Note particularly the reserve +10 db range with no compensation above the 0 db reference level (normal loud listening). All compensation is completely removed (without change in level) for conventional volume control operation by means of a d.p.d.t. switch. To reduce the number of operating knobs, this switch was ganged with the power "off-on" switch.

One might ask whether the flat response settings of the control (see Fig. 1) do not violate the commonly accepted Fletcher-Munson loudness contours. On the contrary, they follow closely the equal-loudness contours for levels of 90 phons, and above, which are practically flat; thus offering a range of compensated loudness control operation complete from below 40 to 100 phons.

A 12AX7 27 db-gain stage follows the loudness control. Its cathode resistor is returned to ground through a $1/2$ ohm resistor (2 inches of 0.75 ohm/ft. resistance wire) which is also in series with the output transformer's secondary ground return. Depending on the polarities of this connection (determined by the damping-factor selector switch), positive or negative current feedback from the speaker output is added to the voltage feedback around the power amplifier. With no current feedback (switch in center position), the amplifier has a damping factor of 7. This means a speaker connected to the 16-ohm terminal sees a 16/7-ohm source impedance. If the damping selector is switched to its "-2" damping position, current feedback is applied with the result that the speaker is driven by a minus 16/2 or minus 8-ohm source. This increased damping condition is frequently necessary with otherwise underdamped speaker systems to reduce transient bass hangover and distortion.

For already well-damped speakers, it might be preferable to use the "+7" or "+2" damping positions. Details of variable speaker damping have been covered in several articles^{5,6,7}.

The output of the 12AX7 feeds through a double RC high-pass filter network (R_{30} , R_{70} , C_{22} , C_{23}). This network was designed to attenuate 27 cps (the most frequently-encountered phono turntable motor "rumble" frequency) by 12 db. The double RC frequency curve rises rapidly (see Fig. 2) so that bass notes of 100 cps are

down only $1\frac{1}{2}$ db. A typical bass tone control curve with this attenuation at 100 cps would only be down 5 db at 27 cps. Conversely, a bass curve attenuating 27 cps by 12 db would still be down 8 db at 100 cps. From this, one can see the importance of having available such a special filter. Obviously, even the best bass tone control is not designed to handle this special turntable rumble situation.

On the other hand, the bass control's characteristics combine nicely with that of the rumble filter to give varying frequency turnovers to the sharp low-frequency filtering as shown in Fig. 2. Although the turntable rumble frequency is not always evident with less-effective loudspeaker systems, its presence still might drive a woofer into non-linear operating regions. The effectiveness of this filter, in this case, is in removing speaker IM distortion. A simple front-panel s.p.s.t. switch shorts out the filter for normal flat operation. Note the 10 megohm "click"-removing resistor, R_{37} .

In the power amplifier, virtues of the ultra-linear, push-pull output were utilized in obtaining 20 watts (at output tube grid current point) with either 6L6GA's or the newer 6L6GB's. The latter tubes allow more compact design and offer the advantages of the more modern button-stem construction.

The output pair is driven by a d.c.-coupled, split-load phase inverter, popularized by Williamson. Used in this stage is the new 12AU7A tube which features low-microphonic construction similar to that introduced in the premium 12AY7 tube.

The 14 db of inverse voltage feedback around this 3-tube system reduces the intermodulation distortion (60 cps:7 kc./4:1) to less than 1% at 10 watts while still maintaining a wide stability margin for use where the speaker load impedance happens to be capacitive.

The performance and flexibility found in a complete unit of this type was unheard of just a few years ago. Combining uncomplicated circuitry and simplicity of controls with complete versatility and excellent response—all in one compact unit—represents another step forward for the high fidelity enthusiast.

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CHAPTER 4

LOUDSPEAKER ENCLOSURES

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THE "REBEL 5"

By
JAY CARVER
Cabinart

Construction data on a compact, portable speaker cabinet which is designed to be used with a 12-inch loudspeaker.

THE smallest of recent "Cabinart" developments in the field of corner horns is the 20-inch high "Rebel 5." The size and price of this particular cabinet might be misleading since, in reality, this enclosure offers excellent performance for so compact a design.

While crowded, the "Rebel 5" will accommodate a separate 3-way speaker with a 12" woofer with crossover points at 1000 and 5000 cycles.

Obviously the great, powerful boomy bass and tonal response which characterizes a "Klipschorn" are not available in this enclosure. Instead, this design gives a response which is as smooth for its size and as free from distortion as could be expected with the necessarily high crossover frequencies.

Naturally, the minimization of distortion requires that the response at the extreme bass end be attenuated. Therefore, if one compares the "Rebel 5" with the "Klipschorn" when playing organ music, one finds that the bottom couple of octaves are attenuated but the sound output remains clean. Its "controlled bass," or lack of a boomy peak of response, does not permit the diaphragm of the speaker or speakers to "free wheel," generating its own frequencies and cross modulating with the signal. Surprisingly, on lighter music of the piano (and especially if the repertoire avoids the lower octave) one has to listen closely to detect the difference between the "Rebel 5" and the "Klipschorn." Of course, it is assumed that the "Rebel 5" in this instance is driven by a 12" bass cone with a heavy magnet, a horn-loaded mid-range unit, and an extended-range tweeter.

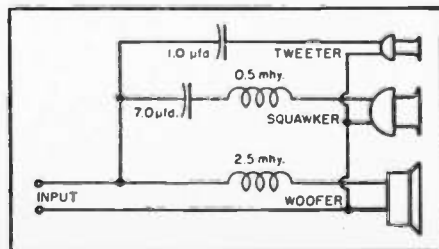
The choice of drivers for the "Rebel 5" can be determined by applying the following criteria: In the \$40-\$60 price range, the best choice would be two-way coaxials incorporating horn-type

tweeters. The single voice coil speaker is not advisable with this design because of the distortion and lack of definition which may result. Drive systems in the \$80-\$120 class may be built up from a 12" cone woofer, a mid-range driver of the "ball park" type on a short horn, a tweeter, and a crossover network. In the case of woofers, high-efficiency units with light magnets should be avoided.

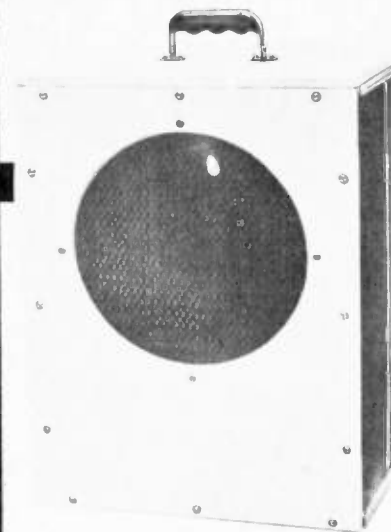
Assuming use of mid-range and tweeter systems, the squawker or mid-range unit as applied to the "Rebel," is required to cover slightly more than two octaves. Preferred are the public address type, compression units using a 2" diameter phenolic diaphragm. The tweeter must be of the type which is free from peaks in the 5000 to 10,000 cps region or an unnatural "presence" peak will exist. Preferable is a diaphragm not larger than $\frac{3}{4}$ " of the phenolic type to produce the internal damping of transverse waves within the diaphragm and a horn not to exceed 3" in length with an adequately small throat for proper loading.

A crossover network for a multiple system in the "Rebel 5" should be one

Suggested crossover network for three-way "Rebels." Constants given are for a 16 ohm voice coil with 1000 and 5000 cycle crossover frequencies. Since this network is a special design, it would have to be made up of individual components. The inductors used are of the variable type, similar to the UTC VI-C13 and VI-C10.



Two models of the "Rebel 5," the one in fine woods (left) and the "utility" model (below) with a carrying handle for portability.



having a 6 db/octave slope instead of the type having a sharp cut-off. When using separate, 3-way drive systems in this enclosure it is possible to achieve the same response and freedom from distortion above 1000 cycles as are available in the "Klipschorn" in the same frequency range. This means that the reduction in size and cost has resulted in a sacrifice in tonal response only in the region below 1000 cps.

With performance such as this, the enclosure is suitable to recording monitoring applications, for use with portable motion picture projection units, and in home sound systems where space is at a premium.

The "Rebel 5" is the third of a series of "Rebel" cabinets put out by *G & H Wood Products Company*, 75 North 11th Street, Brooklyn 11, N. Y. in its "Cabinart" line.

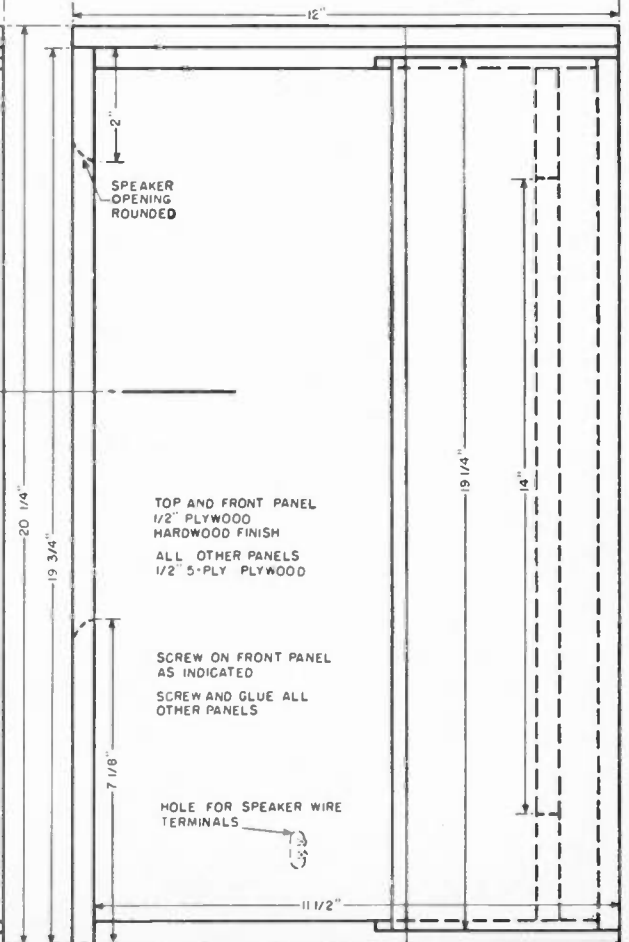
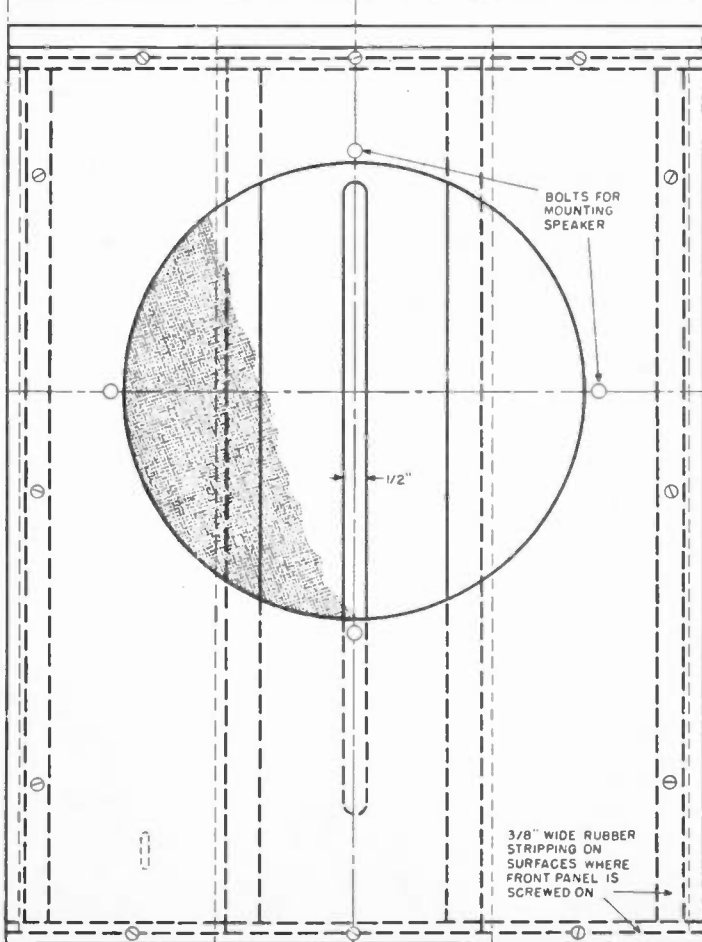
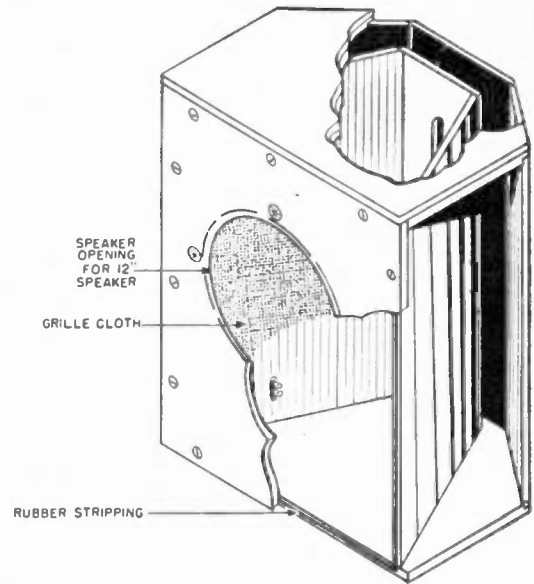
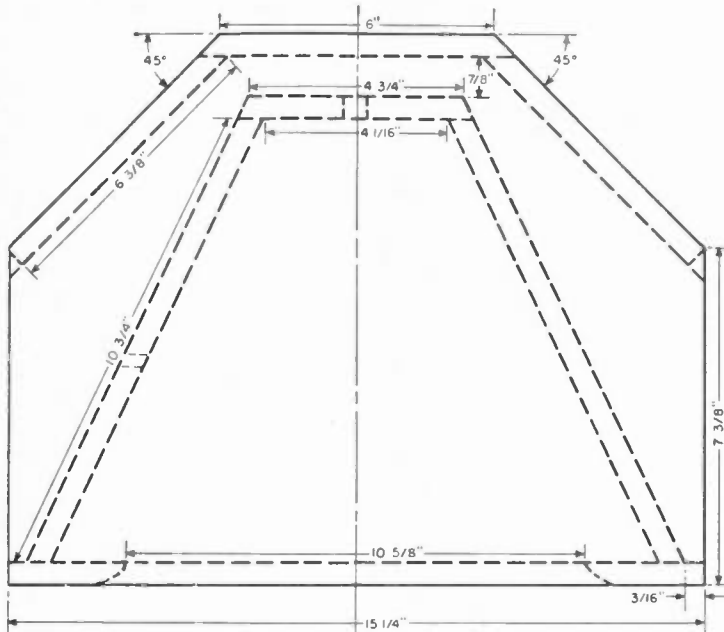
The first, the "Rebel 3" was the largest unit and was designed especially for 15" speakers. The "Rebel 4" was offered in both 12" and 15" versions, the former being described in the October 1953 issue of *RADIO & TELEVISION NEWS*.

This "Rebel 5" is actually the smallest of the group and was designed primarily for use where space is an important factor and portability is desired. It has one added feature in that although best performance is obtained when used in a corner, it does incorporate its own corner horn making it possible to use it against a flat wall. The other two versions did not include this feature.

The "utility" version of the "Rebel 5" is priced less than \$35.00, cut for 8" or 12" speakers and under \$50.00 for the versions in fine woods and leatherette. All required cutouts for the loudspeakers to be used are included in the net prices. Both the portable and fine-wood models are shown in photos above.

Mechanical Details on the "Rebel 5" 12" Speaker Cabinet

Additional notes on the construction: This enclosure is designed for use with any three combinations of speakers: 1. A single 12" coaxial speaker incorporating a woofer and tweeter. 2. The coaxial unit of (1) with the addition of a separate high-frequency horn. The tweeter of the coaxial unit would then serve as a mid-range unit. 3. Individual units consisting of a woofer, squawker (mid-range) and tweeter. Note that the additional cut-outs for the mid-range and tweeter units are not shown in the mechanical details below. Obviously, the size of the cut-outs would depend on the particular units used. They can, however, be added to this enclosure. To do so, simply invert the cabinet or the mounting board so that the woofer unit is at the bottom of the enclosure. Add the other units above the woofer and assemble them directly to the mounting board.



THE KARLSON SPEAKER ENCLOSURE

By
JOHN E. KARLSON

Karlson Associates

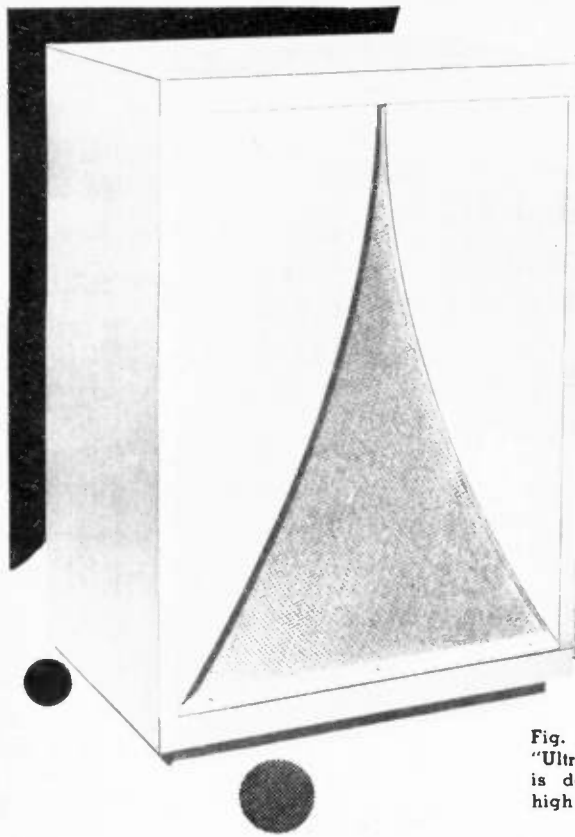


Fig. 1. Over-all view of the Karlson "Ultra-Fidelity" model enclosure. It is designed specifically to house a high-quality 15-inch loudspeaker.

Construction and engineering details on an enclosure for 15-inch speakers which incorporates acoustical coupling to provide improved transient response, good definition.

THE Karlson "Ultra-Fidelity" enclosure has been designed to fill the need for a loudspeaker system whose performance would be at least comparable to that of the other components in a high-quality music system.

In order to bring this worthy motive out of the realm of the fantastic and into the pale of practical accomplishment, some extremely knotty problems had to be licked. These included the design of an acoustic coupler capable of providing a flat response over the frequency range from 20 to 20,000 cycles, an omnidirectional radiation pattern over the same range, accurate tonal phasing, and nearly optimum transient response. Of course,

this unit should also fit into the average sized living room.

In reviewing all of the available approaches to this problem it became clear that none of the existing techniques were adequate in meeting these difficult requirements. For example, horns could provide good coupling down to these frequencies if they were big enough, but even if the size could be tolerated, the frequency range could not be covered by a single horn. When the higher frequencies are radiated from horns, an increased directive beaming effect is experienced which distorts the relative tonal values of the reproduced material. This effect can be reduced by using several horns, but when this is done, accurate

phasing for all of the frequency components in any musical sound becomes virtually impossible. When direct radiators are substituted for horns in the higher frequency ranges, phasing difficulties must result due to the widely differing acoustic paths of the high and low frequencies. In addition, the transient responses of horns and direct radiators are noticeably different.

In continuing this analysis it became increasingly obvious that some new acoustic device was needed which could meet these requirements. As a result the exponential coupler was brought into existence. Fortunately, the inherent characteristics of this type of coupler are almost ideal for our requirements in that it has:

- (a) A flat frequency response over a desired range
- (b) Extremely uniform dispersion of sound at all frequencies when properly designed
- (c) Excellent transient response
- (d) Potentialities for point-source phasing

Probably the least obvious advantage in the list is that of point-source phasing. However, when we consider that only a point source of sound provides absolutely uniform dispersion and phasing at all frequencies, the importance of this feature becomes a little more obvious. Strangely enough,

Fig. 2. Impedance characteristics of test speaker measured without any enclosure.

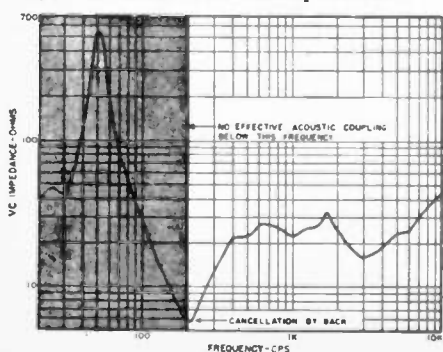


Fig. 3. Same speaker in 2700 cu. ft. infinite baffle radiating into a free field.

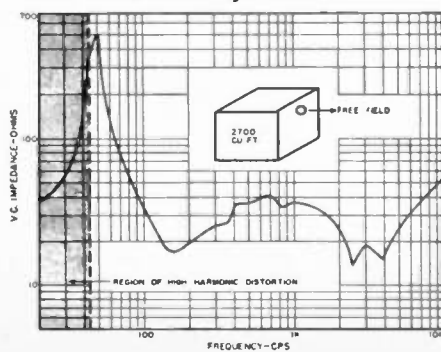
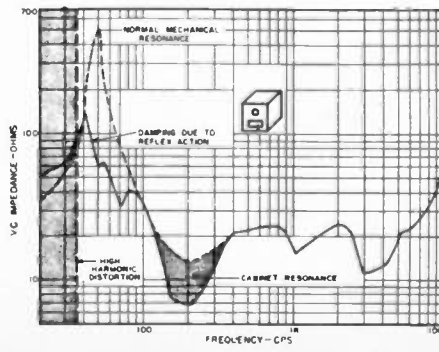


Fig. 4. Same test speaker in a commercial model 6 cu. ft. bass reflex cabinet.



if a radiator does not have uniform dispersion of its sound, the effect will be to pinpoint the location of the speaker due to its more directive beam pattern. This phenomenon has often been erroneously called the "point-source effect" whereas it is actually due to a relatively large directive radiator.

Design Considerations

In evolving the final design for this enclosure, it was obvious that the end result had to look attractive enough for use in even the most affluent homes. Therefore considerable effort was expended in creating a model which had an optimum aspect ratio, suitability to any decor, maximum utility, and adaptability to almost all cone-type speakers.

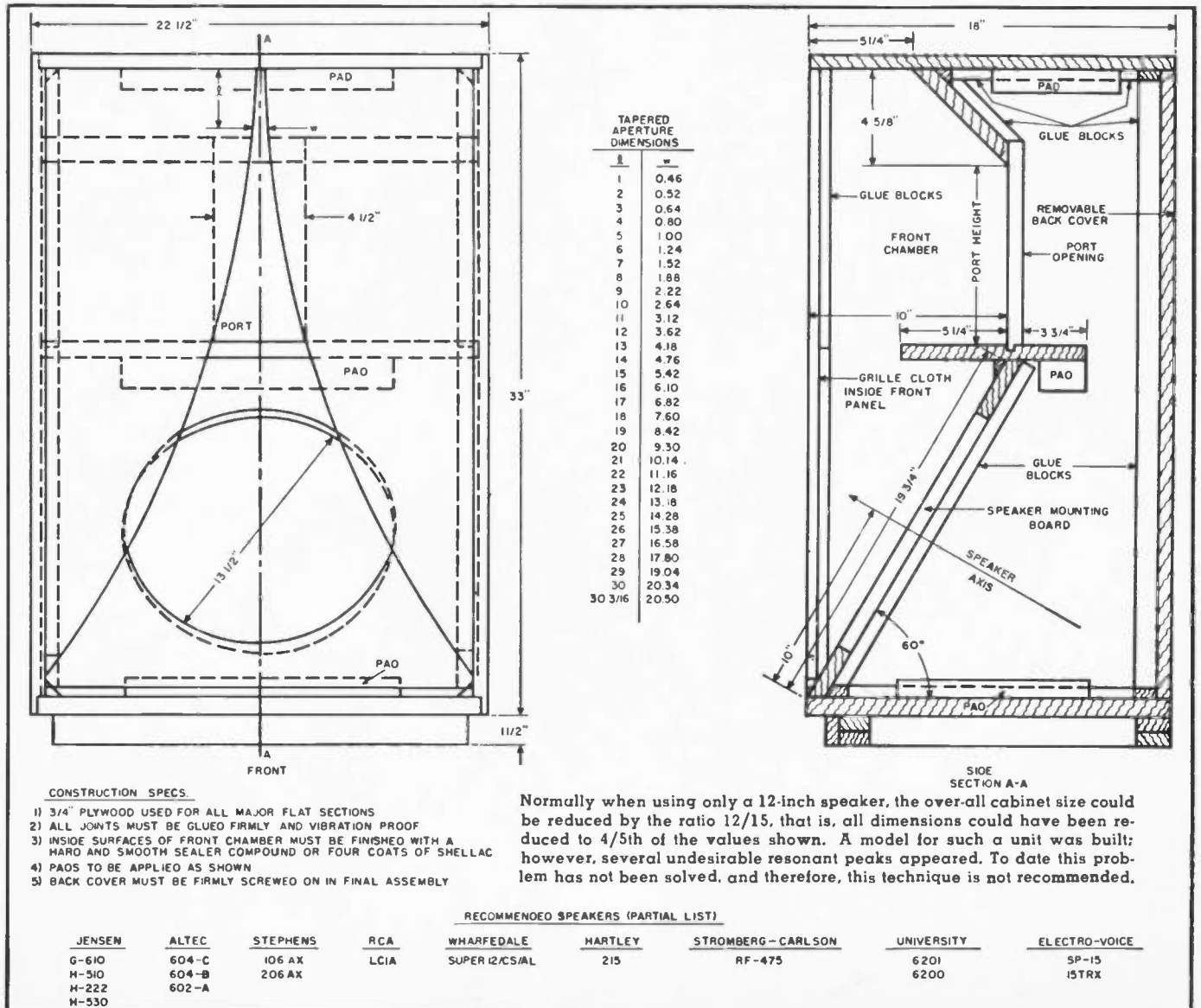
When these factors were decided, perfection of the internal design was begun. The speaker was provided with both front and back loading for optimum coupling. Ordinarily front loading presents difficulties due to the natural resonances of almost any

structure used. However, the exponential coupler solved this problem and the speaker was matched to this coupler in the manner shown in Fig. 5. With this careful matching of the relative coupler and speaker impedances, the size of the speaker is virtually expanded to that of the air in the tapered aperture which is capable of tremendous excursion without distortion. When the front and back loading is properly phased and balanced through the use of the port and matching shelf between the front and rear chambers, the speaker becomes capable of extraordinary intensities of sound without distortion. Even at the extreme low frequencies the output is predominantly fundamental with usually less than 2% distortion at 30 cycles when used with a good speaker. The solid construction of the cabinet also prevents any bellowing or vibration of the cabinet at these frequencies.

The graphs of Figs. 2, 3, 4, 6, 7, 8, 9, and 10 indicate the reasons for the

performance obtained. A highly efficient speaker was used in a series of tests conducted to give positive evidence of the characteristics of this enclosure. Consequently the relative impedance values are high. A 1:1 impedance ratio bridge was used to measure the impedances. The damping indicated in this enclosure is due to the loading of the air (radiation resistance) rather than heavily absorbent materials. This feature is relatively unique in the field because the broadband impedance matching required for this type of air loading has usually only been available through the use of very large exponential horns. Electrical impedance curves may be readily flattened out by employing "lossy" devices such as heavy padding and constricted openings. However, the result achieved is somewhat equivalent to putting the speaker in a pile of pillows. This technique will yield very flat impedance curves, but no sound. With proper air coupling, all of the energy in the speaker

Fig. 5. Complete mechanical details on the Karlson "Ultra-Fidelity" enclosure. A 12-inch speaker can be used by either using a separate conversion board or cutting the speaker opening smaller. This particular cabinet was designed for use with a coaxial type speaker. If a single speaker is used, it should be of the extended-range type.



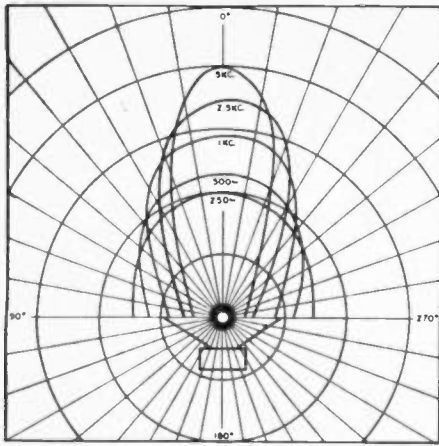


Fig. 6. Relative responses with constant acoustic output for 15" direct-radiator. Side lobes are omitted to simplify graph.

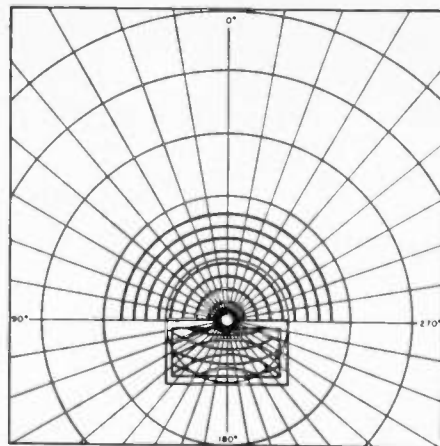


Fig. 7. Uniform dispersion characteristics of radiation from a slot. Very little variation occurs with changes of frequency.

are unnecessary for even the loudest level of operation. Loading only one side of a cone cannot possibly achieve the same results because full control of the cone doesn't exist over the complete cycle of its travel.

Coupling Performance

A series of tests was conducted, as previously indicated, to establish the characteristics of the coupling achieved in the "Ultra-Fidelity" enclosure. The most sensitive indication was considered to be that of taking a series of electrical impedance curves to indicate the differences resulting from mounting the speaker in various forms of enclosures. A curve was first taken of a speaker lying on a bench without any enclosure or baffle board attached. The results of this test are shown in Fig. 2. It will be observed that cut-off occurs in the region of 200 cycles, but that the general slope of the mechanical resonance peak, occurring at 50 cycles, is virtually identical to that found in Fig. 3, which shows a corresponding curve with the speaker mounted in an extremely large infinite baffle radiating into a free field. Apparently there is virtually no difference in the coupling achieved by the infinite baffle type of mounting, and therefore its primary advantage is that of isolating the frontwave from the backwave of the speaker. Naturally any sound occurring in a region of 50 cycles in an infinite baffle of this size would send the speaker cone into a prolonged oscillation. Even when the speaker has this extended travel, it doesn't necessarily produce a considerably greater output at this single frequency because the vibration of the cone alone is inadequate for producing a proportionate amount of acoustic power. In other words, without auxiliary coupling to the air, a cone will simply beat the air ineffectively at the very low frequencies. Some improvement in this mechanical resonance curve is shown by the use of a bass reflex cabinet tuned to anti-resonance at the same frequency. See Fig. 4. Also, the radiation from the port in the neighborhood of this frequency will increase the efficiency of the speaker output in this general range. However, below this resonant frequency, harmonic distortion increases very rapidly with the result that at

is dissipated in the room rather than in the enclosure.

Transient Response

In addition to the greater efficiency derived from the close coupling, a decided improvement in the transient response is achieved, enabling the detection of even the weakest sounds in a musical passage. This feature, when combined with accurate phasing for all frequencies, provides an extremely fine definition of all complex passages of music, so that even in the heaviest orchestral attacks the separate parts may be clearly distinguished. The importance of transient response in electronic music systems cannot be over-emphasized, inasmuch as all natural sounds are composed primarily of transients. Perhaps the simplest test for transient response is that of listening to white noise at high levels. White noise is most readily obtained by adjusting the FM tuner to a position between stations so that the strong hiss is obtained. This hiss usually contains frequency components in the entire audio spectrum. When a loudspeaker system contains any peaks or dips in its response, these will be apparent in a critical listening test. The high frequency peaks will be characterized by a singing predominant tone while low frequency peaks will be characterized by a blow torch effect. The over-all response of a system in comparison with another can also be checked by this method with differences in the high, low, and mid-

dle ranges quite readily apparent. With two flat systems of equal high-frequency responses, the one having the lower frequency range will appear to drop the over-all tonal quality by some appreciable factor.

Distortion

A good deal of the distortion found in reproduced sound can also be reduced or eliminated through the use of strong front and rear speaker coupling. Ordinarily, speaker cones are subject to a wide variety of resonant frequencies inherent to the structure of the cone. If a speaker cone were to be examined under a stroboscope, it would be seen that the cone vibrates in several different modes other than the simple forward and backward motion. When a heavy transient attack occurs, the cone is likely to vibrate at any or all of its resonant frequencies with the result that considerable harmonic distortion is likely to occur. Also, when a cone is subjected to large excursions which take it beyond its limits of linear travel, intermodulation effects occur which generate a whole series of sum and difference frequencies. The net result is likely to be somewhat fuzzy. Now, with heavy damping on both the front and back of the speaker cone, these spurious oscillations are largely damped out or eliminated. Also, due to the heavier loading, the cone requires less travel for the same acoustic output with the result that excursions in the non-linear region of the cone travel

Fig. 8. Impedance curve in Karlson enclosure showing its non-resonant character.

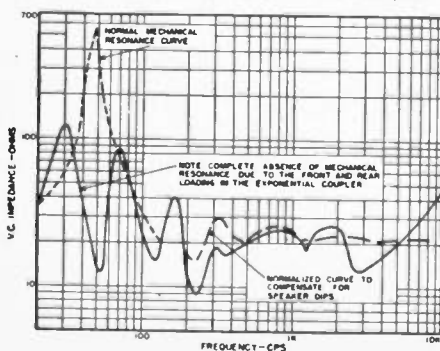


Fig. 9. Impedance curve of matched non-resonant transformer. See text for discussion.

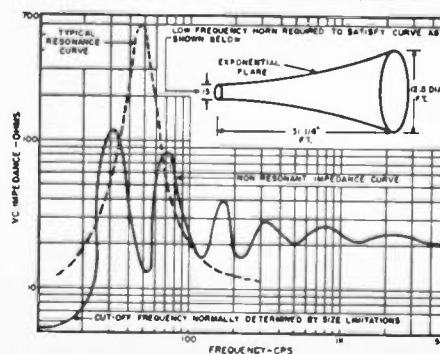
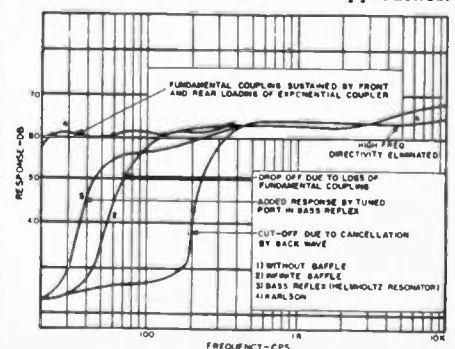


Fig. 10. Frequency response curves of Karlson compared with other standard approaches.



30 cycles fundamental coupling is virtually non-existent. Note also that a serious dip occurs here due to cabinet resonance in the region from 120-350 cycles, the region of maximum voice power. This dip with the consequent adjacent peaks causes a considerable amount of overhang which is typical of many such enclosures. The cabinet tested was one of a standard commercial design with a volume of approximately 6 cu. ft. The impedance curve of the same speaker in the *Karlson* enclosure (Fig. 8) indicates a decided change from the previous tests. First of all, note the complete absence of any indication of a single mechanical resonance curve as is evidenced in the previous tests. The curve resulting from these tests is very similar to that found with very large exponential horns, and is also characteristic of almost all exponentially tapered terminations which are used in achieving a flat response over extraordinarily wide bandwidths. As shown in Fig. 9, this type of curve is peculiar to a special case in the normal resonance formula, and occurs near the point of optimum critical damping at which the resonant frequency of a system is zero. This optimum value occurs only in horns or other distributed parameter devices when they approach an infinite size.

Fig. 10 illustrates the various characteristic frequency response curves resulting from the use of these types of enclosures. The acoustic output of a driver falls off quite rapidly in the lower frequencies, due to its inability to properly couple to the air. With the exponential coupler these difficulties are largely overcome.

Radiation Characteristics

If frequency response were our sole concern, then our design problem would be virtually ended. However, the radiation characteristics of a sound source must also be considered. Ideally the sound from a loudspeaker system should be radiated in a completely uniform manner for all frequencies. When this is not done the loudspeaker system will create entirely different tonal values in the reproduced music than that contained in the original sound. Ordinarily a "hi-fi" enthusiast will quibble over a fraction of a db in an amplifier, and yet it is not uncommon to have a difference of 20 db between the main lobe of a speaker radiation pattern and the side lobes. Obviously high-fidelity reproduction is a misnomer under such conditions. Some may argue that standing waves in a room will vary that much, so why worry about it. However, when we consider that all sound is made up of transients rather than steady-state conditions, the relative intensities of the direct radiation from the enclosure become predominantly important since these transients are not of sufficient duration to create standing waves.

Unfortunately, most loudspeakers and horn combinations have variable

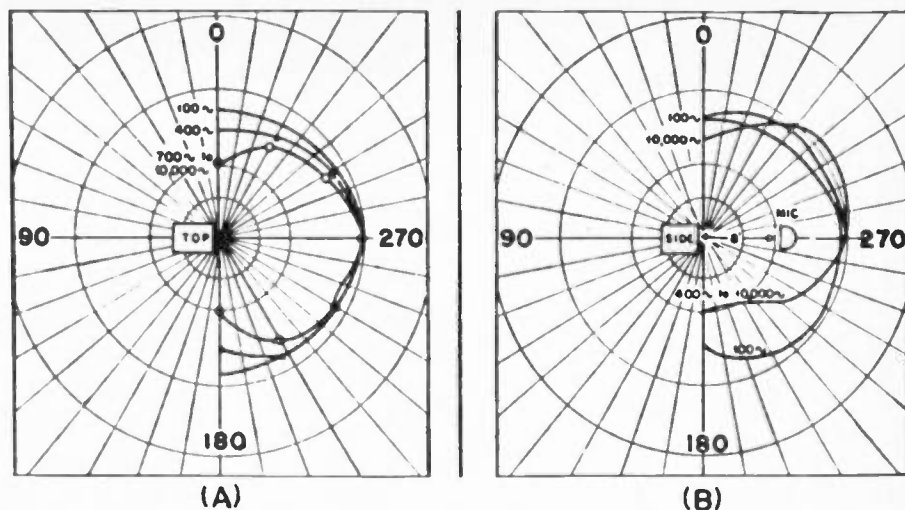


Fig. 11. Polar radiation pattern for (A) horizontal and (B) vertical planes.

radiation patterns for each frequency. At the lower frequencies there is not sufficient deviation to cause real concern. However, when we approach the higher frequencies above 1000 cycles, it is common to observe highly directive beaming effects. If we imagine a different distribution of sound for every frequency above 1000 cycles, we can have some appreciation of why some high-fidelity systems sound distorted and unnatural. Almost all musical instruments operate from relatively small sources of sound with respect to their frequency ranges. Obviously they cannot be reproduced accurately with systems which do otherwise. Fig. 6 shows a typical family of radiation curves of the main lobes inherent to a 15 in. speaker at several different frequencies. The side lobes have been omitted for the sake of simplicity. This should give one some idea of how the sound is frequently reproduced in even the most expensive high-fidelity systems. Various devices have been used to overcome this effect such as small horns, special tweeter speakers, etc., but still the answers achieved are a strong compromise with the ideal. In contrast to this the radiation pattern of sound emanating from a slot is shown in Fig. 7. This, of course, represents the ideal theoretical case. In practice, however, the deviations from this ideal are relatively minor as can be seen from the polar radiation plot shown in Fig. 11, for both the horizontal and vertical planes. The radiation in the horizontal plane is extremely uniform over an angle in excess of 120 degrees. This performance is analogous to that of the TV antennas in use on the Empire State Building in New York City. These are omnidirectional in the horizontal plane and have a narrowing beam in the vertical plane. Some narrowing occurs with the *Karlson* enclosure in this vertical plane. However, the phasing has been so adjusted that even this deviation from the ideal is minimized. As a result, this enclosure has an essentially uniform radiation pattern throughout a solid conical angle of 120 degrees. This angle of radiation, by virtue of the de-

sign, is tipped upward so that its apex falls nearly at the intersection between the floor and the wall against which the unit is placed. This feature still further enhances the coupling between the enclosure and the room due to the reinforcement of these two plane surfaces. When the enclosure is placed in a corner, this reinforcement occurs with three plane surfaces, and an appreciable power gain is thus realized in the low frequency range.

Construction

The construction of the enclosure is reasonably obvious from the drawings of Fig. 5. The side walls have been dadoed out to provide a keyed-in sturdy construction which can resist the onslaughts of the heaviest vibrations and pressures built up within the cabinet. These pressures are quite considerable, as may be experienced by turning up the music and standing immediately in front of the cabinet. The pressure waves will actually be felt. The construction with the panels running from side to side is reminiscent of that of an I-beam and certainly exhibits the same structural strength. The front chamber is finished with a hard, non-absorbent substance to prevent any loss in high frequency response. It is also important to place the 1-in. absorbent pads in exactly the location shown in the drawing. Mechanical vibrations of the cabinet are not heard at any frequency.

The entire assembly is glued, nailed, and screwed together as required with the back being removable for the insertion of the speaker. The speaker chamber has been designed to accommodate all of the well-known types available. Speakers smaller than 15 in. are mounted by means of an adapter plate which fits over the normal 15 in. speaker opening. Since the sizes of many of the parts are quite critical, the production enclosures are all cut with the use of special jigs to obtain absolute uniformity of performance. The parts fit together like a jigsaw puzzle due to the dadoed construction, and are quite readily assembled.

A Novel Push-Pull Speaker System

By GLEN SOUTHWORTH



The front panel and grille cloth have been removed to show internal construction and speaker mounting of the push-pull enclosure.

The push-pull arrangement has several advantages over the conventional method of speaker mounting.

ALTHOUGH a great deal of effort has been expended in the development of multiple speaker systems, there still seems to be several deficiencies that are disturbing to some listeners. The most serious problem seems to be in matching the sound *quality* of a woofer, a mid-range speaker, and a tweeter in such a manner that they blend together naturally and avoid effects such as seemingly having the violins playing in one room, the cellos in another, and the basses in yet a third. This causes a lack of continuity in musical reproduction, and although it may give the impression of added clarity, it may also tend to make a large symphony orchestra sound somewhat like a chamber group.

Specifically, here are some of the problems that may be encountered in putting together a good multiple speaker system using crossover networks to divide the range between low, medium, and high frequencies. First, to match the sound of the three speakers attention must be paid to the distortion characteristics of each unit, to the transient response, to the type of loading used, the power handling capability, the acoustic path lengths between the speakers and the listener, proper phasing of the speakers and crossover networks, and, of course, satisfactory frequency response for each of the three speakers. Unfortunately, the factors of distortion, transients, and load matching seem to be

frequently ignored in commercial speaker systems, as is the problem of matching acoustic path lengths, and such systems might be characterized more as musical instruments than reproducers.

A second series of problems stems from the fact that the speaker system must be supplied with electrical energy from a power amplifier, and the interactions produced may lead to additional distortions. The multiple speaker system, together with its crossover networks, represents a complex reactive load that may cause continuous or damped oscillations when connected to an amplifier using feedback over the output stage. This seems especially true when using highly efficient speakers, due to the fact that the back e.m.f. of the speakers represents a positive feedback component of varying phase and amplitude. However, these problems may be reduced by using an amplifier with only a modest amount of feedback, or by placing a 6 decibel resistive pad between the speaker and the amplifier output.

In order to minimize most of the previously mentioned problems, a number of experimenters have adopted the idea of using clusters of small speakers with light enough cones to adequately reproduce the middle and high ranges, and with sufficient total surface area to move enough air to satisfactorily generate bass tones. In a system of this kind the major problem

is simply to select the right speaker for the performance desired. This, of course, is not necessarily easy, as many small speakers will not perform well below 200 cycles or above 3000 or 4000 cps. A good compromise seems to be found in the 6" x 9" oval speakers, such as the *Oxford 69EVS*. These speakers have a primary resonance of approximately 120 cycles, but are capable of reproducing, with low distortion, at least an octave below this point. At the high frequencies, performance is even more surprising due to the oval shape of the cone, which gives sine wave and transient performance approximating that of a 2" speaker, being virtually flat to 9000 cycles.

The accompanying photographs show four 6" x 9" oval speakers mounted in a three and one-half cubic foot enclosure. Voice coils are connected in series to provide a nominal impedance of 13 ohms, and properly phased for maximum efficiency at low frequencies. The first enclosure uses all four speakers mounted in the same manner, and is capable of excellent performance, the radiating area of the four cones being nearly equivalent to that of a single 15" speaker.

The second enclosure is similar to the first, except that two of the speakers are faced into the cabinet and the polarity of their voice coil connections is reversed in order that all of the cones travel in the same direction under an applied signal. This "push-pull" arrangement of loudspeakers appears to have at least two distinct advantages, the first of these being the reduction of even-harmonic distortion at low frequencies. This is especially important in view of the fact that the conventional cone speaker is an aerodynamic shape that simply moves air more efficiently when it is traveling

outward than when it travels inward, and the air mass tends to slip past the apex of the cone. This seems especially true at frequencies below the primary resonance of the cone where the speaker is no longer mass controlled and diaphragm excursions become relatively large. The result is a lack of symmetry in the acoustic output of a single-cone speaker which may be greatly reduced by using an even number of speakers in "push-pull."

A second advantage of push-pull operation stems from the fact that while the speakers are acoustically in-phase, they are electrically out-of-phase and, as a result, the back e.m.f.'s tend to buck each other and cancel out, thus presenting a more nearly resistive load to the amplifier. However, if sufficient power is available from the amplifier, it may be desirable to place a 15-ohm, wirewound, variable resistor in series with the speaker system. The resistor should preferably be located close to the output of the amplifier in order that effects of cable capacitance on the amplifier may be reduced.

There are a number of mixed blessings in the use of multiple speaker systems of this nature. Chief of these is the fact that small cones usually mean high resonant frequencies compared to large, heavy, single unit woofers. Although this means a peak in the response curve at about 100 to 140 cycles, it also means superior transient response, due to the low mass of the individual small cones. This is especially true in the octave just below resonance, where the speakers are no longer mass controlled. As a consequence, the lower voices of the orchestra, such as the cellos, contrabass, tubas, bass saxophones, etc., seem to reproduce with a fuller, more sonorous sound due to the fact that their transient components are more adequately radiated.

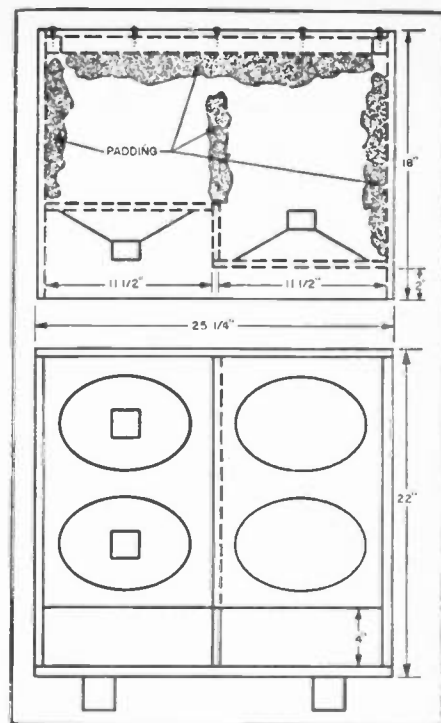
Another advantage in using small speakers is their sensitivity to weak electrical signals. For example, the push-pull speaker system using *Oxford 69EVS's* will produce an audible signal with an electrical input of as low as 1/100 millionth of a watt, and will reproduce natural and pleasing music at peak inputs of one milliwatt or less. This is a very important factor, even when listening at room shaking volume levels, as the weaker signals must be properly radiated or many of the less powerful instruments in the orchestra will be lost or masked out. This frequently leads to poor balance, particularly in the bass region due to the weakness and transient nature of most bass instruments. Even when excessive amounts of electrical equalization are used, in order to make reproduction more tolerable, lack of sensitivity may cause a large symphony orchestra to sound like a chamber group recorded in a small room.

As mentioned earlier, phasing is a problem with any multiple speaker system due to the differing acoustic path lengths of the speakers. In the push-pull speaker system interference

in the mid- and high-frequency range may be minimized in two ways: first by using acoustic low-pass filters to block off high-frequency radiation from part of the speakers; secondly, by staggering the speakers so that the peaks in one set of speakers tend to fill in the valleys of the response curve of the other set.

The complete push-pull speaker system in the three and one-half cubic foot enclosure is essentially flat from 80 to 9000 cycles, being down 10 db at 60 and 10,000 cps. It will reproduce a dynamic range of 90 decibels comparable to the best available amplifiers, and exhibits superior reproduction of transient signals. Sine wave response is clean over the entire useful range of the speaker system. The design is simple and economical and presents few of the problems of conventional two-, three-, or four-way speaker systems.

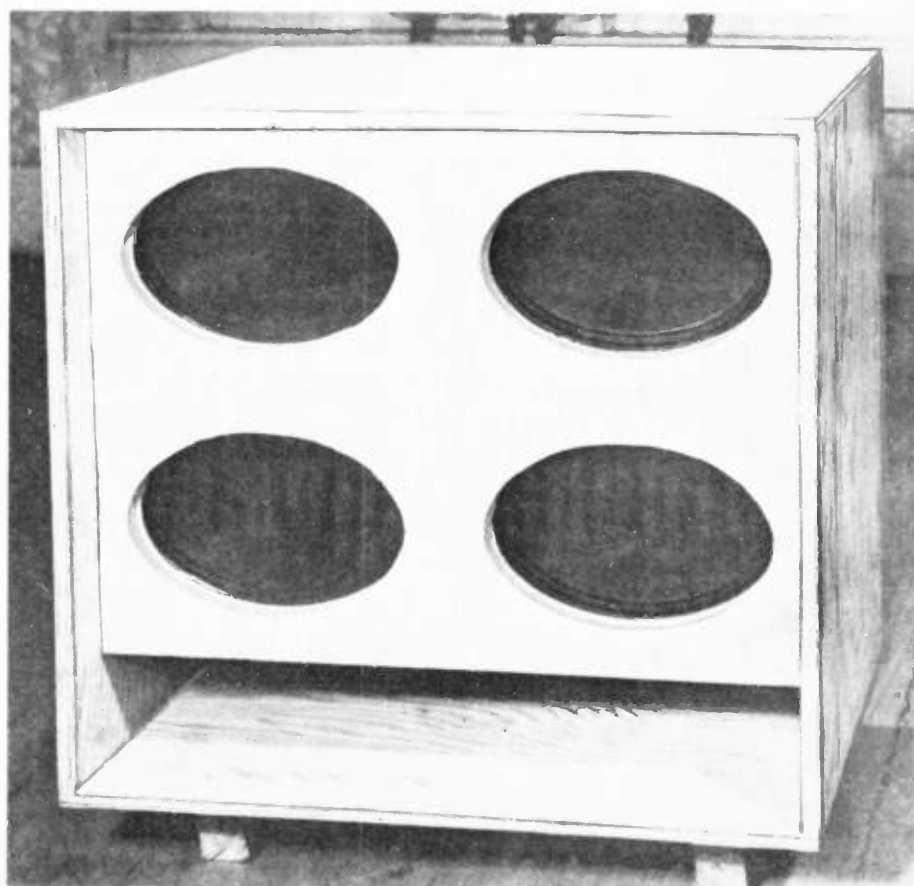
The actual performance, listening-wise, has been carefully observed for over two years in conjunction with a wide variety of equipment, input sources, and acoustic environments, including service in high-quality sound re-enforcement. In all cases it has faithfully reflected the quality of the signal applied to the voice coil terminals. Especially recommended is the use of two of these speaker systems, as this allows some compensation for the effects of room reflections and creates a sound "image" in the space between the two enclosures. The resulting spaciousness of sound creates



Dimensions of 3½ cubic foot enclosure for push-pull speaker system. All panels should be of ¾" plywood. All joints should be glued and nailed, except back which is screwed on. Sides are lined with acoustically absorbent material as shown above.

a much greater illusion of reality, and transmits the acoustics of the original pickup into your living room, even with old shellac recordings. —30—

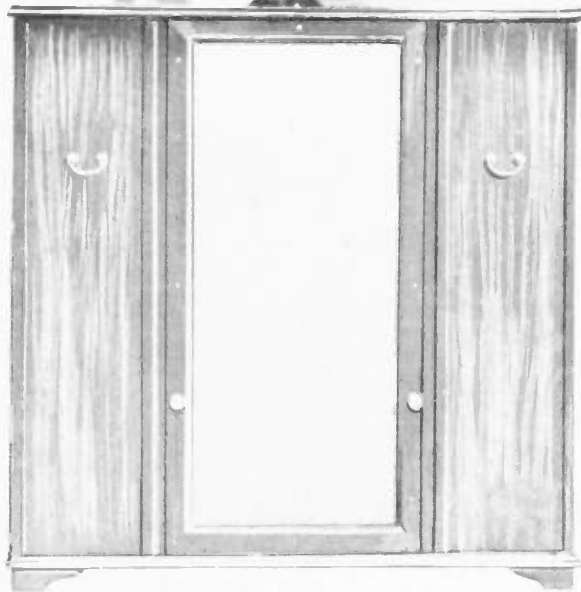
Four 6" x 9" oval speakers are wired in series and mounted in conventional manner. The shape and size of the enclosure is the same as shown in the diagram above.



ALL-PURPOSE ENCLOSURE FOR 12" SYSTEMS

By OLIVER READ

Editor and Asst. Publisher
RADIO & TELEVISION NEWS



Part 1. An improved and more compact design of the "Fold-a-flex" loudspeaker enclosure. It provides a choice of folded horn, bass reflex, infinite baffle.

THE "all-purpose" loudspeaker enclosure described in these pages just one year ago has met with considerable success. Many of our readers built the "Fold-a-flex" from the data supplied for 15" systems¹. Much time has been spent since by the author in a series of experiments with several enclosures (employing the "Fold-a-flex" principle) of smaller dimensions. Working models have been constructed, tested, and measured for response and impedance characteristics for loudspeakers with 5" to 15" cones and several two-, three-, and four-way systems.

As was expected, the larger enclosures outperformed the pint-sized designs in every respect.

Smooth response in the important bass region requires loudspeakers having cones of from 12 to 18 inches in diameter. These require cabinets of sufficient size to provide the all-important characteristics for loading and damping of the cone. In addition, they must be ruggedly built, properly braced, and constructed of wood at least $\frac{3}{4}$ " thick.

Our approach was to determine the minimum dimensions for the "Fold-a-flex" enclosure that would provide the basic characteristics of folded horn, infinite baffle, or bass reflex for the audio range from 40 cycles. Several of the 12" hi-fi loudspeakers were measured and found highly compatible to the design.

There are three basic types of enclosures. Each has its own characteristics

and each will sound different (even with identical loudspeakers). The finest loudspeaker will, in the wrong enclosure, sound worse than many inferior loudspeakers in the proper enclosure. Leading manufacturers of high-fidelity speakers have recognized the importance of designing the enclosure to meet the requirements of a particular loudspeaker. They have been literally forced into the cabinet business—just to make sure that the reproducing system would be capable of providing distortion-free high-fidelity sound at wide dynamic range from their own loudspeakers. Unfortunately, people's tastes differ widely in selecting cabinets and loudspeakers. Many are satisfied with a good 12- or 15-inch coaxial loudspeaker. Others prefer a two-way woofer-tweeter combination. A great many now enjoy the benefits of the fine three-way systems comprising woofer, mid-range horns, and high-frequency tweeters.

Choosing the right enclosure for a loudspeaker is no easy task, and too often the choice of enclosure is made on cabinet shape, size, or appearance alone without carefully considering the all-important acoustical properties of the cabinet and its effect on the performance of a hi-fi loudspeaker system. Too often results are poor simply because a good loudspeaker is installed in the wrong type of enclosure. What do we mean by *types* of enclosures for loudspeakers?

There are three basic types of "baf-

fling" or enclosures. They are: 1. the folded horn, 2. the infinite baffle, and 3. the bass reflex.

The "folded horn" provides an effective loading to the cone of the loudspeaker diaphragm and is capable of providing better bass at higher efficiency than other types. It gives a close coupling to the air, reduces distortion, and eliminates or minimizes the resonant effects of the loudspeaker. Because of the horn-loading effect, the loudspeaker cone moves but a fraction of the distance (in piston-like fashion) than would otherwise occur.

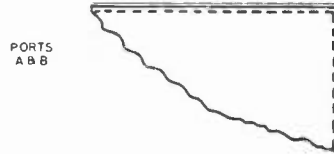
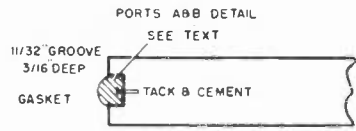
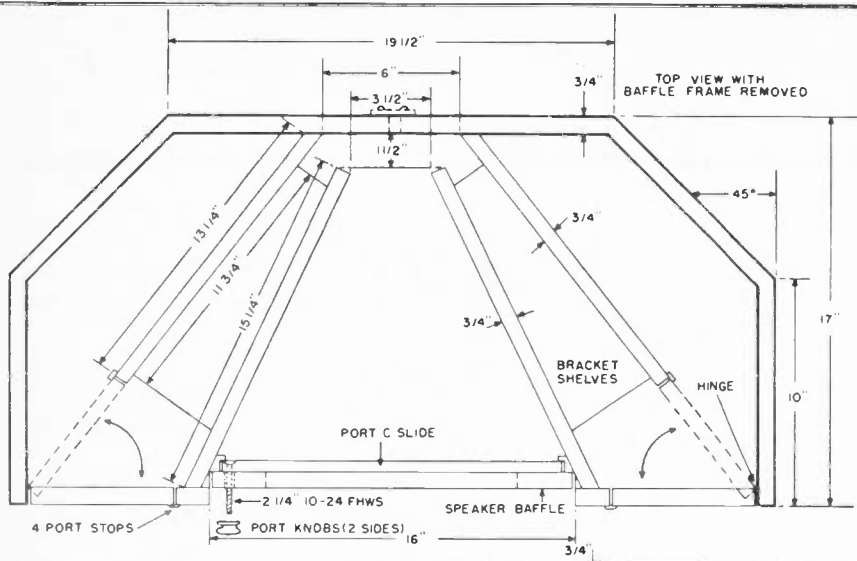
The "infinite baffle" is considered by many authorities to be the best method for mounting loudspeakers. A cabinet providing such characteristics is generally a completely enclosed box (enclosure) carefully braced to prevent vibrations and padded to absorb sounds bouncing around within the enclosure. This type of enclosure (baffle) requires a minimum inside area (air volume) of at least 10 cubic feet for 15" loudspeakers, and at least 6 cubic feet for 12" cones. It is capable of good performance with many single or coaxial speakers.

The "bass reflex" type of enclosure has been popular for many years and is considered to be the least expensive type of those capable of providing satisfactory bass performance. The enclosure employs a port (opening) which is placed below the loudspeaker cone. The area of this port must be exactly related to the inside volume of the enclosure and the resonance of the loudspeaker cone. Unfortunately, commercial reflex cabinets are made with the reflex port dimensions designed for one specific loudspeaker.

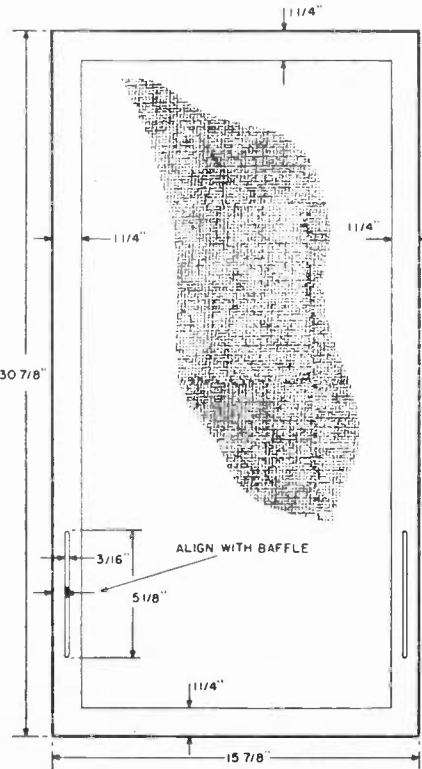
Which One Is Best?

That depends on many things—the size of the room in which it is to be placed, the acoustical behavior of the room, the particular loudspeaker or multiple speakers chosen, the location in the room, and the personal tastes of the audiophile. Any one of the three types of enclosures can, if well constructed and *properly designed for a specific loudspeaker*, give good performance. Is it possible to have all three types of baffling in one cabinet so that each may be compared performance-wise in the living room? Is

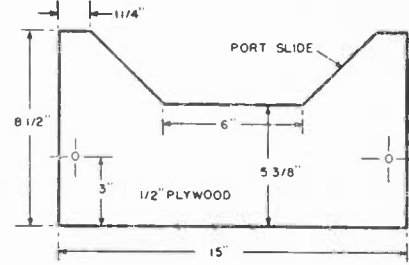
¹ RADIO & TELEVISION NEWS, Oct., Nov. 1953



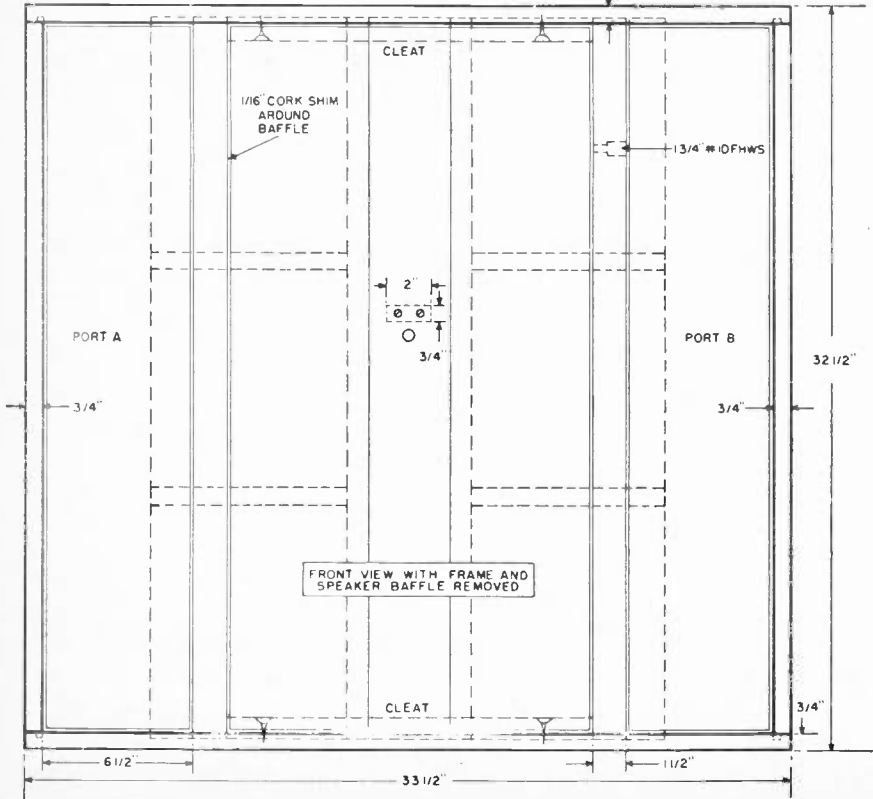
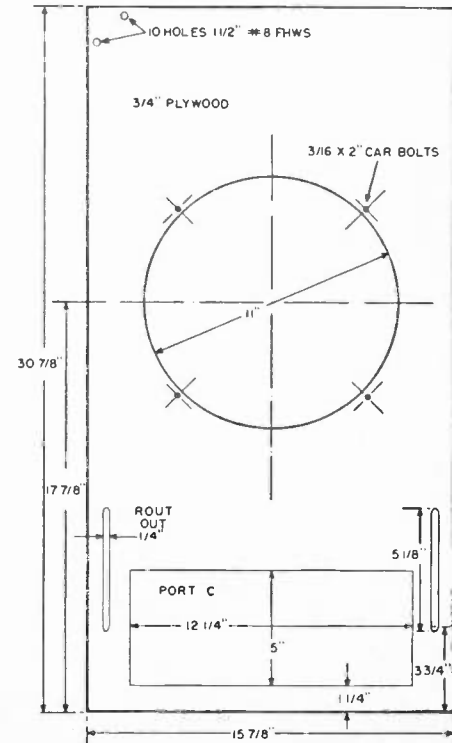
FRAME FOR BAFFLE



PORT C SLIDE



SPEAKER BAFFLE



there a cabinet that has *all three* baffle characteristics built in? There is one all-purpose loudspeaker enclosure. It is called the "Fold-a-flex" because it can be used as a folded horn, an infinite baffle, or as a bass reflex.

The "Fold-a-flex" design consists of three adjustable acoustical ports, as shown in the photo, and a built-in folded horn. The baffle can contain a mid-range horn, a high-frequency tweeter, or both. Space is also provided to mount crossover networks in two- or three-way systems. The "Fold-a-flex" cabinet permits a wide choice of mounting arrangements so that you may use a single wide-range 12" hi-fi loudspeaker. Later you can add one of the excellent tweeters and enjoy a two-way system. If you so desire, a third (mid-range) horn may be added. Thus, depending upon the amount you can invest in loudspeakers, various components can be added later.

The construction diagram of the new enclosure for 12" systems provides enough detail so that it may be assembled in any shop having the necessary power tools for the cutting of the various fins at their proper angles. The entire cabinet is constructed of $\frac{3}{4}$ " plywood, and the outside pieces may be any of the many attractive veneers available from most lumberyards. Ports A and B are hinged as shown, and must be carefully fitted to provide an air-tight seal when in the open or closed position. Note that there is a half-round vinyl gasket secured completely around the two side ports to

provide the necessary air seal. The adjustable slide which moves up and down in the slots is shaped so that the slide will seal off the slide openings when in the closed position. This prevents air leakage through the adjusting slots.

The baffle must be securely mounted to the all-around flange in such a manner that if it were a solid piece of wood it would make the cabinet completely air tight. The frame containing the grille cloth mounts in front of the baffle, as shown, and is used for decorative purposes only. It is simply shown for those who wish a professional-looking cabinet.

There are four braces employed, called bracket shelves. Their function is to prevent vibration of the interior fins. One of the advantages of the design is the lack of parallel-reflective surfaces. A minimum of padding will provide sufficient damping when *Kim-sul* or felt is placed both at the top and bottom of the inside compartment containing the cone.

Action—Folded Horn

There are two hinged doors, A and B, which are the side ports of the enclosure. They are completely sealed by means of gaskets so that no air can pass at their edges when placed in either of the two positions shown. When ports A and B are pushed inward, they become extensions of the inner horn structure and form the mouths of the folded horn. The enclosure may be placed against a flat

wall and will provide excellent bass response, or it may be placed in a corner which will provide even greater extension of the bass response. Port C is closed when the "Fold-a-flex" is used as a folded horn.

Infinite Baffle

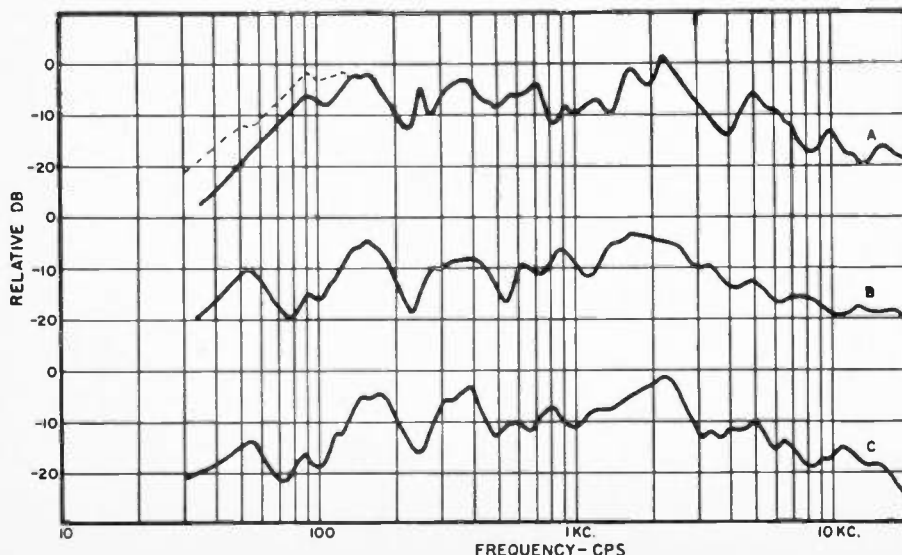
Ports A, B, and C are all closed which, in effect, results in a completely enclosed cabinet that is airtight. Approximately 6½ cubic feet of air loading is provided, which insures proper damping for 12" loudspeakers. This is one of the most satisfactory of all types of enclosure characteristics for single, coaxial, or triaxial loudspeakers, and has similar advantages to a loudspeaker mounted within a wall between two large rooms.

Bass Reflex

Many loudspeakers perform exceptionally well in this type of enclosure. Because the "Fold-a-flex" has an added feature of being fully adjustable to any loudspeaker, there are added advantages to this form of baffling. Ports A and B remain fully closed, the same as when used as an infinite baffle. The slide (in back of the reflex port C) is adjusted by loosening the two knobs and setting the slide to any position from a fully opened port to one which is fully closed. In practice, the port is adjusted by any of the familiar techniques in reflex port adjustment described in reference books.

Part 2. Performance of some popular high-fidelity loudspeakers in the "Fold-a-flex" enclosure and a description of a new, unique comparator technique.

Fig. 1. Response curves of the Stephens 103LX, 216, and 814H two-way system in the original "Fold-a-flex" for 15" speakers described in 1953. (A) Folded horn. Dotted trace is for corner mounted enclosure. (B) Infinite baffle. (C) Bass reflex. Humps at 150 and 300-400 cycles have equal response as a result of tuning reflex port.



THE enclosure for loudspeakers has a definite effect upon the reproduction that can be obtained from various systems. The damping effect on a moving cone of a loudspeaker has a direct control over the impedance characteristics of the voice coil. The amount of damping required will depend upon the behavior of a given loudspeaker mounted in its enclosure. The loading effect produced by a particular enclosure design can and does act as an acoustical brake to the piston-like excursion of a cone in the low frequency range. Horn type enclosures provide this loading advantage.

These are but two of the factors influencing the behavior of a loudspeaker. The leading manufacturers of hi-fi speakers have recognized the importance of the enclosure and have designed cabinets to meet the requirements of their own loudspeakers. Such compatibility has resulted in some very fine systems and has greatly simplified the problem of choice.

The audio enthusiast, technician, and the serious student, to a great extent, like to build their own enclosures and to study the behavior of several loudspeakers before deciding which is best for their listening pleasure. The all-purpose enclosure, described last month, was a direct result of this search for a "guinea pig" cabinet that could meet the demands for checking the performance of any loudspeaker contained in but one assembly—but having the essential characteristics of the most popular enclosures: folded horn, infinite baffle, and bass reflex. The "Fold-a-flex" loudspeaker enclosure¹ was the result. A single cabinet instead of three provided the necessary housing for the experiments.

Frequency response measurements were made under "free field" conditions in an outdoor area free from reflective surfaces. The setup included two calibrated microphones; a dynamic (*E-V 655*) and a crystal (*Shure 98-99*). The low impedance 655 was coupled through a high-quality broadcast transformer to match the 1 megohm input of the a.c. vacuum-tube voltmeter. Because the 98-99 crystal is designed to feed a load of at least 20 megohms it is necessary to use either a correction factor or to modify the circuitry. We chose the latter so that direct readings would result in plotting the curves. The 98-99 microphone has an internal capacitance of 1000 $\mu\text{f.d.}$ and the cable adds another 200 $\mu\text{f.d.}$ The total then becomes 1200 $\mu\text{f.d.}$ The microphone acts as a Thevenin generator with a constant voltage proportional to the sound pressure and a capacitance of 1200 $\mu\text{f.d.}$ connected in series.

Our trusty *Shure* "Reactance Slide Rule" shows that a 1 megohm load is equal to the capacitive reactance of 1200 $\mu\text{f.d.}$ at 130 cycles. The response of the v.t.v.m. will then be down 3 db at 130 cycles and will continue dropping at the rate of 6 db per octave below 130 cycles. Measurements were wanted down to 30 cycles. It was necessary then to connect a capacitor of 3000 $\mu\text{f.d.}$ across the microphone. This causes a 10 db drop in the output of the microphone but it has the advantage of eliminating a correction factor. The result is an essentially flat output down to 30 cycles.

Spacing between the microphone and the front of the enclosure was maintained at 36 inches. A *Heath* audio generator (calibrated against a *Hewlett Packard 200 C*) was used as the signal source feeding a *Fisher 70A* amplifier to drive the test speakers.

The response curves, Figs. 1, 3, 4, show the effect of enclosure characteristics. The advantages of a tunable bass reflex port are clearly shown in the curves. The hump (at 150 cycles) results from cabinet resonance produced by an infinite baffle. Tuning the port results in a reduction of the hump and produces two equal humps of lower amplitude which is exactly the effect wanted. The difference can be easily distinguished

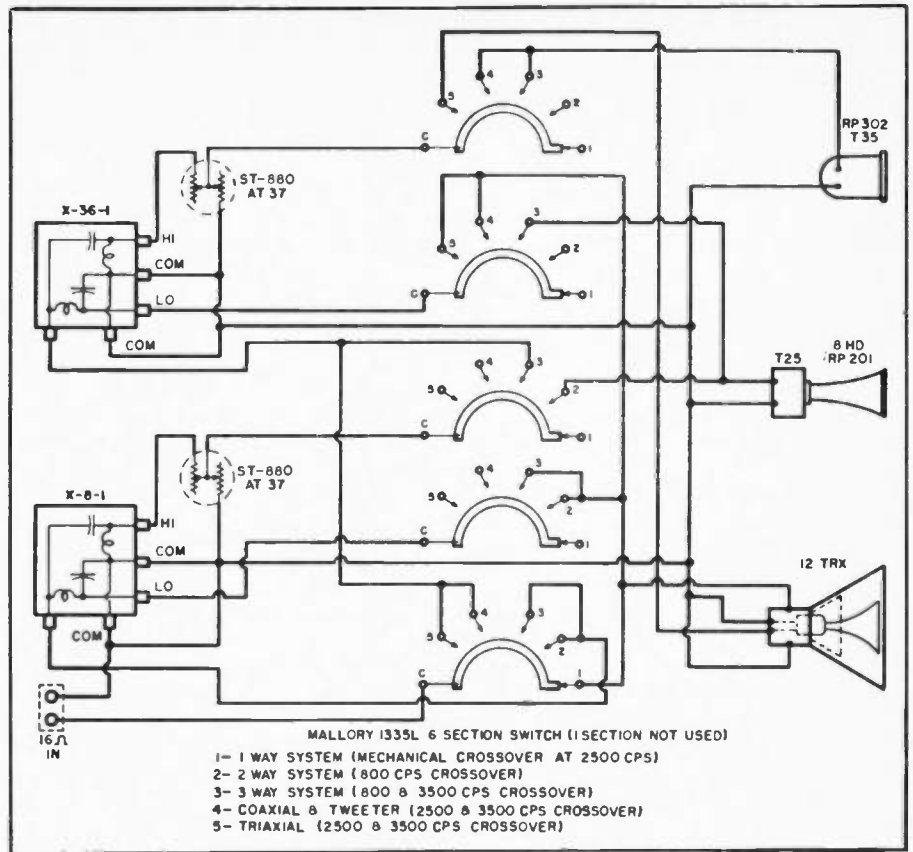


Fig. 2. Five distinct speaker combinations are switch-selected for demonstration purposes or for the study of loudspeaker behavior under various conditions.

when heard in the living room. Measurements made in a typical living room all show considerable improvement in bass response. When the enclosure is used as a folded horn, either on a flat wall or in a corner, even greater extension of the bass range is enjoyed.

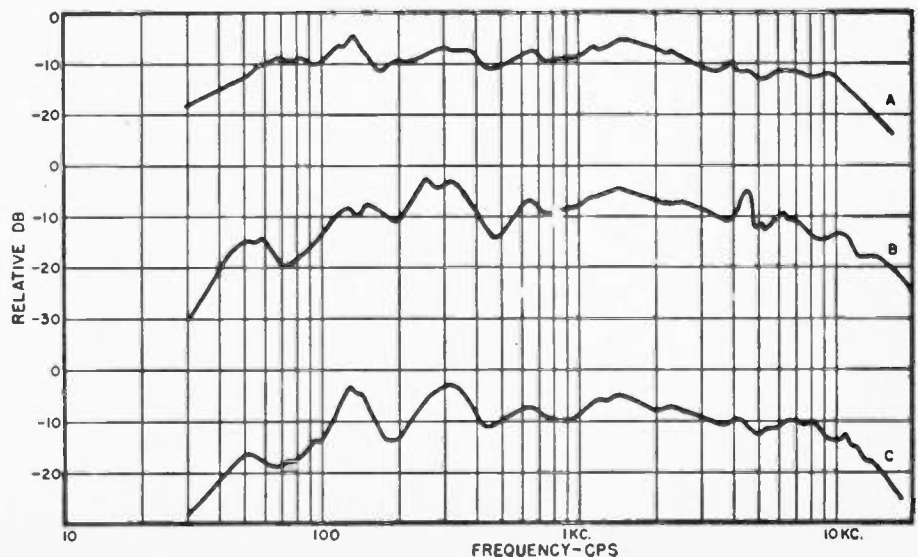
Another innovation was made to permit demonstration of the "Fold-a-flex" at the New York Audio Fair in October. Many of our readers will remember the remote control system

devised to show the wide difference in the performance of single and multiple loudspeakers under different conditions of baffling.

The circuit, Fig. 2, is designed for manual switch selection and is self-contained. Relays were substituted in our demo unit so that each function could be selected from across the room. The following components make up the system: A 12" triaxial loudspeaker with electrically inde-

(Continued on page 90)

Fig. 3. Improved damping and horn shape of the Model 12 enclosure is reflected in these curves of the Electro-Voice 111 two-way system for 12" low-frequency driver and high-frequency horn. (A) Folded horn. (B) Infinite baffle. (C) Bass reflex.



ELECTRO-VOICE

Aristocrat

By HOWARD SOUTHER
Electro-Voice, Inc.

Construction details and performance data on a folded corner horn enclosure. It will accommodate any high quality, low resonant frequency, 12-inch loudspeaker.

THE mind through the ear delights in the stimulus caused by sounds which are mathematically related. The generation of such sounds is called music. It is generally conceded that the widest variety of these sounds causes the greatest satisfaction; thus, we find that loudspeaker systems of the widest response range are the most pleasing, granting good source material of low distortion as an understood prerequisite.

Where space is limited there is no particular problem in achieving excellent response in the treble ranges. Generating mechanisms for producing the rapid, delicate pulses of the higher frequencies are inherently small in themselves. But not so the bass range, comprised of the first two octaves from 30 to 120 cycles-per-second.

The area near a sound generator, in this case the cone of a loudspeaker, is what engineers term a region of high acoustical impedance. To achieve useful transfer efficiency of motion into acoustical energy, we must build up considerable air pressures.

To deliver these sound pressure waves to the listening area, a region of very low pressure or low acoustical impedance, a transformer of some kind is required, just as it is in an electrical circuit.

The recognized scientific means of accomplishing this transformer action is through the use of a horn. This horn must expand in area at a constantly accelerating rate to accomplish its function, the ideal horn being one of infinite length and infinite mouth size.

The Horn Design

This last requirement almost stops the design project before it begins. But let us examine first the requirements of a suitable horn. Fundamental tones of even the largest bass instru-

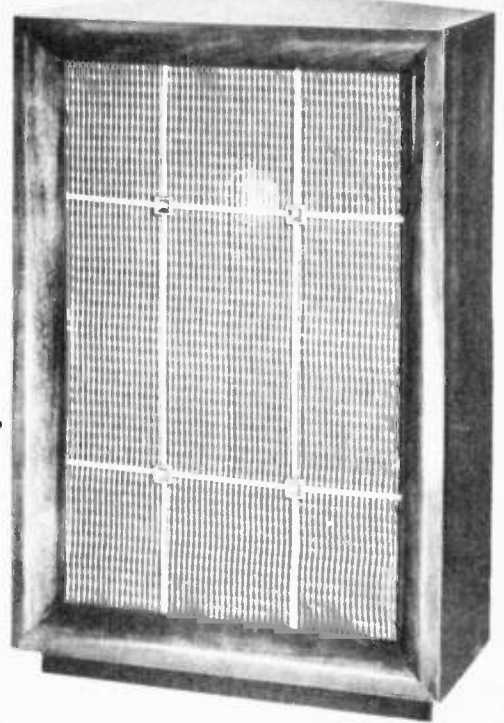
ments, such as monstrous drums and 16 foot organ pipes, start at 30 cycles per second. The mouth requirement for a horn capable of reproducing a 30 cycle tone is $\frac{1}{4}$ the wavelength of this tone, or 111 inches. For a 50 cps tone, this dimension decreases rapidly to 80 inches.

The next thing to consider is the length of the horn. The formula governing horn design says that the taper rate, or the flare, governing the expanding cross section of the horn, shall double every $1\frac{1}{2}$ feet of its length in order to reproduce a 30 cps tone at the mouth (whose cross section we have already computed to be 111 inches).

The Design Takes Form

Our design still has impossible dimensions for the living room; something having a length of 6 to 10 feet, according to the throat size we select at the start of the horn, and a mouth 10 feet across! On the other hand, certain things are in our favor:

Ideally, the lowest tones in frequency lend themselves easily to propagation in a closed cavity of a size such as our living room. Examination reveals a partial horn available in the corner of the room, the mouth of which is in most cases more than 111 inches across! Some years ago, Paul Klipsch, the noted acoustics authority, seized upon the idea of housing only the throat of the required horn in a furniture cabinet and placing it in this corner. In the design of the "Aristocrat," we find that by keeping the driver unit itself small, we have such a throat assembly of very compact proportions. There now results a clean, extended low-frequency response to the 30 cps region, surprisingly free from peaks and valleys in its characteristic curve. But, although the range is well extended and satis-



The E-V "Aristocrat" enclosure. The performance data covered in article was obtained using the E-V SP12-B "Radax" unit.

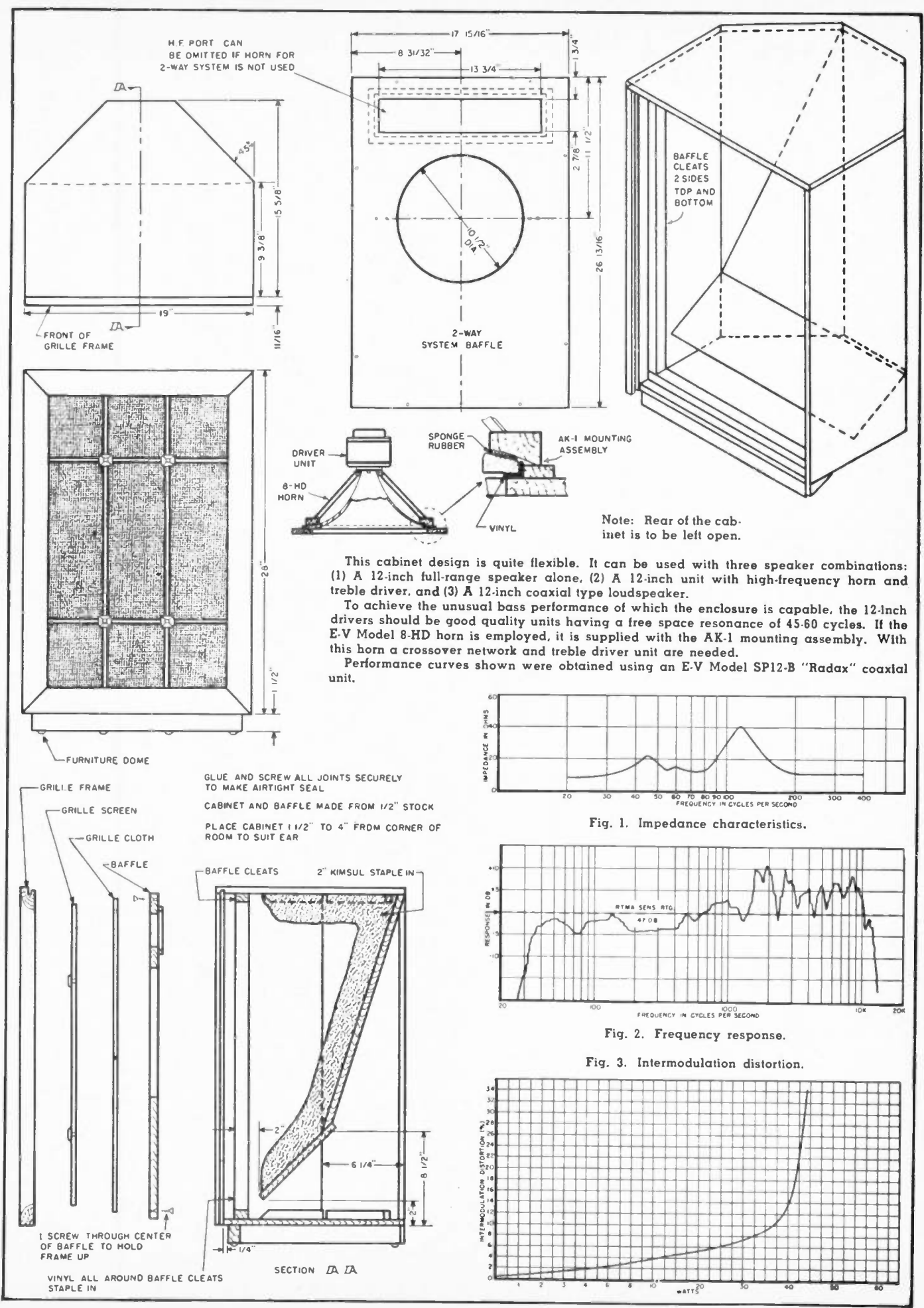
factory, the efficiency in the first octaves is still too low to accomplish a pleasing musical balance. This is true, in the main, because our driving cone is only a piece of parchment, and a far cry from the ideal acoustical requirement for a piston of infinite lightness and infinite rigidity.

Building the Efficiency

By exploiting a phenomenon involving acoustic resonance, the efficiency in the bass range may be augmented as much as 4 to 8 times. Observe the cross-sectional drawing of the "Aristocrat": By utilizing the reactance of the small air mass directly behind the speaker, in conjunction with the high compliance or capacitive factor of the specialized driver cone, the combination can be used to reinforce the sound over a four-octave band with the large air mass, due to the horn, which is presented by the corner.

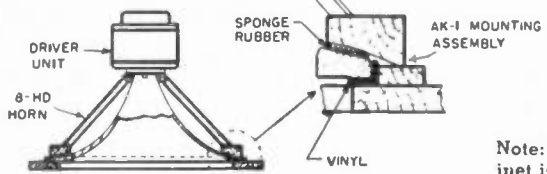
Performance

If the "Aristocrat" is carefully constructed, and care is taken to effect a complete seal of the front baffle board to prevent air leaks from front to back, rather startling bass performance will be realized. Response range will be as shown in the curve (Fig. 2) revealing a response within 5 db of flat to about 30 cps. This operation is supported by evidence disclosed in the impedance characteristic (Fig. 1) denoting a satisfactory reactive component in the voice-coil system well into the first octave. The high impedance at 120 cps is fortunately not reflected adversely in the frequency response curve.



H.F. PORT CAN BE OMITTED IF HORN FOR 2-WAY SYSTEM IS NOT USED

BAFFLE CLEATS 2 SIDES TOP AND BOTTOM



Note: Rear of the cabinet is to be left open.

This cabinet design is quite flexible. It can be used with three speaker combinations: (1) A 12-inch full-range speaker alone, (2) A 12-inch unit with high-frequency horn and treble driver, and (3) A 12-inch coaxial type loudspeaker.

To achieve the unusual bass performance of which the enclosure is capable, the 12-inch drivers should be good quality units having a free space resonance of 45-60 cycles. If the E-V Model 8-HD horn is employed, it is supplied with the AK-1 mounting assembly. With this horn a crossover network and treble driver unit are needed.

Performance curves shown were obtained using an E-V Model SP12-B "Radax" coaxial unit.

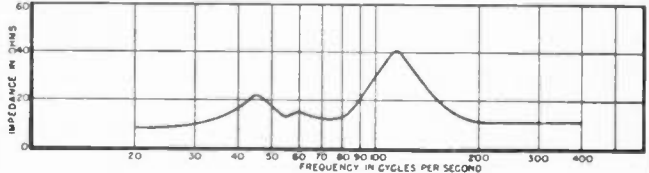


Fig. 1. Impedance characteristics.

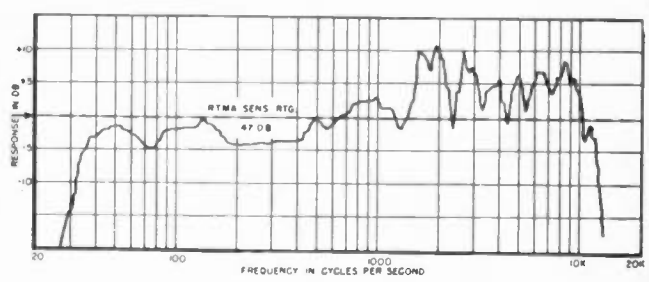


Fig. 2. Frequency response.

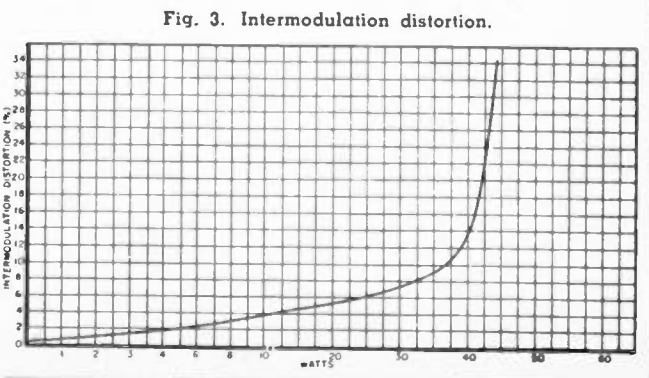


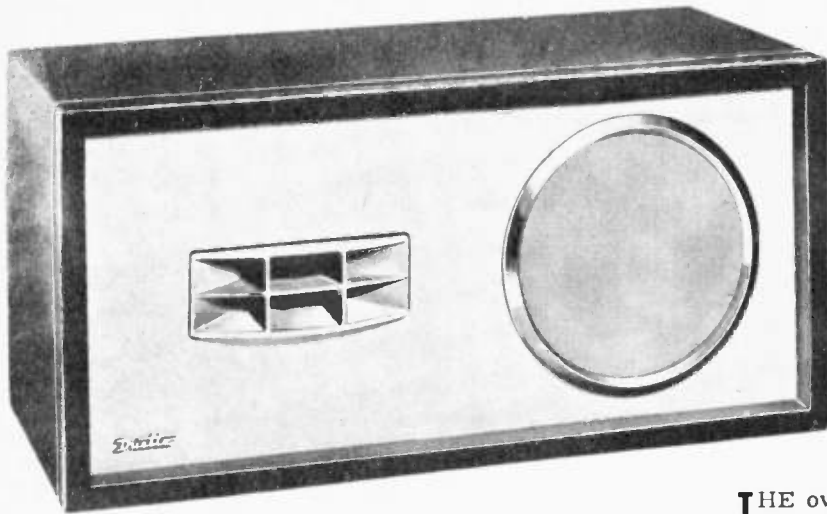
Fig. 3. Intermodulation distortion.

GLUE AND SCREW ALL JOINTS SECURELY TO MAKE AIRTIGHT SEAL
 CABINET AND BAFFLE MADE FROM 1/2" STOCK
 PLACE CABINET 1 1/2" TO 4" FROM CORNER OF ROOM TO SUIT EAR

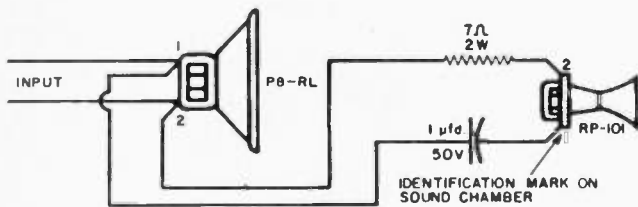
- GRILLE FRAME
- GRILLE SCREEN
- GRILLE CLOTH
- BAFFLE
- BAFFLE CLEATS
- 2" KIMSUL STAPLE IN
- 1" SCREW THROUGH CENTER OF BAFFLE TO HOLD FRAME UP
- VINYL ALL AROUND BAFFLE CLEATS STAPLE IN

SECTION A-A

THE DU-201 *Duetto*

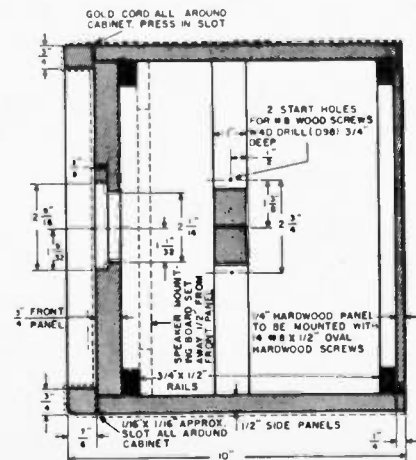
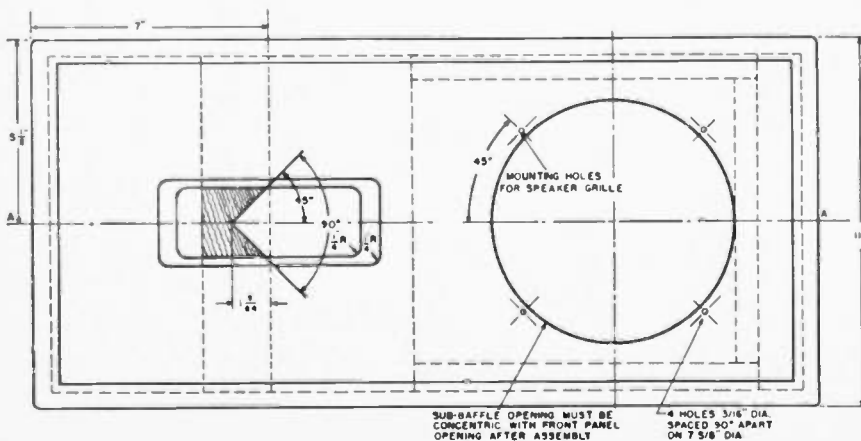
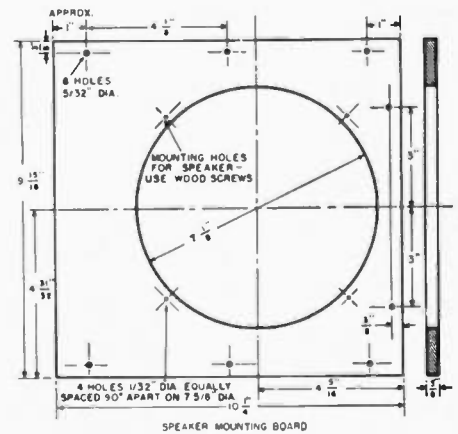
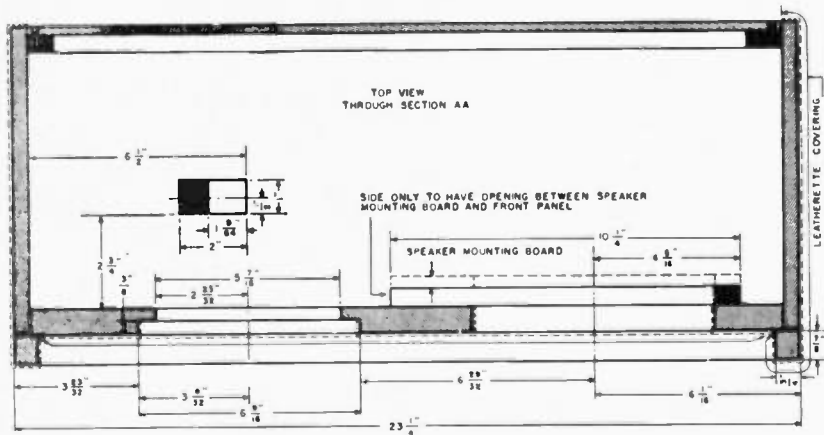


Designed by Jensen Mfg. Co., this unit is compact and easy to build. Its sound reproduction is excellent. Their P8-RL 8 inch speaker and RP-101 horn are used.



THE over-all construction of this cabinet is relatively simple. All dimensions are clearly indicated. One point that bears specific mention is that its theoretical operation is somewhat similar to that of the R-J cabinet in that the waves emanating from the rear of the speaker emerge through a port spaced between the speaker mounting board and the front panel of the cabinet. In this particular design the speaker mounting board is placed $\frac{1}{2}$ inch from the front panel and the left side, looking at the top view of the diagram, is open, providing a port for the sound waves.

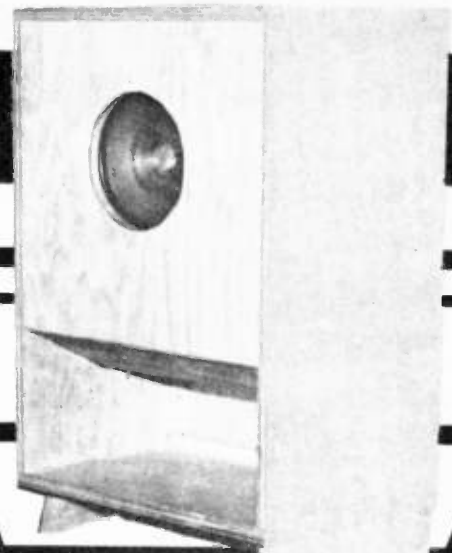
-30-



The "DIFFUSICONE - 8" ENCLOSURE

By
ABRAHAM B. COHEN
University Loudspeakers, Inc.

Over-all view of enclosure for the University "Diffusicone-8", shown without finish.



Designed for an 8-inch speaker, this cabinet received considerable attention at last year's N.Y. Audic Fair.

OPTIMUM performance of a loudspeaker is obtained when it is mounted in the proper enclosure. The enclosure and the loudspeaker must be matched to each other. A 6-inch speaker doesn't require a seven cubic foot cabinet, nor will a two cubic foot cabinet suffice for a 15-inch woofer. The University cabinet for the "Diffusicone-8" was designed to provide the proper load match for this speaker. Since this speaker is a high-efficiency, extended-range radiator, it requires a cabinet that will:

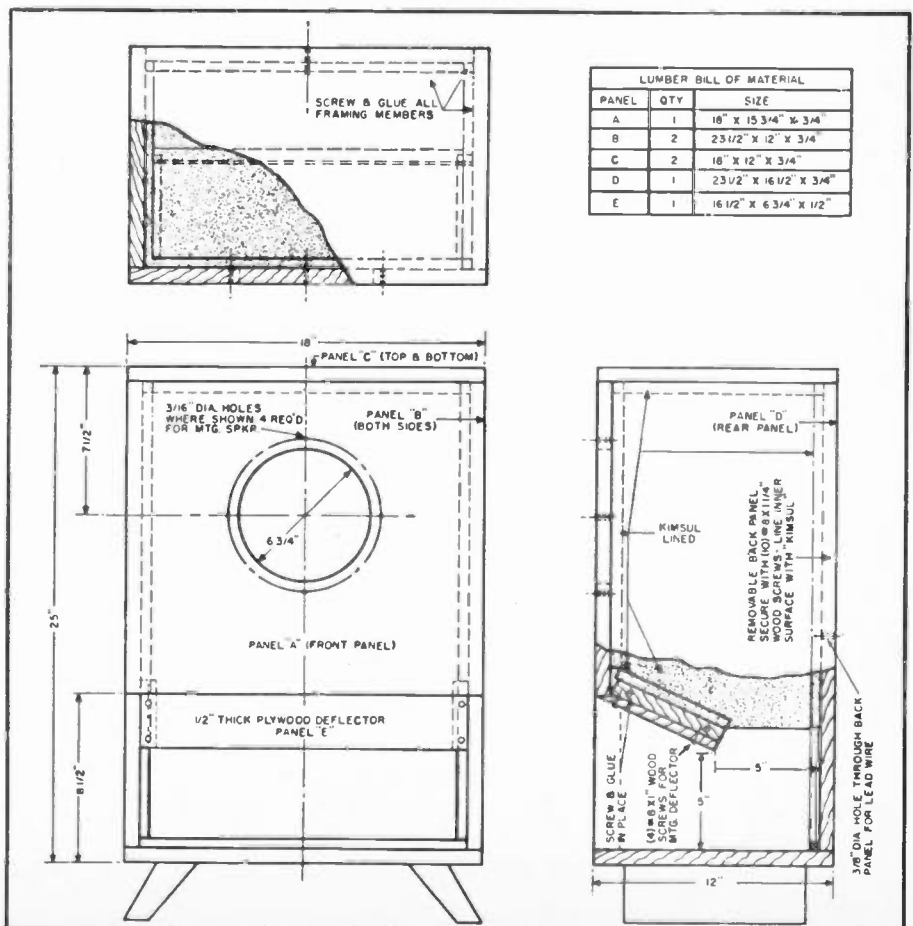
- (a) allow its extended high frequencies to spread into the room without obstructions due to bends or folds in the cabinet;
- (b) do justice to the high-level, low-frequency output of which the speaker is capable; and
- (c) be modest in size compatible with the performance requirements of the "Diffusicone-8"

Requirement (a), calling for an unobstructed path for the high frequencies, makes it mandatory that the enclosure and the speaker constitute a "direct radiator" combination, rather than a horn type. The final design of this enclosure resulted in essentially a bass reflex type of cabinet with the important addition of a horn load applied to the port of the enclosure. Thus, what looks like an exceptionally large port for a bass-reflex cabinet of this size is actually the mouth of the horn which feeds from the proper size port located at the rear bottom of the enclosure. The addition of the horn to the port permits smoother low frequency radiation than is possible from a simple port aperture. This condition holds because the efficiency of radiation of low frequencies rises as the size of the radiating opening becomes larger. Thus, since the mouth of the horn is more compatible, dimensionally, with the wave length of the low frequencies than is the smaller sized port, the horn adds a considerable measure of improvement to the performance of the bass reflex cabinet.

Inasmuch as this horn mouth lies at the bottom of the cabinet structure, the floor of the room acts as a natural extension of the horn walls which effect still further aids the low-frequency efficiency. The relatively small size of the enclosure allows it to lie flat

against a room wall or in the corner. Due to the high efficiency of the "Diffusicone-8," the sound pressures developed within the enclosure for the lower frequencies require that the cabinet be constructed of heavy rigid plywood paneling with all sections

Mechanical details of cabinet construction. Although any 8-inch speaker can be used, its design and performance characteristics are based on the use of the University "Diffusicone-8" speaker. The cabinet is simple to build and is, fundamentally, a bass reflex cabinet with a horn flare. The entire upper section of the cabinet should be Kimsul lined. All inside corners are reinforced by using 1/2" square wood strips glued and screwed to the sides of the cabinet. Dimensions for the legs are not given and can be designed to suit the builder or omitted.



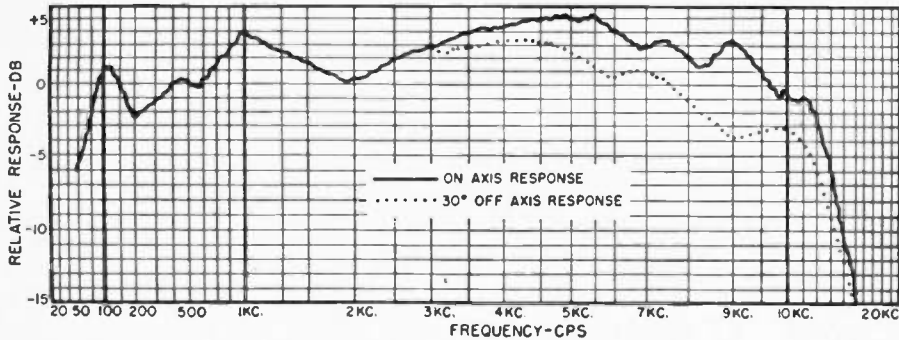
well glued and screwed. Internal damping is provided by "Kimsul" insulation.

This cabinet has created such wide interest during public demonstrations that it has been exhibited without the decorator's finishing touch (as in the photograph) in order to satisfy the

countless listeners that there was only one 8-inch speaker working during the demonstration. The home constructor will of course want to put the finishing touches to the cabinet as best suits his needs. The important factor concerning the acoustics of this finish is the grille cloth that covers the face

of the speaker. The University "Diffusicone-8" employs the patented high-frequency diffusing element which disperses the high frequencies over a wide angle. Thus, whatever grille cloth is used should be selected more on the basis of acoustic transparency rather than artistic appearance. A good rule of thumb to use in selecting a grille cloth is that the cloth be about 50% optically transparent. If these precautions concerning the physical construction of the cabinet are followed, then the combination of the enclosure and the "Diffusicone-8" will give clean strong bass reproduction with well balanced middles; and if the proper grille cloth is used the top end of the spectrum will be completed through the extended-range, wide-angle, high-frequency performance of the diffusion system.

Response frequency characteristics of enclosure using the "Diffusicone-8" speaker.



All-Purpose Enclosure

(Continued from page 85)

pendent tweeter, a mid-range horn, a small tweeter, a crossover network (800 cycles) and another at 3500 cycles. The two networks should have a common built-in connection between input and output. The switch is a 6-section (1 not used), 5-position rotary type.

Our demo unit contained an *Electro-Voice* 12TRX triaxial, a T25 driver,

and 8HD horn and the *Jensen* RP302. The crossover networks are *E-V* X-8-1 and X-36-1. Level controls permit setting the mid-range horn and the tweeter to best balanced listening. The following are the functions available as selected by the switch:

Position (1): 1-way wide-range system. Here, the audio feeds directly to the voice coil of the 12" duo-cone. Its built-in mechanical crossover at 2500 cycles affords extended single cone coverage.

Position (2): A 2-way system. This

any of the power tubes, such as 6BG6 frequency horn to existing 1-way systems. Audio feeds to the 800-cycle network and utilizes the 12" cone as the woofer (800 cycles and below) and the horn for 800 cycles to the h.f. limits of the horn.

Position (3): A 3-way system comprising low frequency woofer, mid-range horn (800-3500 cycles) and u.h.f. tweeter (3500-15,000 cycles).

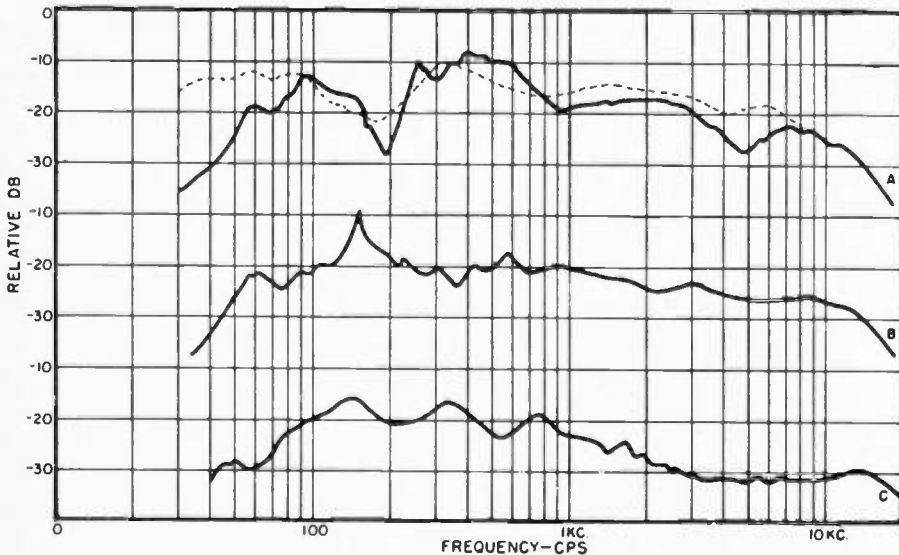
Position (4): Coaxial (2500-cycle mechanical crossover) and tweeter (3500 cycles). This demonstrates the effect of adding a tweeter to existing coaxial speaker systems.

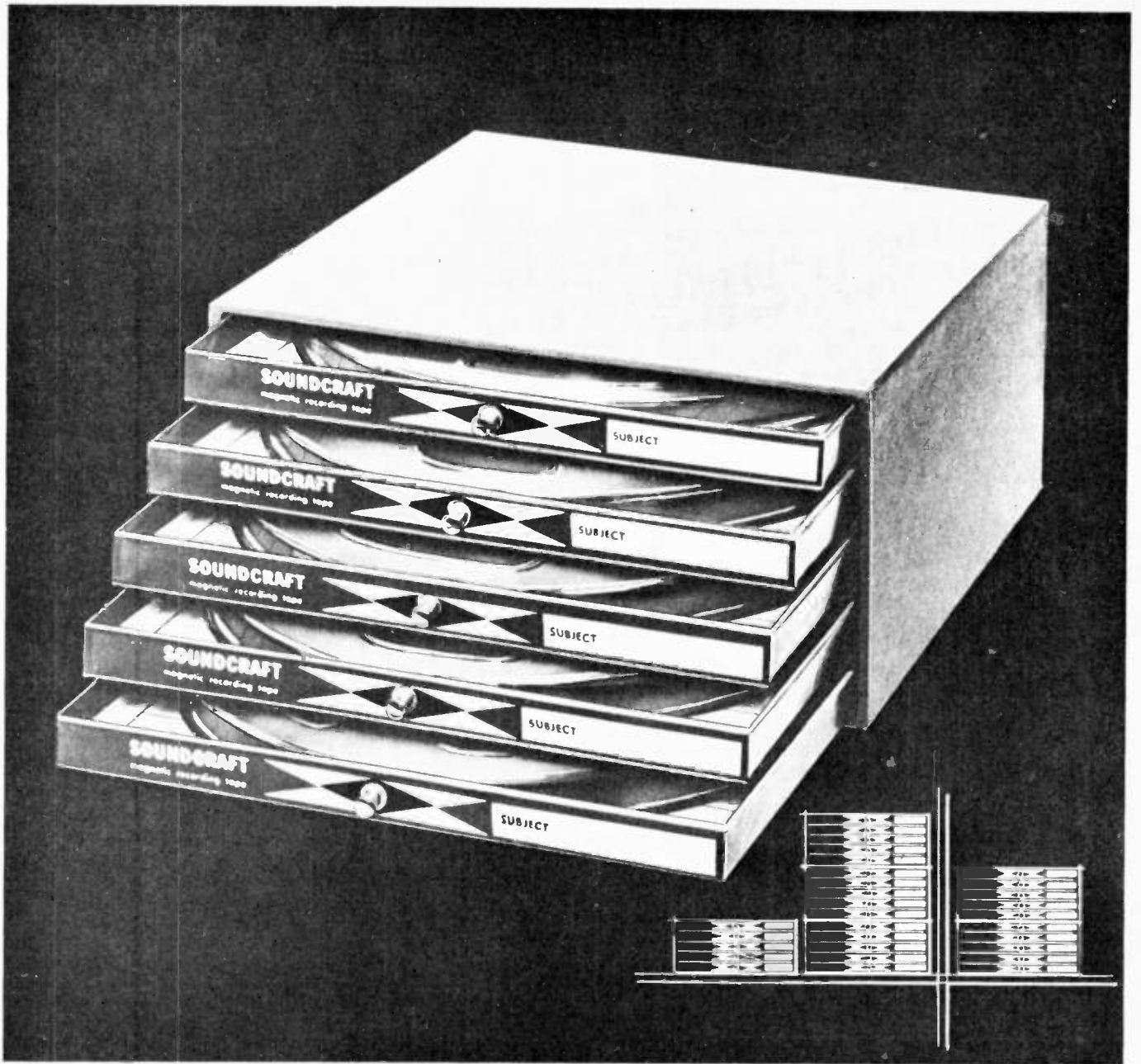
Position (5): Triaxial. Showing the advantages of single point-source sound. The self-contained tweeter in the 12TRX is now employed.

Thus, five different combinations of units are available at the turn of a switch. Because the "Fold-a-flex" enclosure may be adjusted for three types of baffling it is possible to hear and to analyze fifteen distinct performance effects.

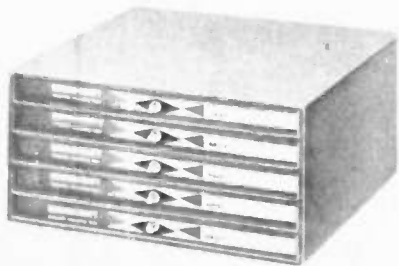
This may seem unnecessary from a practical sense. It probably is—but such a system has its place in the modest hi-fi showroom for consumer demonstration and does reduce inventory requirements for cabinets of different types. And it permits a convenient means for the study of the behavior of loudspeakers under widely varied conditions.

Fig. 4. Response of the Jensen "Tri-Plex" system in the original enclosure. Dotted trace shows effect of corner mounting in a well-draped living room. (A) Folded horn. (B) Infinite baffle. (C) Bass reflex. See author's discussion in article.





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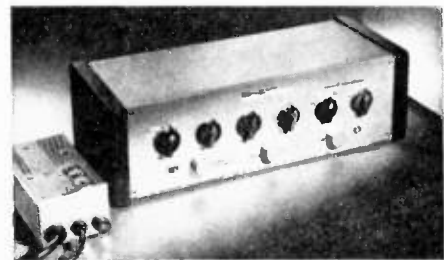
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CHAPTER 5

SELECTING HI-FI EQUIPMENT

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A REVIEW

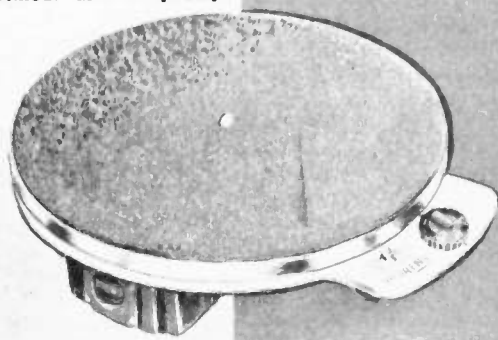
of New Record Players

One of the basic units for any high-fidelity music system is, of course, the record player. Here are some of the more recent models.



Swiss-made CBA-83 changer being offered by Thorens. It is of direct-drive motor design.

The Thorens E-53PA professional-type turntable. It is supplied without arm or pickup.



The Model CD-43 record changer. It has a fine-tuning adjustment knob.



AMONG the new record-playing devices which have been released to the public in the past few months are those pictured and described here. Most of these new units offer interesting and unusual features which should be called to the attention of both the audiophile and the audio service technician.

Collaro Transcription Unit

A new British transcription turntable, the *Collaro Model 2010*, is being introduced to the American market by *Rockbar Corporation*.

Driven by a four-pole, dynamically-balanced, hum-shielded induction motor, the unit is designed to operate on all record speeds, 33 $\frac{1}{3}$, 45, and 78 rpm. The motor is shock-mounted by means of lateral springs which effectively damp out mechanical vibration. The turntable itself is cast and machined (weighs approximately 8 $\frac{1}{2}$ pounds) and is so formed that the greater part of its weight is in the rim for flywheel effect. The material is non-magnetic.

The turntable rotates on a 3 $\frac{1}{2}$ -inch long steel shaft which rides in a self-lubricating bearing. The vertical thrust is taken by a single steel ball. There is minimum frictional loss. The motor spindle is fitted with a 3-step pulley which couples to a single idler which, in turn, drives the inner rim of the turntable. Speed may be selected or changed at any time.

The turntable comes complete with a low-mass, non-resonant arm which houses the company's transcription pickup—a crystal cartridge with two mechanically isolated sapphire styli which are used turnover-wise for either standard or microgroove records. The nominal output of this cartridge is suitable for use with conventional preamplifiers.

The Presto "Pirouette" three-speed turntable. It is available with a four-pole induction or a hysteresis synchronous motor.





Two of the new V-M Corporation models. The Model 1250 is shown above and the Model 1200 at the right. Both units feature four speeds (33, 45, 78, and 16 "talking book" speeds). The three standard speeds are automatic, the 16 rpm is obtained manually.

The Model 2010 will accommodate all discs up to 16" and the pickup arm is designed to give good tracking on all sizes. The price, complete with cartridge and pickup arm, has been tentatively set at \$72.00 (east of the Rockies).

Also of interest is the fact that the *Collaro Model RC54* changer, which has been on the market since late last year, is now being supplied with a pre-cut mounting board and with a power cord and amplifier connecting cables attached, at no increase in cost.

Component Corporation Console Turntable

Component Corporation is marketing a transcription console version of its belt-driven "Professional" turntable as the Model 70.

A double shock-mounted, continuous-duty, constant-speed induction motor turns a three-step motor pulley, accurately machined, in its own bearings. An endless belt couples the proper pulley to the turntable's outer rim to drive it at 33½, 45, or 78.26 rpm. An expanding collet spindle accurately centers discs with oversize center holes.

The Model 70 has what is said to be the industry's heaviest turntable (25 pounds machined cast steel) which reduces rumble to -70 db, wow and flutter to .05% and speed variation to .25%.

Completely free from metal-to-metal contact, the turntable runs in a nylon sleeve and on a single ball-thrust bearing. Tempered steel, felt-damped springs provide over-all shock mounting and the console may be accurately leveled by rotation of these springs.

A thick cork cushion on the turntable protects record surfaces. Instantaneous cueing is provided by slipping the record (the turntable accommodates 16" transcriptions and 17¼" masters). There is ample room for mounting two or more arms on the console surface and accessory space at the rear of the console for mounting standard 19½" rack panels up to 22" high. The console is priced at \$295.00, FOB, Denville, N. J.

Ercona Intermix Changer

The Electronic Division of *Ercona Corp.* has a new automatic record changer which will handle 12", 10", and 7" discs intermixed without wow, hum, or rumble.

The "Dekamix" will operate at all three speeds. It has a single-phase, four-pole asynchronous motor with auxiliary phase displaced by a capacitor. Operated at 110-125 volts, 60 cycles, a.c., power consumption is 10 watts. This same unit is also available for d.c. and 6 volt operation on special order.

The crystal cartridge that comes with the changer is of the turnover type and has two sapphire needles. The plug-in head will accept standard magnetic cartridges. A muting switch short-circuits the cartridge during change cycle.

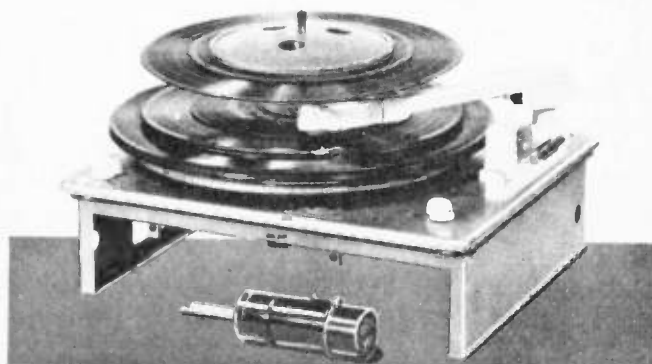


Ercona Corp.'s "Dekamix" intermixing record changer. It will handle 33, 45, and 78 rpm, 12", 10", and 7" discs intermixed.



The Collaro Model 2010, three-speed turntable. It features a four-pole, dynamically-balanced, hum-shielded induction motor.

The German-built "Rex AA" record changer being distributed by Fenton. It will intermix any size records between 6" and 12".



Audiophile net for the "Dekamix" is \$49.95 including the crystal pickup, turnover-type dual sapphire styli, and spare pickup shell.

German-Built "PE Rex AA" Changer

The U.S. distribution of the German-built "Rex AA" record changer is being handled by *Fenton Company*. Specially designed to accommodate American cartridges, this new changer will intermix any odd size records between 6" and 12".

The mechanism consists of a shock-mounted, four-coil capacitor motor. Even the narrow frequency band caused by the vibration of the motor drive is eliminated through the damping of double chassis, suspended on factory-tuned springs.

The change-cycle mechanism is of new design. At the end of a record it is automatically activated whether or not the record is provided with fast-finishing grooves. To assure silent operation during playing, the driving gear of the change cycle retracts after each cycle change. Thus only the drive mechanism engages the turntable during the playing cycle.

With its long, small vertical mass, non-resonant pickup arm, and friction-free horizontal bearing, the unit provides perfect tracking on any record irrespective of its irregularities. The arm weight is easily adjustable through a knurled knob on the side of the pickup arm. As an added feature, each unit is supplied with a short manual spindle so that single records can be played. The unit will shut off automatically even when used as a player rather than a changer.

The "Rex AA" comes equipped with two empty plug-in shells or with one plug-in shell either with the PE-8 crystal or P-600 series magnetic cartridges. These cartridges are now supplied with standard American mounts. This changer is priced at \$59.50.

The Presto "Pirouette"

Under the tradename "Pirouette," *Presto Recording Corporation* is now marketing a three-speed turntable which will handle 33 $\frac{1}{4}$, 45, and 78 rpm discs.

The new unit replaces the company's Model 15 in the line. Like the T-15, it has a 12" diameter cast-aluminum turntable. As an added feature, however, the turntable carries a 45 rpm

disc, permanently attached to the turntable spindle which retracts under the surface of the turntable when not in use.

The drive system utilizes three rubber idler wheels, one for each of the three turntable speeds. The idlers are interchangeable so that one spare may replace any one of the three operating idler wheels.

A single control lever, operating in a horizontal plane, selects the correct speeds or shuts off the mechanism. The control locks positively in each of the three speed positions and, in the "off" position, retracts the idler from the drive shaft to prevent flats from developing on the rubber surface.

This model is available with either a standard four-pole shaded induction motor at \$53.50 or with a hysteresis synchronous motor at \$108.00.

New Swiss-Made Units Introduced by Thorens

Thorens Company is currently introducing several new Swiss-made units which feature enhanced performance and new operating convenience.

The new units are powered by a direct-drive motor utilizing a separate gear for each standard speed. Operating convenience is enhanced by the adoption of a dial action control knob for selecting the three standard speeds. Concentric with this dial is a fine-tuning knob which permits "exact" pitch adjustments within a 5% latitude above and below each of the standard speeds, during audition. This feature is of special interest to the serious musician blessed with perfect pitch.

The CD-43 record changer and the CBA-93 "Audiomatic" record player both have provision for manual operation. A flick of a switch disengages the automatic trip mechanism, allowing greater flexibility. The CD-43 is \$93.75 audiophile net while the CBA-93 is priced at \$67.50 for audiophiles.

The company's professional-type turntable has been designated as the E-53PA and includes the same operational innovations as the changer and player. The turntable is offered at \$60.00 audiophile net without the tone arm or cartridge.

V-M Corporation Changers

One of the newest record changer mechanisms in the *V-M Corporation*

line is the Model 1200 which will handle three speeds (33, 45, and 78 rpm) automatically and operate at the new 16 rpm "talking book" speed manually.

The changer has a new patented-drive, four-speed motor which insures constant speed at all times. The low-torque mechanism offers minimum wow and silent, rumble-free performance. A new three-spring mounting provides absolute stability and balance.

The die-cast aluminum tone arm is balanced for minimum needle pressures as specified by the needle or cartridge manufacturers. The underside of the tone arm is calibrated to allow exact adjustment. A new anti-skate mechanism positively controls the motion of the tone arm after landing, preventing skating even under severely tilted conditions. This same mechanism also allows a point-thrust bearing on the tone arm to reduce side wear on record grooves thus enabling lighter needle pressures for proper tracking to substantially reduce record wear.

Another convenience feature of this model is the "Easy-Lift" record support arm for front loading facility. The Model 1200 is available with a *G-E* variable reluctance cartridge as well as with dual-needle ceramic cartridges. Both versions can be purchased with a matching pan for open shelf or table-top use. The Model 1200 is \$46.50 list.

A second model in *V-M's* line is the Model 1250. Like the Model 1200, it will handle four speeds, the 33, 45, and 78 rpm discs automatically and the 16 rpm discs manually. The mechanism specifications are the same as for the Model 1200 but, in addition, the Model 1250 includes a convenience outlet

In addition, the changer incorporates a "Siesta Switch" which turns the entire mechanism off; including the amplifier, after the last record has been played. An auxiliary output, with 8 ohms impedance, is provided for use with external speakers. This unit is \$59.95 list.

Three other units are also available in the firm's current line: the Model 1275 4-speed unit, housed in a portable case, which retails at \$79.95; the Model 1285, another 4-speed unit in a table cabinet which can be converted into a console with the addition of optional legs, at \$99.95; and the low-priced Model 155 portable at \$49.95, which also offers 4-speed operation and several other features.

Buying a Hi-Fi Amplifier?

Before you buy—make sure you know what features in a power amplifier are most important.

EVERY industry has its phases, one of which is the condition where individual manufacturers feel compelled to include some features that do not contribute anything useful to the operation of their products. Because his competitors are advertising these features and the public is insufficiently enlightened as to their value, the individual manufacturer decides to include them in his line to maintain his competitive position. Eventually the public is educated to the true value or worthlessness of such features. This may be started, either by a truly courageous manufacturer who is prepared to stake his livelihood on introducing the simplified version to the public and at the same time persuading them that they do not require all the extra features, or by some individual, with somewhat less at stake.

The audio amplifier industry seems to be in somewhat this position at present. For some time manufacturers have been producing amplifiers to better and better specifications. Individual manufacturers, when discussing their policies for the design of next year's products, are faced with the problem of not only making their amplifiers perform well, but also of being able to quote truthful figures that compare favorably with those quoted by competitors.

The newcomer to audio thus encounters real confusion when he decides to choose an audio amplifier. Naturally enough, as in selecting any other commodity, he starts by consulting catalogues, with the idea of sorting out a short list of the best from which to make a final choice at demonstrations. Right here he encounters his first problem. What do these specifications mean? How much power do I really need? How good does the frequency response really have to be? How little distortion must I have, if I'm really going to have clean-sounding reproduction?

These are the "how good?" problems. In addition to these, he has to make sure that the amplifier will fit in with the loudspeaker system he proposes to use, and also work satisfactorily from his pickup, tuner, or whatever he wishes to play through the amplifier. So let's take some of these questions in order.

How Much Power?

The kind of answer to this question that one will get by asking different people varies widely. This is largely due to the wide interpretation of what is loud or what is quiet.

If you look at a table of loudness figures you will find that the range covers 130 decibels from the threshold of audibility to the threshold of pain. This represents a power ratio of 10,000,000,000,000! A comfortable listening level, corresponding to average conversation or a program heard in an average auditorium, is about 50 decibels above the threshold of hearing. In the average living room, with a loudspeaker system of average efficiency, this can be achieved with an average power of about 300 milliwatts.

The average intensity of a sound is considerably below the maximum peaks, which occur occasionally in the same program material, whether we are considering speech or music reproduction. An amplifier must have sufficient margin to cover the peaks without going into distortion. Allowing a good margin for peak overshoot, 10 watts should be ample to cover an average power of 300 milliwatts, with the highest possible transient peaks likely to be encountered.

However, the interesting point is that the loudness scale covers such a tremendous power range. The difference between what



Use this guide
in making
your selection

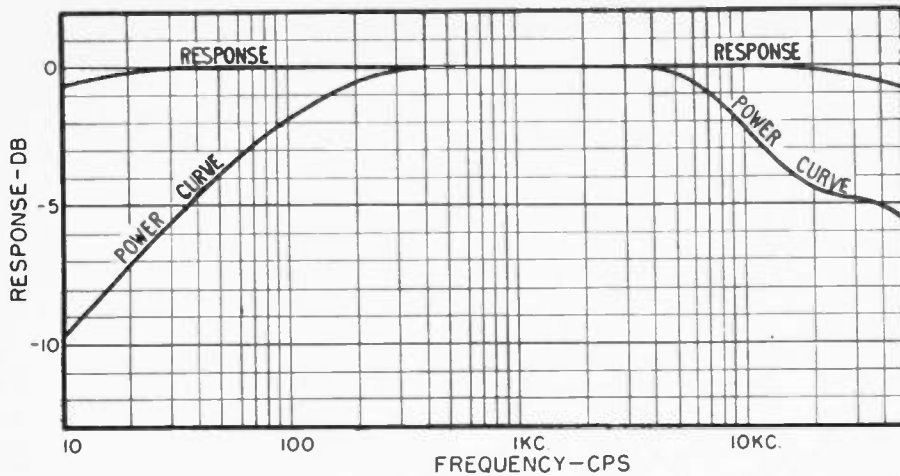


Fig. 1. Comparison between frequency and power response curves for a typical amplifier. The frequency response is plotted at a comparatively low level, where there is no risk of running into distortion. The power curve shows the maximum output at each frequency with a specified standard of distortion. Refer to discussion in text.

some people would call average conversation and the way others normally converse—which might more accurately be called shouting—can be at least 10 db. An increase in power level of 10 db represents 10 times as much power. This means that, to produce a consistent level, with head-room for peaks, equivalent to some people's "loud conversation," we should need a 100-watt amplifier instead of a 10-watt unit.

The difference between the two is that the 10-watt amplifier, turned up to a point where it would never overload on the highest peak, will provide us with a comfortable listening level in our own living room, while the 100-watt amplifier will provide the neighbors two doors away with a comfortable listening level, as well as ourselves, that is if the loudspeakers will handle that much. So, assuming that our definition of "being neighborly" means we allow our neighbor to select his own program on his own equipment, and not rely on listening to ours, we should not need more than about 10 watts to get the kind of level we need.

Popular amplifiers have outputs extending up to about 50 watts, but the reader is cautioned against a common mistake of thinking that a 50-watt amplifier turned up to give full output will sound 5 times as loud as a 10-watt amplifier turned up to full output. When we consider that 50 watts is only 7 db louder than 10 watts, and that 3 db is generally recognized as the smallest detectable change in loudness, we realize that a change from 10 watts to 50 watts represents a little over twice the smallest change in loudness that can be definitely noticed. The important thing to realize is that, although this change may not be very noticeable in our own house, it can represent the difference between inaudibility and quite an annoying audibility in our neighbor's house. Keep this in mind when planning a system.

If you are one of those fortunate people who live where you can use 50 watts without annoying the neighbors, and if you also like to have your or-

chestral music so the crescendos really sound like crescendos, then by all means get a 50-watt amplifier. But if you live in an apartment, or some place where it is good to consider the neighbors, then you are advised to buy a smaller amplifier, and automatically safeguard yourself against giving unwitting annoyance. You will be surprised to find that 10 watts doesn't really sound very much quieter than 50 watts in your own room.

Frequency Response

The next question is, how good does the frequency response have to be? We already have a hint at the answer to this question from the fact that 3 db is the smallest change in loudness that can readily be heard. This statement applies to general program material. On the loudness of a single tone 1 db is just noticeable. This being the case, it is fairly obvious that anything less than 1 db deviation from flat in overall frequency response is going to be impossible to detect audibly. So from the listening standpoint it is pointless to have an amplifier with a response of better than ± 3 db over the audio band.

This is one of those cases where the fact that other manufacturers are giving specifications to fractions of a db, such as .1, .2 or .5, has encouraged competitors to design amplifiers whose specifications do not look unfavorable compared with the best.

Frequency responses are usually given within tolerances of db from flat, and also between frequency limits at which it is assumed the amplifier ceases to be flat. A popular range is from 20 cycles to 20,000 cycles. This is certainly as wide a frequency response as you will ever need. From a practical viewpoint, musical tones of any kind seldom, if ever, get below 40 cycles, so a response down to 40 cycles is all that is necessary to reproduce any kind of program material you are likely to encounter. At the high end, few people can hear above 17 kc., and they have to listen hard to hear that. Quite commonly hearing ceases above 12 or 13 kc., so again it is obvious that

20 kc. is an absolute limit to satisfy even the most critical ears.

Some amplifiers, however, specify a frequency response from less than 20 cycles to over 20 kc. Various reasons are given for doing this, associated with the performance of the amplifier, but it is obvious from the foregoing that this cannot contribute to listening enjoyment.

It has been pointed out that one can detect the difference in sound reproduced through amplifiers whose response goes beyond 20 cps to 20 kc. limits and those whose response rolls off at these limits. This is perfectly possible, but the difference is not necessarily an improvement in quality of reproduction. A more critical examination of the facts shows that the extended frequency range tends to increase the background noise level, which is audible in the form of hiss, so there is a somewhat higher hiss level in the wider range amplifier. Some people seem to have picked up the erroneous impression that the presence of a nice lot of hiss indicates good high-frequency response.

Surely it is obvious that realism in high-frequency response requires the absence of artificial hiss, while maintaining a faithful reproduction of the high frequencies in the program material. This is better achieved by having an amplifier with a response flat between the audible limits and then rolling off gradually at both ends.

Power Response

Another aspect of power output and frequency response relationships is given by some manufacturers under the term "power response." An amplifier may give its rated output, of say 50 watts, over a band of frequencies in the middle range, but may not be capable of giving its full 50 watts over the entire frequency range specified. Its frequency response may comply with the specification at a level of, say, 10 watts, but it will not give 50 watts at the ends of the specified frequency range. Some take the view that such an amplifier does not conform to its specification, or that the specification is misleading.

If the output is given as 50 watts and the frequency range is specified as 20 cycles to 20,000 cycles, then some argue this should mean the amplifier will give 50 watts all the way from 20 cycles to 20,000 cycles. However, few amplifiers listed as having a frequency response of 20 cps to 20 kc. and a power output of 50 watts will give the full 50 watts at 20 cycles or 20 kc.

This is the reason why some manufacturers specify power response, which is a curve giving the maximum output, or the output with a given amount of distortion, plotted against frequency. Fig. 1 illustrates power response plotted against the frequency response of the same amplifier.

To design an amplifier that gives the full rated output at the extreme ends of the specified frequency response range requires the use of a much more expensive output transformer. So

the question will arise, is this extra cost worth it, in terms of improved performance?

This is a subject about which there has been some controversy. But the fact remains that any audio program material possesses nowhere near the full energy level at either the low end or the high end. Consequently the full power is not necessary at the two ends of the frequency response to reproduce any acceptable program material. It may be desirable in a special-purpose laboratory amplifier, the purpose of which is to make measurements over the whole range of frequencies, but this is an application not considered in this article.

An additional aspect would seem to argue in favor of not having the full output available at the end of the frequency response. Full rated output at the high end accounts for burn-out of a number of tweeter units, which can occur if there is any instability giving rise to either high-frequency oscillation or excessive over-emphasis of the high frequencies. If these oscillations or over-emphasis occur, say, at 20 kc. where they are not audible, but the high-frequency unit has to take the power, then the voice coil of the high-frequency unit, which is usually not very large, has to absorb the entire power output of the amplifier, and will burn out if this continues for any period of time.

At the low end of the frequency response, 50 watts at 20 cycles represents a very large movement of a lot of air, because a considerable volume movement has to take place to transmit the necessary energy at the low frequencies. This means one of two alternatives must be chosen. Either the low frequencies must be provided by a number of large low-frequency loud-speaker units, so the 50 watts can be pushed into the room with a reasonable diaphragm excursion, or else the power must be limited, down in this range, so the unit used does not have its diaphragm pushed clean out of the gap, if a stray 50 watts at 20 cycles somehow manages to get through the amplifier.

Distortion

Next we are asked, what do the distortion figures mean? Picking up current catalogues, one finds distortion figures quoted from .05% and even lower, up to 2 or 3%, yet all the units are billed as high-quality amplifiers. So the question naturally arises, what figure can be considered acceptable?

Of the two methods of specifying distortion, harmonic and intermodulation, the former, giving the total harmonic present in a reproduced waveform from a pure sine wave, remains slightly the more popular. Some years ago, the fact was noticed that less than 5% of a second harmonic was difficult to detect audibly, while other forms of distortion produced by the same amplifier (getting only 5% second harmonic distortion) were quite noticeable. This led to a search for alterna-

tive methods of specifying distortion, based principally upon intermodulation checks. The various intermodulation products are more readily noticeable on the reproduction of program material, because they introduce completely spurious tones rather than simple harmonic products which modify the timbre of a tone. But this does not necessarily mean that the method of measuring and specifying distortion is any more indicative of the auditory performance than was the simple harmonic method of measuring distortion.

Last year Mr. C. J. LeBel presented the results of some experiments in a paper before the Audio Engineering Society, in which he also discussed various theoretical relationships between the harmonic method of measuring distortion and the various methods of measuring intermodulation distortion. The specification of intermodulation distortion is further complicated by the fact that there are various standards for making the measurements. An interesting result of Mr. LeBel's experiments was that, while the theoretical relationship seems to hold fairly well with comparatively simple, non-feedback type amplifiers, the modern high-feedback, low-distortion amplifiers did not seem to give such consistent results.

This leads us to ask the question, "Supposing I take two amplifiers, in which identical methods of measuring distortion are used, and identical results are obtained from each measurement; will both amplifiers give me the same apparent distortion *on a listening test*?" The answer is that they may give widely differing results!

To understand how this can be, we need to know a little bit more about the character of distortion. Taking first the older method of measuring distortion, by checking total harmonics, the reason for a difference can be seen fairly readily. Experiments have shown that the second harmonic is the most easily tolerated, and is practically unnoticeable up to about 5% on a single pure sine wave. The third harmonic, however, becomes noticeable at a considerably lower level—somewhere around 1½ to 2%; and higher order harmonics become progressively more noticeable.

Suppose we take an amplifier, well designed but without feedback, which gives 3% harmonic distortion, all of which is of the second variety. Now suppose we put in some extra gain and add 40 db of feedback around this am-

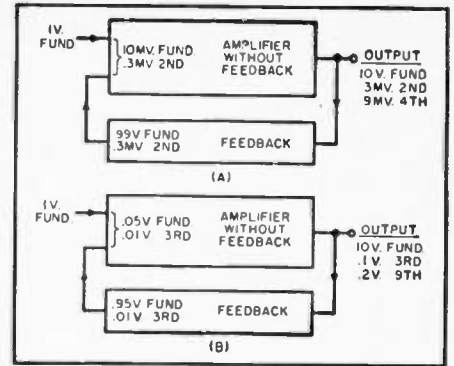


Fig. 2. (A) How an amplifier can increase the order of harmonic while reducing its magnitude, by adding feedback. (B) Another example of raising harmonic order by use of feedback. See text for details.

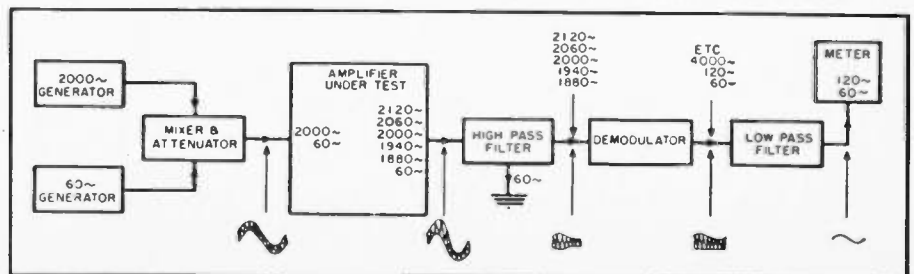
plifier. According to feedback theory we could knock the second harmonic from 3% down to .03%. But we have overlooked something.

The input to our new amplifier, inside the feedback loop, (which has 40 db more gain than the original amplifier) is now 99% feedback balanced against 100% of the original signal. The resultant signal put in—the remaining 1%—will contain 3% second harmonic—or very nearly that much—to offset the 3% which the amplifier is going to generate, resulting in only .03% residual second at the output. Now this 3% of second harmonic, at the input to the amplifier, will generate 3% second harmonic of itself on the way through the amplifier, as well as offsetting the original second, resulting in a component at the output, 3% of 3% fourth harmonic, or .09% fourth harmonic. This is illustrated, with voltage figures, in Fig. 2A.

So our 40 db feedback has knocked our original second harmonic down to .03% and has won us .09% fourth harmonic which we didn't have at all at the beginning. Multiloop amplifiers can do a good job of multiplying the order of residual harmonic that gets left!

Suppose now that the original design is not too good, so that, without any feedback at all, the distortion component might be 20%. It is quite easy to knock this 20% down to 1%, merely by using 26 db of feedback. Assume that the 20% is third harmonic this time. So 26 db of feedback will reduce 20% third down to 1% third. But at the same time 1/5th of the resultant input to the amplifier, inside the feedback loop, will be third

Fig. 3. One way to measure IM distortion. Frequency components are marked to show first and second order components and corresponding first-order wave envelopes.



harmonic; and this will produce 1/5th third harmonic of itself on the way through the amplifier—maybe not quite as high as this because the level is lower than the fundamental. Assume perhaps that the fed back third generates 1/10th of itself, ninth harmonic. This still means that we have 2% of 9th harmonic, although we have reduced the third harmonic to 1%. Fig. 2B illustrates this case.

By dividing the feedback loops up into sections we may keep the resultant harmonic at any one point below 20% and hence avoid producing a high-order harmonic as high as 2%. But the principle is evident. Working in this manner, unless we design the amplifier to have low distortion without feedback, we have high-order harmonics, instead of the low-order harmonics that the old fashioned type of amplifier more commonly had.

Turning now to intermodulation products: most of the *theory* concerning intermodulation products is based on components caused by the same curvatures in the amplifier that produce low order harmonic products. The commonest intermodulation test consists of applying a combination of 60 cycles and 2000 cycles, in amplitude ratio 4 to 1, and then applying a filter system to the output which eliminates the two original signals and measures the residual intermodulation product.

The intermodulation we are looking for is a modulation of the 2000-cycle signal by the high amplitude 60-cycle wave. To measure this, first the 60-cycle component is filtered off by a high-pass filter. Then the residual 2000-cycle component is passed through a demodulator, similar to that used in a receiver, which will detect whether there is any modulation of the 2000-cycle signal. This entails filtering out the residual component of 2000 cycles, and getting left with a modulation consisting of 60 cycles and upward. If only low order components are produced, 60 cycles and 120 cycles may be the only spurious modulation components present, after this process. This procedure is shown in Fig. 3. But if high-order intermodulation products are generated, frequencies as high as 900 cycles may easily be pres-

ent, and a filter network intended to eliminate a carrier of 2000 cycles will also practically eliminate components as high as 900 cycles at this point.

High-order intermodulation products, like high-order harmonic products, are much more noticeable. High-order intermodulation products cause a general "muddiness" in the reproduction. In addition to having a greater "annoyance factor," the high-order products are apt to be inaccurately measured, for the reason just described.

Hum and Noise

This is one more performance figure given in amplifier specifications which can be indicative of how the amplifier will sound. Sometimes hum and noise figures are given separately and sometimes a combined figure is given. Let's take them separately to see what each method of specification can tell us.

First suppose we are given a figure of hum level. Will two amplifiers both specified as giving, say, 90 db below rated output sound the same when fed into the same loudspeaker system? (Assuming of course that the two amplifiers have the same rated output.) Again the answer is, not necessarily.

A look at the loudness contours near the threshold of audibility—which is where we hope to find the hum level—shows that the ear becomes decreasingly sensitive at the rate of about 18 db-per-octave as frequency goes down. The frequencies present in amplifier hum range from 60 cycles upwards. The fact just mentioned means that -90 db hum level, in which the only component is 120 cycles, is no better than a hum level of -72 db, composed entirely of 60 cycles.

A 60-cycle hum is usually due to "break-through" from tube heaters, fed by 60 cycles from the line transformer. A 120-cycle hum is usually caused by residual ripple on the "B+" supply, which is full-wave rectified. Sometimes 180-cycle hum may be present, due to a radiated field from the power line transformer. This will be even more perceptible to the ear. For example, a -90 db hum level, in which the principal component is 180

cycles, will not be better than a 61 db hum level at 60 cycles.

Even this is not the worst possible disparity. Another variety of hum is the ticky, static kind, that can be due either to charging pulses on the storage capacitor of the rectifier filter system or, if choke filtering is used, to current switchover between the two halves of the rectifier. Either way, the resultant is a short duration pulse, of which the hum meter will measure an r.m.s. value, integrated over a 60- or 120-cycle waveform on which it appears. Thus its instantaneous amplitude may be as much as 10 times its r.m.s. reading.

Assuming that the pulse duration is 1/20th of the 60-cycle period, and the amplitude is 10 times its r.m.s. reading, such a ticky hum level can readily fall within a specification of -90 db and yet be noticeable against most program material.

Noise level, which is generally interpreted to mean tube hiss and kindred noises, although sometimes the figure given includes hum, can also have a variety of interpretations according to the precise nature of the noise. In general, a good flat "white" noise is not too noticeable. Even if it is noticeable during quiet periods of the program, it cannot be considered objectionable. But if the amplifier has a sharp roll-off, or a tendency to peak at an ultrasonic frequency, this can give marked coloration to the noise, making it sound like a definite hiss instead of just a background. Differences of this nature can be equivalent to a deviation in measured value of between 10 and 20 db.

This, it is true, is not such a drastic deviation as can occur in specification of hum level, but when the two are given as a combined figure the value does not really convey much. A level, specified as -90 db relative to full rated output, is usually practically inaudible in the average living room, unless you put your ear fairly close to the loudspeaker. If care has been taken to keep the components of noise and hum to the less audible variety, and a figure of -90 is achieved, then you will not be able to hear the background noise, even by putting your ear right into the loudspeaker.

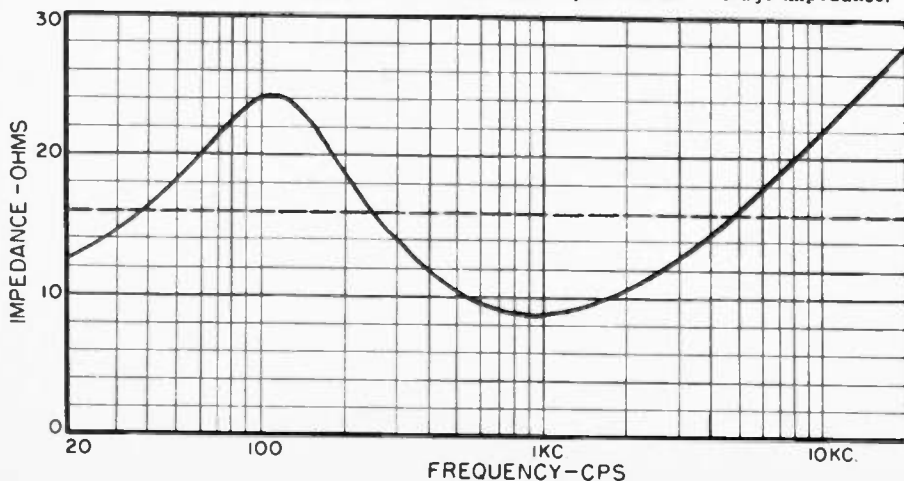
Matching

There are still some more things specified about audio amplifiers that we need to check before making our purchase.

The output impedance must provide for the particular loudspeaker system we have in mind. If the loudspeaker system operates at 16 ohms, then we need an audio amplifier with an output impedance rated at 16 ohms. The writer has one observation to make here however: a little while ago, measuring a number of loudspeaker units, he found that the impedance of the unit at a mid-range frequency was considerably below the rated impedance.

Some loudspeaker manufacturers

Fig. 4. Speaker impedance characteristic. Dotted line represents the average impedance.



have been following the practice of rating the impedance on the average value over the audio range. Fig. 4 illustrates this on a typical impedance curve of a loudspeaker. Now, an amplifier is tested to give its full rated output when it is loaded with a dummy resistance of the value specified for the output tap used. If the amplifier is tested on the 16-ohm output tap, a 16-ohm dummy resistance load will be used for measurements. Many loudspeakers will be found to dip below their nominal impedance value over a range of mid-frequencies and, for this reason, they will absorb more power from the amplifier than is indicated by measuring the voltage on the voice coil terminals. This means that when the amplifier is connected to a loudspeaker it will not appear to give its rated power output.

This method of rating also tends to make the efficiency of the loudspeaker look better because the calculation of power input may show only, say, 9 watts where the loudspeaker may actually be absorbing 15 watts. The writer suspects that this method of rating loudspeaker impedance may have arisen due to an endeavor on the part of the speaker manufacturer to make the sensitivity of his unit appear favorable in comparison with other units, and average impedance is quite a legitimate interpretation.

As the reader will not usually have facilities for checking the impedance of a loudspeaker system at different frequencies, he had best take the loudspeaker manufacturer's word for it and match it to the amplifier according to its rating.

Next comes the specification of input impedance and loading level. By loading level, the manufacturer means the voltage input required to give full output. The input impedance should match whatever the user intends to connect to the input. If the audio amplifier does not include a preamplifier and the user has in mind a separate preamplifier, then the output impedance of the preamplifier and input impedance of the power amplifier should match, both in impedance and level.

Many preamplifier outputs are cathode followers rated at 600 ohms. However, in this case the output impedance rating is not indicative of the load with which the preamplifier should be terminated. It will invariably work better into a high imped-

ance grid, than into a 600-ohm load. This will be discussed more fully in a subsequent article on the choice of a preamplifier.

Power amplifiers have an input, either high impedance to grid or a line impedance of 600 ohms through a transformer. The former is the more practical arrangement for most purposes. If the output from the preamplifier, or whatever source of program material is going to be used, is matched down correctly for 600-ohm loading, then an audio amplifier with 600-ohm input and the right gain for the levels to match can be used.

A high-impedance input to grid has the advantage that it can be connected to a 600-ohm output of the same voltage level rating, without any problems at all, but a high-impedance output from the preamplifier, or other unit which precedes the power amplifier, cannot be fed into a 600-ohm input in the same manner. The 600 ohms will shunt away the high-impedance output, probably causing considerable distortion and certainly reducing the level.

This whole problem is avoided by using one of the combined audio amplifiers, which includes a "front end" in the same unit. This is the present trend. It brings us to the last question we have space to answer here: "Should I buy a separate preamp and power amp, or a combination job?"

This choice depends upon two factors: (a) how much you want to spend; and (b) what kind of system you have in mind.

There is the obvious advantage, from the cost viewpoint, that combining the two units saves a considerable number of components, an extra chassis, and extra power supply. So the combined job is bound to be cheaper than the two separate units.

On the other side of the picture, however, the two separate units offer the advantage, if your system is going to be a fairly large one, that the power amplifier and the preamplifier need not be all together in the same place. The preamplifier can be located on a small table beside your favorite arm chair, while the power amplifier can be located in a cabinet that also houses the rest of your audio equipment. The combined unit may be rather large to locate next to your favorite arm chair, although there are some nice compact units, with outputs in the region of 10

watts, that are very convenient for this purpose.

What about difference in performance between the two types of equipment?

It is difficult to make an over-all statement but the trend is for the combined unit to have a higher hum level.

In separate units, the power transformer for the preamplifier has to handle very small power and it is a simple matter to use a low flux density design of power line transformer, which makes it easier to minimize hum transfer. In a power amplifier, the amount of power to be handled by the power transformer necessitates use of a unit with much higher flux density, and consequent possibility of hum transfer to low level stages such as are used in a preamplifier.

As was pointed out earlier, a 180-cycle component of hum, registering the same on the meter, is almost 30 db more audible than a 60-cycle hum. In the combined units the kind of field radiated by the power transformer is principally 180 cycles, so this is more likely to be the frequency of hum that will appear in a combined unit.

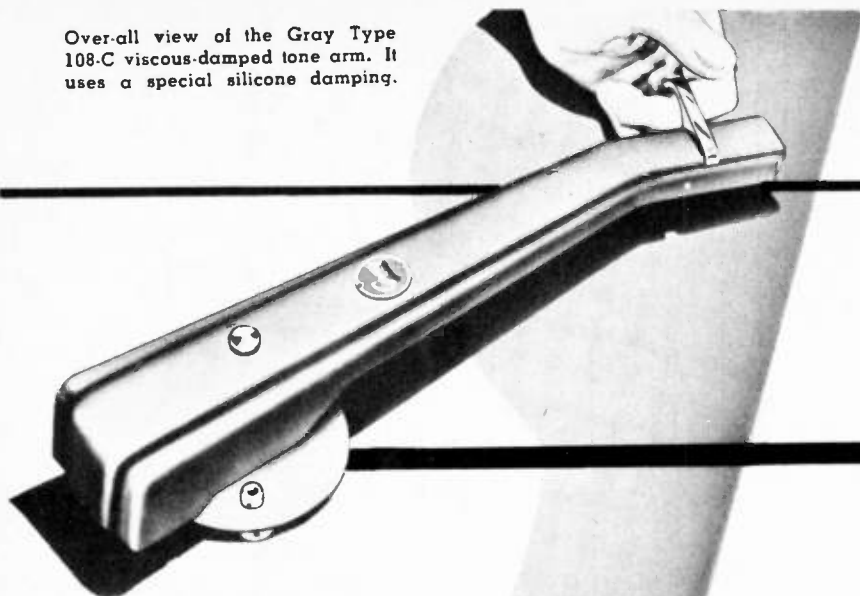
Although careful design can improve these conditions, the problem is much more acute in a combined unit than where two separate units are used. So in making a comparison between systems of the two types, the writer suggests that you should look especially for difference in apparent hum level *when listening to it*. It may be that the specifications will give hum level pretty much the same, but the thing to watch for is whether the *audible* hum level is the same.

We hope that the reader has not formed the impression that we could have said: "listen to it; don't trust specifications" in much fewer words. In discussing what specifications mean, we hope to have given him something to listen *for*; and also to have convinced him that failure of specifications to tell the whole story does *not* mean that the specifications, or those who write them, lie. Intelligent reading of good specifications *can* be a good guide for that "short list."

EDITOR'S NOTE: The factors that enter into the selection of an appropriate preamplifier for use with your power amplifier will be covered by the author in an article next month.

-30-

Over-all view of the Gray Type 108-C viscous-damped tone arm. It uses a special silicone damping.



New Hi-Fi Tone Arm

OF INTEREST to owners of high-fidelity sound systems is the new tone arm recently released by Gray Research & Development Co., Inc. of Manchester, Connecticut.

The Type 108-C viscous damped arm incorporates a silicone damping fluid that is placed at the interface between a ball and socket, as shown in Fig. 1. An adjustable cone-point pivot screw allows the degree of damping to be readily controlled and, at the same time, provides practical freedom from static friction with arm movement. This damping virtually eliminates troublesome low-frequency resonance which frequently causes groove-hopping and distortion on loud passages. Vertical damping also prevents damage to the record and stylus due to accidental dropping of the arm and improves the tracking of warped records.

All of this is made possible by the use of a silicone, which is chemically inert, whose viscosity remains relatively constant over a wide range of temperatures. Unlike most hydrocarbon compounds, it will not oxidize or change its characteristics over a long period of time.

The principle behind this feature is that the viscosity of the liquid controls the arm damping, with additional corrections made possible by various settings of the pivot adjusting screw.

The arm itself is supported by the single needle-point pivot in the center of the ball and socket. See Fig. 1. It is this gap between the ball and socket—a few thousandths of an inch—which is filled with the damping fluid.

The adjustment of the arm's damping is a relatively easy process for the user. He merely holds the arm level and then presses firmly down around the pivot. The pivot screw can then be turned slowly in the clockwise direction until it contacts the cone point of the pivot. A single one-half turn of the screw in the same clockwise direction gives the maximum adjustment for damping.

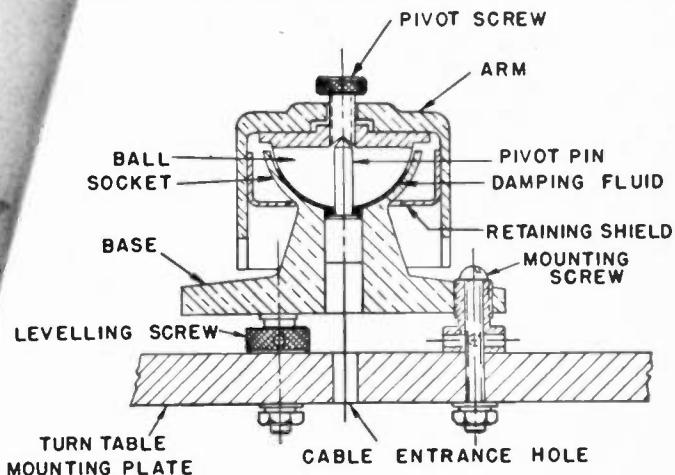


Fig. 1. Cut-away view of damping fluid "well" and its adjustment devices.

The use of a special silicone regulates both vertical and horizontal movement and prevents disc damage from dropping.

The useful range, however, extends about $1\frac{1}{2}$ turns beyond this point. The user merely turns the screw clockwise in one-quarter turn increments, with a delay of about one minute between trials, to achieve the optimum condition whereby it takes about two seconds for the arm (with cartridge in place) to drop one inch to the record.

Each tone arm comes with three cartridge slides of the user's choice. The correct weights for the specific cartridges to be used are included with the arm assembly. The company has developed slides and weights for *Electrosonic* ESL-111; *Fairchild* 215A, 215B, 215C, 220A, 220B, and 220C; *General Electric* RPX-046 and RPX-046 (3 mil); *Pickering* 120, 200, 140, and 240; and *Weathers* cartridges.

The stylus force is adjusted at the factory by lead weight positioning at the back of the arm. Proper force for each cartridge is automatically fixed by the small weight associated with each of the slides. From 6 to 8 grams force is obtained with the combined cartridge-weight-slide assembly. Identical weights are used for the two

Pickering models because the 2.5 and 3 mil cartridges are about 5 grams heavier than the 1 mil model, thus providing proper force.

Three special base levelling screws are provided with the arm and the user, after temporarily inserting the correct cartridge slide, can then make the necessary adjustments to insure that the bottom edge of the arm is parallel to the surface of the record.

All groove widths, all record diameters up to 16", and all normally used stylus forces can be accommodated with this single arm. By utilizing the quick-change slides, cartridge interchange is facilitated. Each slide and cartridge assembly is preset to proper stylus force, reducing to a minimum the danger of unauthorized tampering.

Over-all length of the arm from the stylus end to the center of the pivot screw is $11\frac{5}{16}$ ". By proper placement of the arm, the unit can be used with virtually all sizes of records. The various dimensions for the different types of installations are given in the instructions which accompany each arm.

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FACTS TO KNOW

By N. H. CROWHURST

PROBABLY more thought has gone into the arrangement of the controls and the features to include in a pre-amplifier than for any other item in most manufacturers' lines. A good pre-amplifier has to compensate for differences in program material, differences in individual disc maker's recording characteristic, and differences in individual listening conditions or taste. It should also provide for different kinds of program input both from different types of pickups, so you can make your choice from the available types, and also for FM-AM and other sources. These program inputs will come in at different levels and different impedances.


Thus the preamp manufacturer is faced with the problem of how to cater to all these possibilities in the simplest possible manner. The solution offered by the individual manufacturer has invariably been the result of a considerable amount of thought and discussion. The pity of it, from the buyer's point of view, is that the dealer was not able to be present at all these conferences, and hence be conversant with the factors that made each manufacturer decide on his own particular arrangement. Had this been possible, he would be much better qualified to advise on the best preamp for your particular requirements.

In discussing preamplifiers here, we include not only the separate preamplifiers, but also that part of a composite amplifier where the preamplifier and power unit are combined on a single chassis. In this case the term "front end" might be more applicable. Last month's article covered the requirements of a power amplifier. The facts discussed apply equally well whether the power amplifier is a separate unit or is part of a composite unit. Now we will consider what is required at the input end.

Input Arrangements

If you have any kind of high-fidelity objective, do not take a preamplifier with a single phono input. The time is bound to come, sooner or later, when you will be dissatisfied with the quality of your system at some point and you will want to change various units in an endeavor to improve your system. The one thing you are certain to want to try is the relative merits of different kinds of pickups. To provide for this eventuality, your preamplifier should have at least two separate phono inputs; one identified as "high gain" (low level) and the other "low gain" (high level).

The high-gain input provides maximum gain in the amplifier and is used



When Buying a Preamp

Of importance are specifications, size, cost, and styling. "Listening" tests will not help here.

in conjunction with low-output pickups such as magnetic, ribbon, and moving coil types. Most amplifier manufacturers specify the input for use with both *G-E* and *Pickering* magnetic pickups. Actually all types can be used by making a slight change in the loading resistor in the preamplifier. Most pickups require a variation in the loading impedance and the manufacturer's suggestions should be followed for best frequency response. (Complete details on this will be covered in a forthcoming article.)

The low-gain input is for use with crystal and ceramic type pickups. These units have a much higher output level and therefore require a lower gain amplifier. The *Weathers* type can use this or the radio input.

You will also want to play radio through your high fidelity system, either AM, FM, or both. So you will also need a high-level (low gain) input to receive the output from your radio tuner. All-in-all you will require a minimum of three inputs: two phono and one radio. If you plan to use a tape recorder, either now or some time in the future, it would be advisable to plan on a fourth input stage. The preamplifier, or the front-end of a combined unit, may be regarded as the hub of your high-fidelity system.

Controls

From this we come to the question of how many knobs, with how many positions. Some preamplifiers provide con-

trol entirely by means of rotary knobs while others mix rotary knobs with lever switches, and yet others employ push-buttons for some functions.

For taking care of the record equalization characteristic, two methods are adopted. One employs separate means of adjusting the low-frequency end and the high-frequency end, while the other uses a single control for both functions. The use of two separate controls provides the greatest number of possible combinations. The question is: Will all of these combinations be used? A single knob (known as the equalization control) that takes care of both ends of the response by turning to the appropriate recording characteristic is the simplest for most people to use.

For the real high-fidelity fan who wants to have the maximum range of adjustment for a recording characteristic, separate push-buttons or knobs which can be arranged in any desired combination seem to give a more versatile unit; for example, two five-position switches give five times five, or twenty-five possibilities. The writer feels that this is a rather unnecessary refinement: the preamp usually carries a bass and treble control, separate from record equalization control, to handle variations in program material. The purpose of the equalizer setting is to provide a basic "flat" response for the recording characteristic selected. Equalization other than this can only add unnecessary complexity.

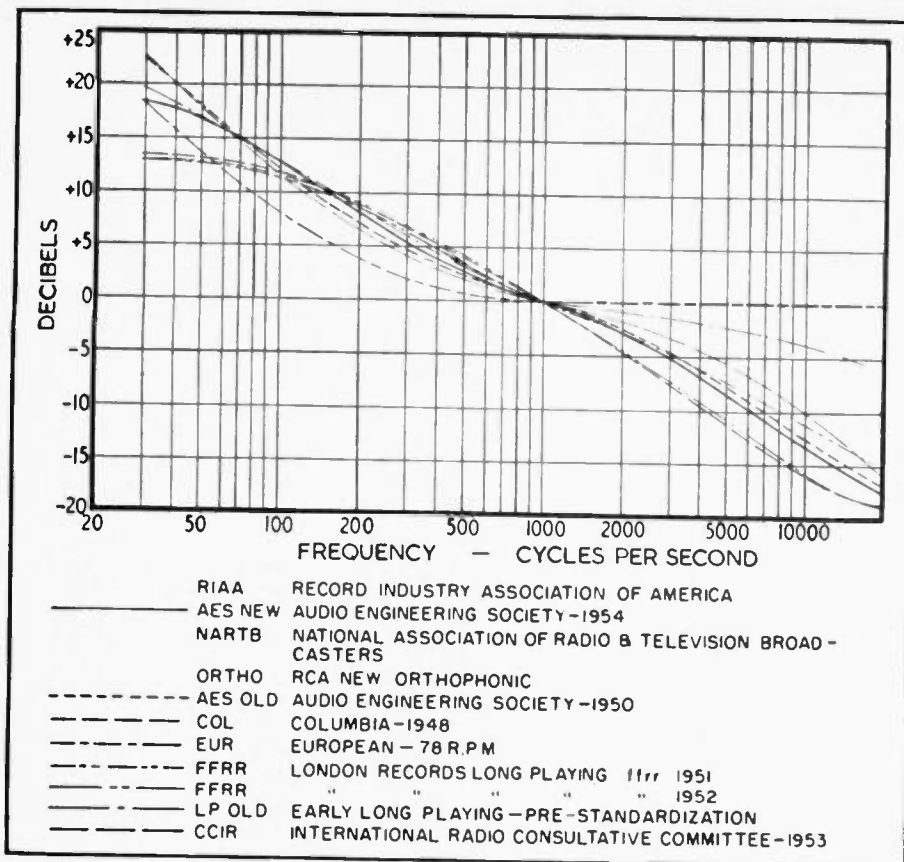
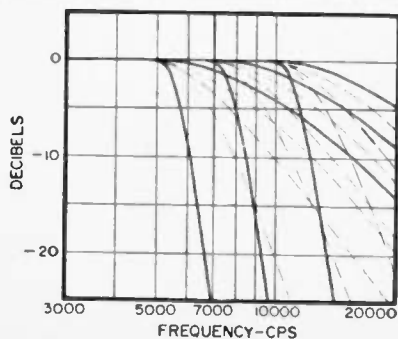


Fig. 1. Playback equalization characteristics for various recording standards. Dates given indicate the time the standard was established, where available.

Some manufacturers have reduced the number of positions on the equalization control since, for example, the AES and NARTB curves are so similar that the user can rarely hear the difference between them. An average curve is given for the two which is considered sufficient. Other related characteristics are covered by a single position. This results in a less expensive unit, because the number of precision components, necessary to produce accurately controlled equalization characteristics, is reduced.

It was pointed out in last month's article that a deviation in frequency response of 1 db or so cannot be detected by the human ear. Consequently the difference between the AES and NARTB equalization characteristic is difficult, if not impossible, to detect.

Fig. 2. Characteristics of one type of variable low-pass filter arrangement. Solid lines represent limiting responses at each frequency position. Dotted lines show kind of variation achieved by the continuously variable control. See discussion in text.



On the other hand, selection of the correct equalization characteristic assures the user that he has flat reproduction to start with before he adds other "quality" adjustments for program material or listening conditions.

To help the reader sort out the confusing range of equalization characteristics, Fig. 1 shows the variety encountered, and identifies the significance of sundry mysterious groups of letters!

On the question of equalizer switching, a point that is worth noting when listening to your equipment is whether or not there are any switching clicks when the equalization is changed. Many of the cheaper units do not provide click suppression, so adjustment of the equalization during a program results in unpleasant clicks. It is much nicer, and easier to tell what change is effected, if the clicks are suppressed so that the only change due to adjusting the switches is in the program quality.

Low-Pass Filter

The next control to consider, especially if you want your equipment to handle old records and make the best of them, or to receive radio programs of varying quality, is a variable low-pass filter arrangement. This provides good high-frequency response and noise rejection at the same time. The best equipment provides a selection of roll-off frequencies, with a continuously variable roll-off slope, or else provides a selection of roll-off slopes with a continuously variable roll-off frequen-

cy. Either arrangement will enable the position and roll-off to be adjusted to get the best possible performance from any given program material and background noise. Fig. 2 shows one way of achieving this.

However the addition of such a feature makes the preamp cost more; if you don't have occasion to play inferior quality program material through your system, you may consider such a refinement unnecessary, or perhaps be content with a variety that doesn't require two controls to adjust for best conditions.

Bass and Treble

Next we come to the bass and treble controls, which are designed to adjust the balance between low and high frequencies in order to improve original program material or compensate for the peculiarities of your listening room. Unlike the variable high-pass filter, these controls will not noticeably affect the relation between program material and background noise. They will affect the output at the bass and treble frequencies with relation to the middle.

There are two ways of making the response adjustable, as shown in Figs. 3 and 4. Which is better? That's a tough question to answer. To some extent it depends on the program material, so it might be advantageous to have a double control at each end—making four bass and treble control knobs in all!—so that we could vary the response both ways at once.

But let's not recommend any more knobs. The better choice is probably the type shown in Fig. 3, as the variation is positive and meets most requirements for this type of control.

Loudness

The next feature we'll discuss has been the cause of much argument. It is the loudness control. The argument involves just what we expect a loudness control to do.

A program reproduced at a level different from its original level does not sound real, because of the difference in the frequency response of the human ear at different levels. The term "scale distortion" was invented to cover the apparent change in quality which occurs when the reproduction level is changed from its original loudness.

From this the theory was developed that we need to compensate for the difference in the frequency response of the human ear at different levels, so that the music would appear to have the same frequency content or response at whatever level it is played. This may be what we require but this is open to question.

It is certain that it will not give an impression of realism. For example, a brass band at a certain distance has a certain loudness and program content. If the band moves farther away, or we go farther away from the band, the loudness and its apparent quality change. If we try to reproduce the

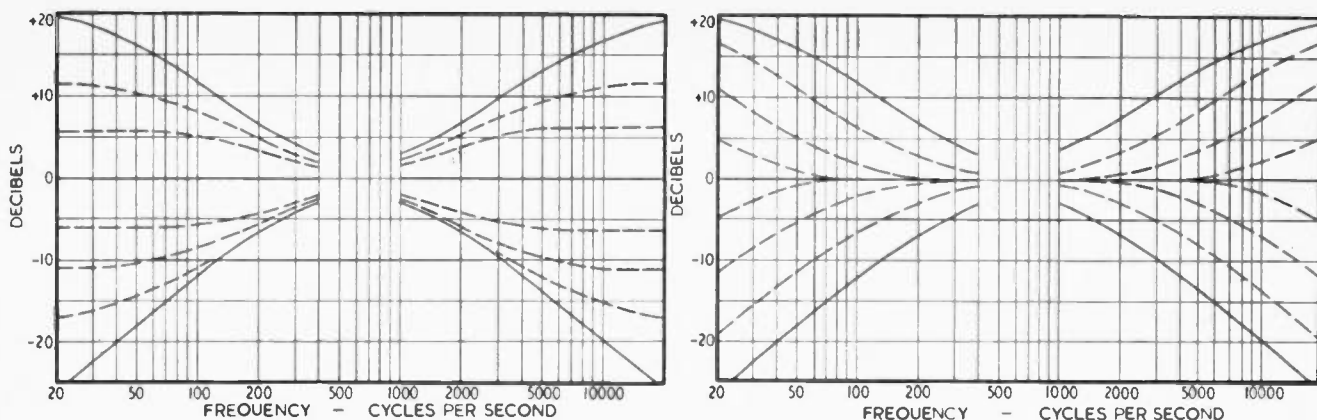


Fig. 3. One variety of bass and treble characteristic provided by some preamplifiers. The solid curves show limiting positions, while the dotted curves show typical response at intermediate settings.

Fig. 4. An alternative form of response variation provided by some bass and treble controls. This can have advantages over the form shown in Fig. 3, but for most program material the former serves best. A really deluxe control might provide a combination of both, requiring four knobs in all for bass and treble control.

band's music at a lower level than its original recorded level, but preserve the same apparent frequency components, it may give us something that is pleasing reproduction, but it will certainly not sound like a band either close to or far away. It will be as if the instruments are playing quite softly close to. And if you can imagine a trumpet playing full blast softly, then you may also consider that this kind of reproduction sounds real. But is realism in this sense our objective?

Turning to another aspect of the problem: we naturally like to hear all of the program material. If there are some bass instruments present, or a triangle contributing some very high-frequency components, we want to hear all of it. In point of fact, when listening to an orchestra from a distance or at low level, the low frequencies will actually be inaudible. But when we listen to reproduced program material, we know the low frequencies ought to be there and so we listen for them.

This means that we like to be able to increase the level at these frequencies so that we can hear all the instruments, even though they might not be audible in the original performance. The loudness control enables us to provide quick compensation for this so that as we change the reproduction level we don't lose the high and low frequencies, which happens if we get a long way from original sound.

Having decided this much—that we would like to have something in the nature of a loudness control—the question is: just exactly what do we want it to do?

An important feature to remember here is that *at no listening level is the response of the human ear flat*. Consequently, if we put in a correction for every different loudness contour of the human ear, we are putting in some compensation between the original program sound and our reproduced sound at every listening level. This

means that even when the sound is reproduced at the *same* level as the original, it will not be reproduced proportionately because there has been some correction.

Obviously then, our loudness control should correct for the *difference* in loudness between the original recorded level and the level at which we wish to listen. The average recording level is somewhere around 70 phons.* Most probably you will want to listen at somewhere around 50 phons, if you intend to really listen to the program, or if you merely want pleasant background music, while you are giving attention to something else, such as conversation (I apologize to audiophiles for making this outrageous suggestion, but *some* people want it) a level of about 20 or 30 phons may be adequate.

So the loudness control should provide a differential compensation between the original 70 phon level of recording, and a 30 to 50 phon level for reproduction. If you want your reproduction as loud as the original, which is sometimes an objective so you can compare your reproducer system with original live program material, then you will also need a loud flat position, so that you can reproduce the material in your living room at 70 phons, and see whether you get the full experience of a live orchestra.

So if you want a loudness control, it should be separate from the volume control. This, of course, will raise the cost of the preamplifier. If cost is an important factor the loudness control could be omitted. Adjustment for deficiencies of the human ear can then be made by adjustment of the bass and treble tone controls.

Some preamplifiers provide a loudness control in addition to the volume control, so that a switch can select one of the loudness contours, while the volume control provides a fine loudness adjustment at any individual level. Other units provide an alternative volume or loudness control.

The difficulty of the latter method is that use of a loudness control to set the reproduced volume means that the

differential action cannot be controlled apart from the volume. In this way it is not possible to compensate separately for differences in recorded level on the disc and for differences between the original loudness in the studio and in the reproduced program in the listening room.

Using separate volume and loudness controls, the volume control can be used to adjust the gain of the preamplifier to compensate for differences in modulation on individual discs, due to the fact that not all recordings are made at the same level in the wax, although they may have been recorded at a 70 phon level in the studio. The loudness control should be set in a position to correspond to the listening level actually desired. In this way good compensation can be achieved under all circumstances.

Background Noise

To get the best discrimination against noise from the preamplifier a matching arrangement should be employed between the preamplifier and power amplifier so the full power output from the power amplifier is obtained when the preamplifier gives its rated output.

Most modern preamplifiers and power amplifiers provide considerable overlap in this regard. For individual cases the rated output of the preamp, in conjunction with the rated input for the power amplifier, should be consulted, and an appropriate network made up so as to get the best operating conditions under all circumstances.

Not all preamplifiers incorporate all the features mentioned in the foregoing discussion. A preamplifier with all of these provisions would have a lot of knobs to twiddle and the average person would find it somewhat confusing to operate. It would be difficult to know which knob to turn to achieve a given effect. But the modern range of preamplifiers provides a good selection of these controls to give reasonable versatility with simplicity of operation. Each manufacturer has chosen his own way of solving the problem.

* A "phon" is a unit of loudness based on the average human ear where the ticking of an average-priced watch three feet away will register 30 phons.



HOW TO IMPROVE YOUR HI-FI AMPLIFIER

By MILTON S. KIVER

Don't try to re-design your amplifier—but there are several simple improvements that can be made on any unit.

INTEREST in high-fidelity is at the highest pitch it has ever been and from all indications it is headed even higher. Alert service technicians welcome this growth for it means new sources of revenue. The expansion is also a boon to the hi-fi enthusiast because the more interest the public shows in a product, the more time and effort manufacturers will spend trying to improve it. As a result, everyone benefits.

The heart of a high-fidelity system lies in the amplifier. Here is the place where the audio signal receives the major portion of its amplification and, frequently, the major portion of its distortion.

Commercial amplifiers on the market and build-it-yourself kits come in a variety of power capabilities and in a range of qualities. It may very well be that the amplifier you finally purchase (either factory-wired or in kit form) does not have as low a noise level, or as clean a sound, or as wide a response as you would like it to have. But to get what you really want would entail more money than you care to spend. Must you, then, be satisfied with less or is there something you can do to improve the quality of your system?

While changes in basic circuit design are seldom feasible, there are a number of "fringe" modifications that can be made to measurably improve the characteristics of an amplifier. In most instances, the additional component costs are nominal and anyone with a basic understanding of amplifier operation may make the changes readily.

The important attributes of an audio amplifier include hum, noise, frequency response, output, and distortion (at rated output). For a well-designed amplifier, hum and noise would be on the order of 70 db or more below rated output and frequency response would be flat from about 30 to 50 cycles at the low end to well above 20,000 cycles at the high end. Distortion, to have any significance, must be measured at rated output and anything less than 2 per-cent intermodulation distortion can be considered excellent.

Here, then, is a set of figures to work against in evaluating the performance of your particular system. If you feel that your amplifier could stand improvement, certain simple steps can be taken to step up its performance.

Hum

The presence of hum in the loudspeaker is an indication that a.c. is somehow reaching the signal path of the amplifier. As a first step in dealing with this problem, a complete

measurement should be made to determine just how much voltage there is. With this information on hand, you will then be in a position to know exactly how much improvement each corrective change provides.

The hum measurement is made as follows:

Disconnect the speaker voice coil from the amplifier and substitute an equivalent resistor of the proper wattage rating. In the usual case, this might be an 8- or 16-ohm resistor rated at 10, 20, or 30 watts, depending on the amplifier output. Then, with nothing connected to the input of the amplifier and the volume control wide open, measure the hum voltage appearing across the load resistor. This can be readily done with an oscilloscope and might appear as shown in Fig. 1. A v.t.v.m. can also be used for the same purpose provided it is sensitive enough. The oscilloscope, however, is the more desirable instrument.

If the hum power is desired, take the square of the hum voltage (r.m.s. value) and divide it by the resistance of the load. The result is hum power. Then, by the use of the formula:

$$db = 10 \log (P_1/P_2)$$

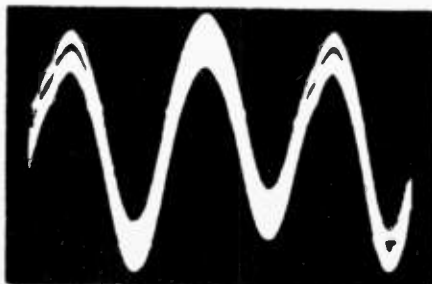
where: P_1 = rated power output of amplifier

P_2 = hum power output

you can obtain the number of db the hum level is below rated output. Any value in excess of 65 to 70 db can be considered good.

The sources of hum in an amplifier are many and diverse, but in the present discussion only those which can be corrected fairly readily will be considered. For example, parts placement has a very definite bearing on the amount of hum heard in the speaker. However, to remedy this situation would mean we would have to rearrange the chassis parts layout and this is seldom feasible in existing

Fig. 1. Appearance of hum on an oscilloscope connected across amplifier output.



equipment. Hence, this point will not be considered. To a certain extent, the same is true of shielding. Large components, such as transformers and chokes, are not readily shielded by adding auxiliary shields. On the other hand, tubes can be shielded quite easily. The problem, then, is to see what we can do to reduce hum simply, economically, and effectively.

One of the important sources of hum in an audio amplifier stems from the use of a.c. to heat the cathodes of the low-level stages. This alternating voltage can reach the other tube elements by electrostatic and electromagnetic coupling (between the cathode and the grid, screen grid and plate) and also by means of leakage paths from heater to cathode. To avoid these difficulties, many higher priced amplifiers (and preamplifiers) use d.c. to heat the cathodes. The required current can be obtained from a separate power supply (Fig. 2A) or by utilizing the cathode currents flowing through the power output amplifier (Fig. 2B).

In established circuits, a simple approach to hum reduction and one which is frequently quite effective is demonstrated in Fig. 3. A small 100-ohm, 2-watt potentiometer is shunted across the heater windings of the power transformer. The center arm of the control is grounded. (If the transformer winding itself has a center tap, this should be left unconnected.) Then, with the system in operation, the potentiometer arm is rotated until the hum level is at its lowest point.

An elaboration of this circuit is shown in Fig. 4. The potentiometer is retained, and a small positive voltage has been introduced into the heater circuit. The purpose of the "B+" voltage is to make each tube heater sufficient positive so that no leakage current can flow from heater to cathode, thereby eliminating this method of hum introduction. The amount of positive voltage required is small, on the order of 30 to 40 volts, and it may be obtained by connecting appropriate resistors across the power supply output.

Another useful method of minimizing the amount of hum reaching the grid of an amplifier is through the use of a single grounding point for the grid and cathode circuits of any single stage. The reason for this stipulation is that if the grid and cathode circuits are grounded at separate points, any a.c. flowing through the chassis will develop a minute voltage across the chassis resistance between the two grounding points. This voltage is thus placed effectively in series with the signal voltage, causing hum to appear in the tube output.

As a matter of general practice, it is best to use a single bus bar (or No. 12 or 14 wire) as the grounding line for the entire amplifier. The bar is everywhere insulated from the chassis except at one point. This is usually chosen at the input to the system. By this means, the introduction of hum from currents circulating through the chassis is avoided.

Finally, all heater wires traveling from tube to tube should be carefully twisted about each other to effect as much of a cancellation of magnetic fields as it is possible to achieve. An interesting practice which may assist in hum reduction is to run the heater wires above the chassis—rather than along the underside. Holes are drilled at the heater terminals of each socket, enabling the wires to make suitable contact with the terminal lugs. In this way, the chassis acts as an effective shield between the 60-cycle a.c. fields and the various circuit components.

The foregoing hum sources have all been directly associated with the power line and hence the frequency is 60 cycles. There is still another source of hum, that due to insufficient filtering of the "B+" voltage. The ripple frequency here is 120 cycles due to the full-wave rectifying action common to all amplifier power rectifiers. As far as audibility is concerned, both sources of hum are equally annoying, however, the frequency is stressed so that the technician may use this difference to determine whether the hum is being brought in from the power supply or the heaters.

The obvious solution to hum in the "B+" line is additional filtering. As a start, shunt additional filter capacitors across the "B+" line and note whether the hum level decreases. If it does, and the existing capacitors are judged to be good, then the extra capacitors may simply be added to the system, or the original units may be removed and higher-valued capacitors inserted in their place.

Each of the foregoing suggestions will afford a certain amount of relief from hum and all together will reduce hum substantially. In instances where the technician has difficulty locating the source of the hum, a useful approach is to start at the grids of the output stages and work back to the input, one stage at a time. The grids of all stages prior to the one being worked should be grounded so that any hum they may contribute is prevented from reaching the part of the system being worked on. Try to reduce the hum of each stage to as low a level as possible before moving on to the next stage. In this way, systematic correction can be worked out.

Noise

Another important characteristic of amplifier operation is the amount of noise that the system develops. As with hum, it is the low-level stages that are most critical in this respect and it is specifically toward them that corrective measures are directed.

Noise in an amplifier is due to two main sources—thermal agitation and tube hiss.

Thermal agitation arises from the random motion of electrons in any conductor. The movement of the electrons, in both directions, constitutes a current flow. Since there are usually a few more electrons moving in one direction than in the other, a voltage is set up across the conductor which is

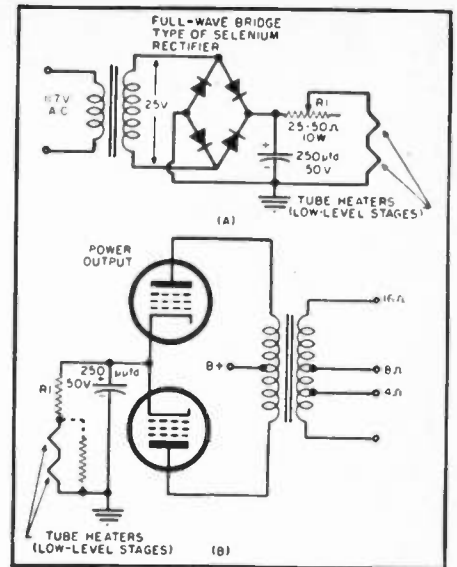


Fig. 2. (A) A suitable d.c. heater supply. The setting of R_1 depends on voltage output of rectifier and number of heaters in series. Also, shunting resistors for tube heaters may be needed if tubes require different currents. (B) Another method for obtaining d.c. power for the heaters of low-level stages. R_1 is chosen so that its voltage drop plus those of the heaters equals required stage bias. If too much cathode current is available, a suitable shunting resistor will be required.

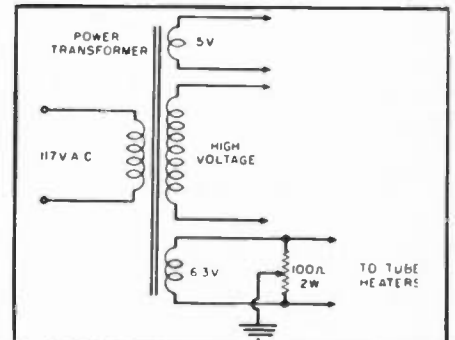
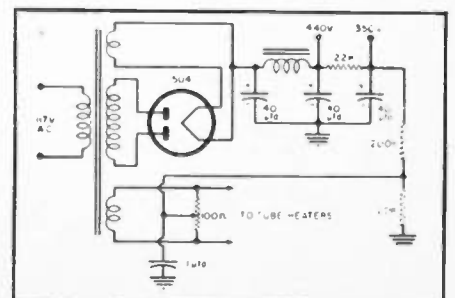


Fig. 3. A simple hum-reducing control. This can readily be added to existing amplifiers.

proportional to the net current flow and the value of the conductor resistance. The polarity of the voltage due to thermal agitation changes constantly, depending on the direction in which the maximum number of electrons is moving. Because of this, there is no definite pattern to the random voltage, or, for that matter, any one frequency at which the electrons move. It has been found that the energy of

Fig. 4. How a small positive d.c. voltage can be introduced into the heater circuit.



the disturbances is distributed uniformly throughout the entire frequency spectrum used for communications.

The amount of voltage that is developed by thermal agitation in conductors is computed from the following relationship:

$$E^2 \text{ (r.m.s.)} = 4KTR \times (f_2 - f_1)$$

where: E = the r.m.s. value of the voltage generated across the resistance.

K = constant = 1.37×10^{-23} watt-second/degree.

T = the temperature of the conductor (it is expressed in absolute degrees, Kelvin, which is equal to 273 plus the temperature in degrees, centigrade).

R = the value of the resistance, in ohms.

$f_2 - f_1$ = the bandwidth of the system, here actually from 30 to 15,000 cycles since this is the normal limit of audibility for most people.

Since the amount of noise generated is quite small, only that appearing in the first one or two stages of a system is important. Work which has been done on noise evaluation in resistors' indicates that the amount of noise generated decreased with the wattage rating of the resistor. Thus, 1-watt resistors of a certain value produced less noise than $\frac{1}{2}$ -watt resistors and the latter were better than $\frac{1}{4}$ -watt units. Furthermore, deposited film resistors and non-inductive wirewound resistors produced considerably less noise than composition resistors of the same rating. If these results are put to use in the first few stages of an amplifier, the noise reduction can be substantial.

The other main source of internal noise is in the tubes. We obtain a series of overlapping impulses due to the fact that the current flow from the cathode to the plate of a tube is not a continuous fluid but a moving congregation of separate particles, the elec-

trons. This is known as the "shot effect." We obtain noise even when so-called steady current is flowing, because, at any single moment, more electrons are impinging on the plate than at some other moment. Over any time interval, the current is steady, but instantaneous fluctuations represent the noise component. Examination has revealed that the energy of the noise is distributed evenly throughout the frequency spectrum. In this respect it resembles the noise arising from thermal agitation.

With respect to tube noise, very little can be done. There are special low-noise tubes which have been designed for application in the low-level stages of audio amplifiers, but unless your system contains such a tube, it is ordinarily not suggested that a change be made, since this would require a redesign of the system. However, in the choice of amplifiers, it is well to keep this fact in mind. The best known low-noise tube is the 12AY7; this is a dual triode. Another is the RCA 5879, and the British Z729. The latter two are pentodes.

Circuit Balance

Two of the major advantages of push-pull output amplifiers are their practical elimination of second harmonic distortion and the greater permissible drive at the input providing a stronger output with less odd-harmonic distortion. These advantages, however, are possible only when both halves of the system are properly balanced and both are receiving the same amount of input signal. The greater the deviation from this balance, the less benefit we receive from the use of push-pull operation.

In checking for balance, there are a number of factors to keep in mind. First of all, the push-pull output stages themselves must be balanced and this is usually determined by measuring the cathode current through each tube.

Another item to check is the applied signal to each tube and whether or not these are equal. A good test frequen-

cy is in the range of 1000 to 2000 cycles. Any indication of unequal input signals means that the preceding phase inverter is unbalanced. It may be that both halves of the phase inverter (if two sections are used, as in the Williamson amplifier) are not functioning with equal vigor, or resistor values may have changed. Whatever the cause, it should be corrected.

At this point, many technicians might feel that the system was balanced. For the middle range of frequencies this might be true; but what about the low- and high-frequency ends of the band? At the low end, supposedly equal-valued coupling capacitors may differ considerably from each other and this would cause each push-pull tube to receive unequal amounts of signal. At the high end, the balance may be upset by stray shunting capacities. With signals of the proper frequencies, check response at both ends and only when equal signals are obtained at all frequencies can you be assured of receiving *all* the advantages of this form of operation.

Stray capacitance is a function of parts layout and a better balance will be obtained if care is taken to make the system leading into the push-pull stages as physically symmetrical as possible. A little attention to details will pay off in cleaner sounding high notes. At the low-frequency end, check supposedly equal coupling capacitors and matched grid resistors to see whether they come within 5 per-cent of each other. If they do not, replace them with units that do.

Low-frequency response can also be improved by using larger filter capacitors in the decoupling networks between stages. Ten microfarads is a common value; raising these to 30 or 40 μ fd. will serve to improve the low-frequency response.

The foregoing suggestions are put forth with the idea of helping technicians improve the quality of existing amplifiers. Basic design has, at no point, been altered; all that has been recommended is refinements. -30-

Choosing

A PHONO PICKUP

By

N. H. CROWHURST

A PERFECT pickup has not yet been designed and is not likely to be. So the best we can do is to find the nearest thing to perfection among the different varieties of cartridge available. Because none of them is perfect, the decision as to which one comes closest to perfection is a matter of personal choice. As in many other fields of endeavor, one school of thought claims the correct way of doing it and thinks any other way is incorrect. The fact sometimes remains that the man who adopts a method dubbed as "incorrect" often achieves results quite as good as, and sometimes better, than the man who adopts the supposedly "correct" method.

The whole field of audio has its own special jargon, but probably no part of the field has a greater concentration of specialized terms than the beginner encounters when he sets out to acquire a high-fidelity phonograph reproducing system. The only way to decide for yourself is to be well armed with a knowledge of the significance of the different technical terms involved, so as to know what to look for and listen for in each particular type of instrument.

Transducer Element

Let's start by considering the different methods by which the mechanical movement, picked up by the stylus, is transformed into electrical energy to be amplified by the system. This is chosen first because it is a very important point in achieving freedom from distortion. Also there are quite a number of names here, some of which have overlapping significance, so the newcomer can be confused as to just which type is which.

The method that has been in use the longest is called, variously, moving iron, magnetic, or variable reluctance. The term "moving iron" derives from the days of radio when there were just two kinds of loudspeaker. The older variety used a fixed coil with an iron armature that moved when signal currents were fed through the coil. The improved type of that day had the whole magnetic structure fixed and the coil was allowed to move with the

audio signal currents in it. These types were designated respectively moving iron and moving coil. The same principle is applied to pickups where, in reverse, the movement causes the electrical currents instead of *vice versa*.

The term "moving iron" signifies that there is a moving armature with a fixed coil and that movement of the armature in some way varies the magnetic field passing through the fixed coil.

This construction is also called magnetic because the movement of the needle or stylus causes changes in the magnetic field which, in turn, generates an electrical output.

It is also called variable reluctance because reluctance is the quantity which causes the magnetic field to change. "Reluctance" is a term in magnetism similar to resistance in electricity. By changing the length of air gap in different alternate magnetic paths from the magnetic pole pieces, the reluctance of those paths changes and hence the magnetic field or flux changes its course. Fig. 1 illustrates the basic structure of a pickup of this type.

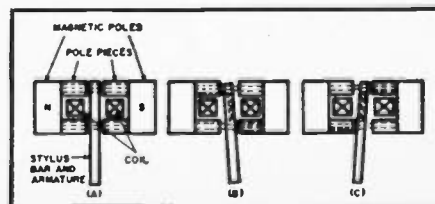
The next group comes under the general heading of dynamic. This is really an abbreviation for the full term electrodynamic which means that the audio output generated derives from movement of the electrical "conductor"

in some way. In this group are two sub-groups, moving coil and ribbon, each of which employ the same basic principle. The chief difference lies in the fact that a coil contains a number of turns wound into a solid form, while a ribbon is a single half turn of flexible conductor that is free to move in the magnetic field.

In a pickup the moving coil or ribbon has to be extremely small in order to maintain good frequency response out to the high frequencies.

There are two basic ways in which the moving coil can move. These are illustrated in Fig. 2. In Fig. 2A the coil is rotated in the magnetic field in such

Fig. 1. Basic principle of moving iron, magnetic, or variable reluctance type pickup. At (A) the armature in its central position does not carry any magnetic field in the coil. At (B) some of the magnetic field, shown dotted, passes through the coil, because the armature is displaced. (C) Displacement in the opposite direction reverses the direction of field through coil.



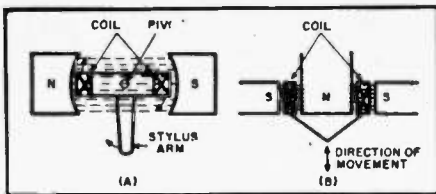


Fig. 2. Basic principles of moving coil pickups. At (A) the coil rocks about an axis due to the movement of the stylus. At (B) the whole coil moves as indicated.

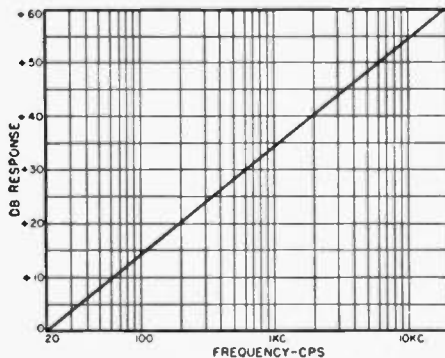


Fig. 3. Relation between constant-velocity and constant-amplitude characteristics. If a variable-frequency disc were recorded at constant amplitude and played back with a velocity-sensitive pickup, the response would follow the curve indicated above.

a way that the voltages developed in it, due to its rotation, are additive in the opposite sides of the coil. In the arrangement of Fig. 2B the coil follows a push-pull movement and in this case the whole of the coil is active in producing voltage because the whole of the coil is between a north and south pole. The latter construction is basically similar to that of a moving coil speaker except, of course, that it is very much smaller.

All the types so far mentioned use electromagnetic principles for transforming the mechanical movement into an electrical output. The difference lies in that the first type produces the electrical output by changing the configuration of the magnetic field, while the second (dynamic) type keeps the magnetic field constant—and it should be uniform throughout the area in which the coil moves—and produces the electrical output by moving the coil (or electrical conductor in the case of the ribbon) in the uniform magnetic field.

This is where our theorists would have us believe there is a right and wrong type. According to theory it is impossible to obtain a perfectly uniform change in the configuration of the magnetic field with the movement of the armature. Early studies of magnetism show that the attraction between poles of opposite kind is proportional (a) to the strengths of the poles and (b) inversely to the square of the distance between them.

If we consider the construction shown in Fig. 1, statement (b) means that in a condition of balance, where the distances are equal, as shown, all the magnetic pulls are uniform; but as soon as the armature gets a little off-balance it will tend to pull to one side

or the other according to the direction of movement. This is overcome in practice by use of a centering force, which also often serves as damping, enabling the armature to be restored to its central position. Careful attention to the design of the magnetic field construction can also produce a magnetic centering force.

But whatever construction is used, it is theoretically impossible to obtain an electrical output that faithfully follows the mechanical input, because of the inherently nonlinear manner in which the magnetic field changes with movement of the armature. In practice, the fact that the phonograph needle or stylus moves such very small distances, in the course of playing a phonograph record, means that the effective linearity has to be maintained over a very small movement, and not over the entire distance through which the stylus *could* move before hitting both pole pieces. This fact means that, by very careful design, a close approximation to linearity can be achieved over the small amount of actual movement encountered in playing phonograph discs.

The protagonists of the dynamic, that is, the moving coil or ribbon type, phono pickup tell us that it is inherently linear, because, if the coil moves in a uniform magnetic field the output will always be proportional to the mechanical movement causing it no matter how far the stylus moves.

In practice there are many other things that enter into the design of a phono cartridge, which means that it is not an essential fact that a moving coil or ribbon cartridge is free from distortion, while a moving iron cartridge must have distortion. The electromechanical transducer effect is not the only thing that can cause distortion. The mechanical suspension arrangement, responsible for controlling the movement, can also cause distortion in the way the stylus moves, or follows the groove. This fact can mean that a moving coil may not give any better performance than a well-designed pickup of the magnetic or moving iron variety.

All of the pickups thus far discussed require some kind of damping or controlling element to center the movement of the mechanism. For satisfactory performance of the unit, this material has to possess both a stiffness, or compliance, and a resistance to movement, or damping. Various kinds of rubber and plastic materials are employed for this purpose.

At different times in the history of phonograph development, individual pickups have been built, which at the time possessed perfect damping. The big problem is that none of these damping materials so far produced will last indefinitely. Their qualities deteriorate with time, and hence the performance of the pickup will not maintain its "new" standard. Improved materials have been developed which considerably lengthen the time for which the pickup will give its "new"

performance, but to date no material has been developed which will last indefinitely.

This means that any pickup in either of these classes has a limited life before the damping material must be renewed, if original performance is to be maintained. In practice, of course, the adoption of the plug-in cartridge idea has enabled the user to remove the old cartridge and plug in a new one of identical type, when performance begins to deteriorate.

Another feature that all the pickups discussed so far possess in common, is the use of the *velocity* principle. This means that the electrical output from the cartridge is dependent upon the *velocity* at which the stylus tip moves. Technically this means that the peak of an audio waveform occurs when the stylus tip is moving at its maximum velocity which will be the middle position of a fluctuation in the groove. When the stylus is not moving the output is zero momentarily.

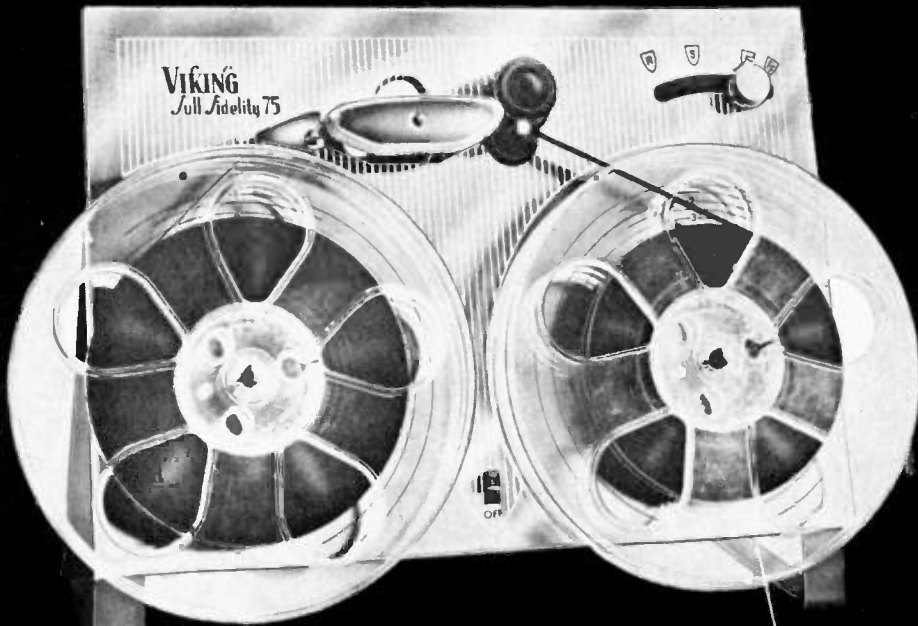
This would appear to be just a matter of phase relation and, as such, would be comparatively unimportant, but there is another fact that derives from it. If we consider a theoretical disc on which a groove of varying frequency has been cut, and the amplitude of movement of the groove is the same at all frequencies, then obviously the stylus will have to move very much faster at the high frequencies to reach this amplitude than it does at the low frequencies. In fact, the velocity of movement, for constant amplitude, will be proportional to frequency. Since the output from this type of cartridge is proportional to velocity, this means that a constant-amplitude recording will produce an output rising at 6 db-per-octave, in order to be proportional to frequency, as shown in Fig. 3.

In practice, most of the cutter heads used for making recordings also employ the constant-velocity principle so that the recordings would be of constant velocity with frequency rather than constant amplitude, if it were not for the equalization employed, both in recording and playing back. If a completely constant-velocity principle were used in making recordings, the width of groove modulation would increase at low frequencies, inversely proportional to frequency.

Because this would result in prohibitive groove excursion, wasting record space at the low frequencies, the response is rolled off as part of the equalization characteristic on recording, and a corresponding low-frequency boost is introduced in the playback amplifier, following the pickup.

Similarly, because the very small excursion that results at high frequencies will not be much bigger than the little bits of dust that collect in the groove and cause noise, the high frequencies are pre-emphasized, so as to be of larger amplitude, before recording, and the playback equalizer introduces a high-frequency roll-off to bring these upper frequencies back to their

(Continued on page 133)



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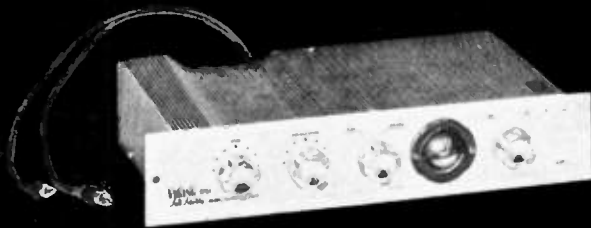
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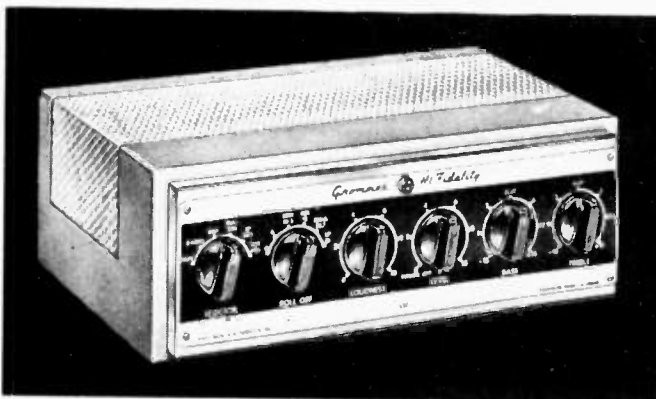
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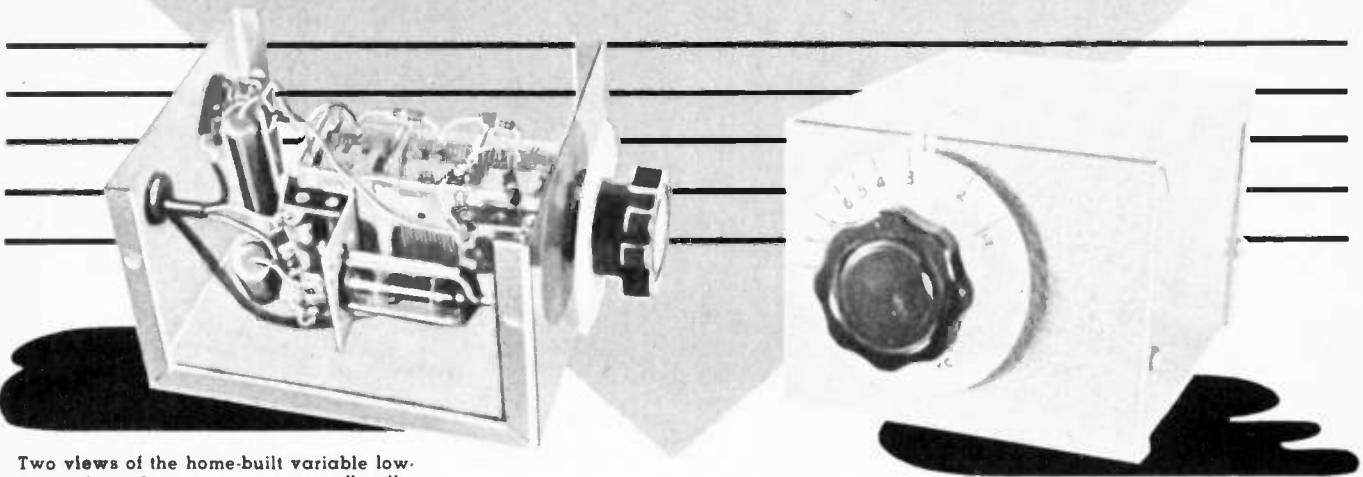
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CHAPTER 6

MISCELLANEOUS

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A VARIABLE LOW-PASS FILTER



Two views of the home-built variable low-pass filter. It is housed in a 3" x 4" x 5" aluminum utility box which is small enough to be tucked in almost anywhere.

By LAWRENCE FLEMING
Consultant

Build this versatile unit for your hi-fi system or communications receiver at a total cost of \$7.50.

TONE CONTROLS for hi-fi have become quite standardized in recent years. About 15 db boost or cut available at either the bass or the high end, with the crossover at 600 cycles, has shown itself over long experience to be the best simple tone modifier for building into an audio music system.

But there are plenty of other things that one can do to a response curve. The most useful among these is a rather sharp high-frequency cut-off, obtained with a low-pass filter. With perfect program material, such filtering is not needed, but with much of the program material from broadcast tuners and discs the low-pass will discriminate against noise and hash much better than a regular treble control. Such 9 kc. low-pass filters have been used very extensively, for example, in movie theater sound systems, as giving the best compromise between best fidelity and background noise on the sound track.

The trouble with a low-pass filter of standard design is that it has to be built for a specific cut-off frequency and cannot very well be made variable. A set of such *LC* filters is rather costly, since the special inductances required cost at least five dollars each.

Variability is quite necessary in hi-fi work. An old scratchy disc may sound best with a 5-kc. cut-off, but with a

tape or a good LP record we need a little dial on the filter with which the cut-off frequency can quickly be run up to 15 or 20 kc.

Such a variable filter has another use: it makes an excellent high-frequency-response tester for the whole sound system, including the record. Merely crank the dial down until the degradation in music quality is just noticeable. The dial reading will then tell about what the top frequencies are in the signal. For a speaker previously measured as starting its high-frequency cut-off at 8500 cycles, for example, this listening test with the filter gave an answer of 8 kc.

The range of cut-off frequencies available with this filter makes it applicable to ham communication as well as to public address and hi-fi work. High cut-off settings of around 8 to 10 kc. will help to take the "edge" off of sound from records which have roughness on loud passages, much more effectively than the usual tone control. A cut-off setting of around 4 to 6 kc. is helpful in playing old 78 rpm records. A 2 to 4 kc. cut-off is useful in ham communication. Here the filter may be

inserted ahead of the last audio stage in a communication receiver, since it will handle around 15 volts of signal without overload; or it will work nicely in the speech amplifier system of a phone transmitter.

Performance

Since it is not practical to make the necessary chokes and capacitors variable in a conventional *LC* filter, circuits have been developed in which tubes are hooked up with variable *RC* filters to give sharp cut-off characteristics as good as those obtained with chokes. Excellent and versatile wide-range variable filters of this sort are sold commercially for a few hundred dollars each.

The filter shown in the photographs costs \$7.50 for parts, including cabinet and dial. In versatility this little gadget does not compare, of course, with the store-bought laboratory instrument, nor is the shoulder of the cut-off curve as sharp. But it is stable, reliable, effective, and quiet, and hum and noise are 70 db below normal operating level.

Characteristics of the filter described here are given in Table 1.

The unit is designed to fit in between a hi-fi preamplifier and the main amplifier, borrowing its power from the other amplifiers after the fashion of the little preamps sold for use with *G-E* phonograph pickups.

Response Curves

Fig. 2A shows the response taken at typical points along the scale of the "tuning" dial. The response is, in general, flat within ½ db up to about 80% of the cut-off frequency. Cut-off frequency is here given the usual defini-

Table 1. Characteristics of the home-built variable low-pass filter.

Cut-Off Frequency:	1700 cycles to 20 kc., continuously variable
Attenuation Rate:	18 db per octave
Type:	Low-pass only
Input Impedance:	400,000 ohms (approximately)
Output Impedance:	15,000 ohms (approximately)
Normal Operating Level:	1 volt
Hum and Noise:	200 microvolts (70 db below 1 volt)
Gain:	1
Tube Complement:	6C4 tube
External Power Required:	6.3 volts a.c. @ .15 amp.; 150-300 volts d.c. @ 2-4 ma.

tion of the frequency where the response is 3 db down. On the low end, the response is flat down to 25 cycles, and down 1 db at 10 cycles.

The shoulder or "break" of the curve where the response really starts down is slightly rounded, but not enough to affect the results discernibly for listening purposes. At settings below about 10 kc. there is a slight rise in response just before cut-off, of 1 db or less; this is unavoidable in the type of circuit used, but the effect on the ear is undetectable. When the program material is good enough, the dial will be cranked all the way up to 20 kc. anyway.

Circuitry

The principle of the circuit is simply that of an RC filter with its effective "Q" stepped up by means of a tube amplifier stage and a feedback connection. The effect of the tube on the response is shown in Fig. 2B. Curve 1 shows response or transmission characteristic of the RC filter alone, at an intermediate dial setting. It is exceedingly gradual and droopy, hardly worthy of the name of a filter. Curve 2 illustrates the same RC filter at the same dial setting, with the 6C4 tube in operation and the feedback connection completed. The curve now resembles fairly well the characteristic of a classical LC filter.

Fig. 1 is the complete circuit diagram. The RC filter circuit is simply made of three resistors and a 3-gang t.r.f.-type broadcast tuning capacitor. The output of the RC filter goes to the grid of the 6C4 tube; output of the complete device is taken from the plate of this tube. The feedback connection runs from the output back to the input of the RC filter via the 680,000 ohm resistor R_2 . A 2.2 megohm grid leak completes the d.c. circuit to ground for the grid of the 6C4.

There is nothing critical about any of the circuit constants or supply voltages. Standard $\pm 10\%$ tolerance resistors are satisfactory. The unit works equally well at plate supply voltages of 150 or 300. Plate supply voltage should be well filtered. Since the current drain is small, 2 ma. at 150 volts (4 ma. at 300 volts), the voltage can usually be "stolen" from the supply for the preamp or one of the voltage amplifiers.

If extra filtering is necessary, use a 4700 ohm $\frac{1}{2}$ -watt resistor and a 20 μ fd. electrolytic.

The gain of the unit is determined by the ratio of R_2 to R_3 . If they are equal, it will be slightly less than 1. The gain should be within 30 per-cent or so of 1, in order that the feedback factor will be somewhere near the right value to do a good job of squaring up the response curves.

The impedance of the signal source should not be too high. 200,000 ohms source impedance is the top limit, and around 50,000 ohms is better. Too-high source impedance will not only drop the gain, but it will round-off the shoulder of the response curves and make the actual cut-off frequencies lower than the dial reading. The effect on calibration is negligible at source impedances lower than 100,000 ohms. Since practically all preamp circuits have an output impedance lower than this, there should be no trouble.

Construction Details

The model shown in the photos was built in a 3"x4"x5" aluminum utility box. The tube socket is mounted on a home-made bracket which is screwed to the tuning capacitor frame. Thus most of the wiring can be done before the capacitor tube assembly is mounted in the box, although actually there is plenty of room to do it afterward. The tuning gang is mounted directly to the front "panel," for which purpose it was necessary to run a 6-32 tap through three small holes which were already present on the front of the capacitor frame. Lead dress is unimportant, since the signal level is high and the gain low.

The input terminal is a regular RCA-type phono jack on the back, and the output a phono plug on the end of a couple of feet of shielded wire.

The dial is a 2½ inch dial plate which happened to be handy, screwed to the back of a regular 1½" instrument knob. The dial plate was painted (auto enamel or most any kind of paint will do), then degreased by swabbing with a hunk of Kleenex dipped in household ammonia, then lettered with India ink. Over this, clear lacquer.

The calibration was made using an audio oscillator and a Heathkit AV-2 audio voltmeter, the original points on

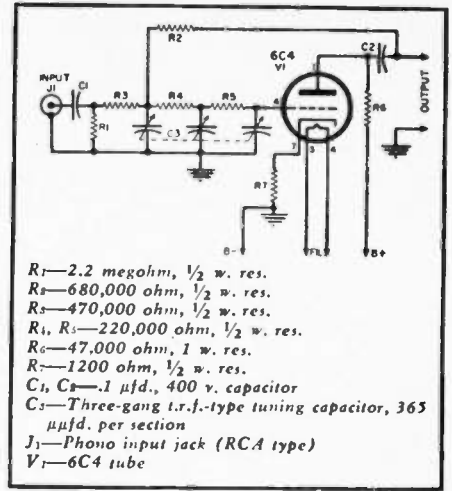


Fig. 1. Complete schematic of the variable low-pass filter. Standard, non-critical parts may be used in its construction.

the dial face being marked lightly in pencil.

Lacking an AV-2, use the low a.c. range on a Voltohmyst, Heathkit V-7, or similar v.t.v.m., but be sure the instrument is the type that uses a diode rectifier for a.c., not the older type with 1000-ohms-per-volt a.c. ranges using a copper oxide rectifier. These disc rectifiers are only good up to about 3000 cycles at best. For this reason the a.c. ranges on most regular voltohmmeters are no good for the purpose.

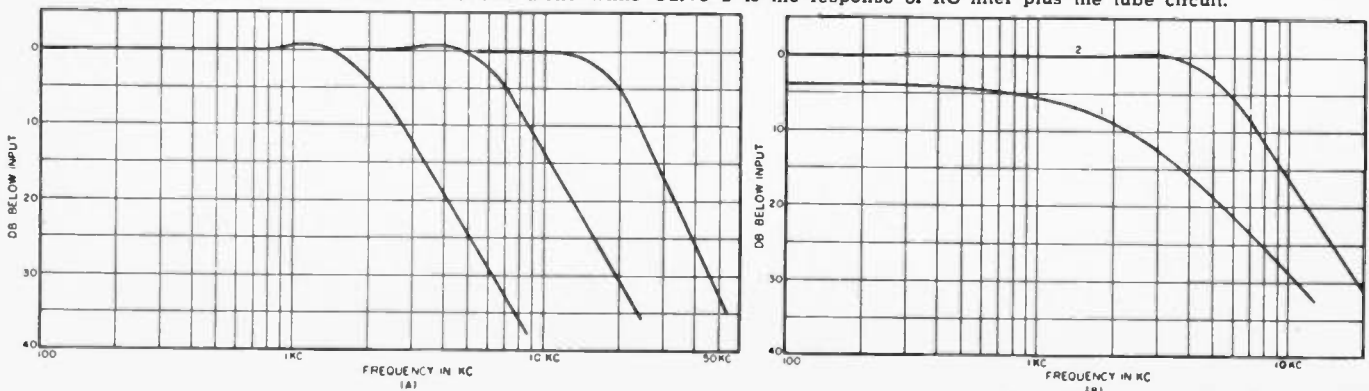
Lacking everything but an oscillator, which is indispensable, connect the just-completed filter to the main audio amplifier and improvise an output meter. Merely hang across the voice coil terminals a 1-ma. d.c. meter, a 1N34 or any other type of germanium crystal diode (not a disc rectifier), and a 1000-ohm resistor, all in series. Lower-range d.c. meters will do; increase the series resistance proportionally, e.g., 10,000 ohms for a 100-microamp meter.

Although the author originally designed this filter for use with his audio amplifier, it works beautifully on ham equipment too.

REFERENCE

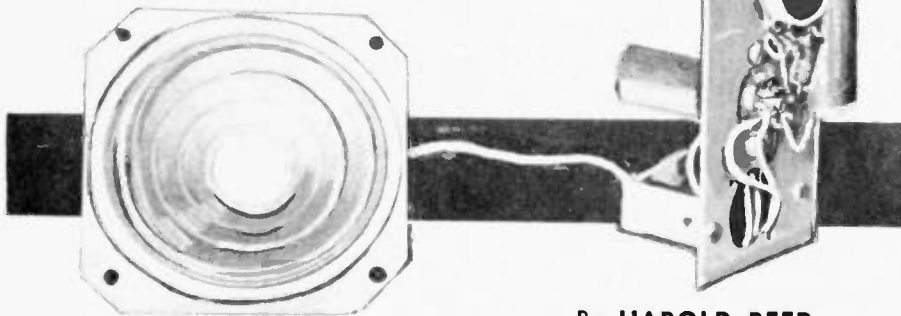
1. Fleming, Lawrence: "Acoustic Measurements for the Audiophile." RADIO & TELEVISION NEWS, September 1954.

Fig. 2 (A) Response characteristics of the filter at dial settings of 1.8, 6, and 17 kc. (B) Curve 1 shows the response characteristics of the RC filter circuit alone while Curve 2 is the response of RC filter plus the tube circuit.



A SINGLE-STAGE TAPE RECORDER MONITOR

The author's home-built monitoring amplifier and speaker. It is designed to fit into the carrying case of his recorder.



By **HAROLD REED**
Electronics Engineer
U.S. Recording Company

Details on a compact, 1-tube amplifier that is small enough to be tucked into the recorder's carrying case.

THERE are commercially available today, several good tape recorders which, although satisfactory in most other respects, have the annoying disadvantage of not being equipped with a monitoring amplifier and loudspeaker. The *Magnecord "Voyager,"* PT6-VAH is an example of such equipment.

Manufacturers have several good reasons for not including these desir-

able items in their products. First, these recorders are built to meet a certain price level, therefore, all the advantages of costlier machines cannot be included. Secondly, these units are designed with compactness and portability in mind. Third, and this ties in with the first and second reasons, if the manufacturer included a monitor amplifier and loudspeaker he would want to meet certain specifications, similar to those of the rest of his circuitry, particularly in the matter of low noise and distortion figures which would necessitate more than a single stage in order to obtain the necessary beneficial feedback requirements.

The individual electronics constructor, however, can build a monitor of the type described in this article and, keeping its limitations in mind, will find that its advantages greatly outweigh any disadvantages even though it may not be the highest of hi-fi. Its principal purpose, of course, in this case, is to eliminate the wearing of headphones when making a recording or playing back a tape for checking or editing.

This little unit was designed especially to be used with the *Magnecord "Voyager"* but, of course, could be adapted as well to other recorders. The photograph shows the experimental model constructed on a sheet of alu-

minum 3 by 5 inches, and the 4-inch loudspeaker used with it. The component parts can be built into a small "Minibox" and because no very low signal levels or high a.c. fields are involved the parts placement is non-critical.

The circuit diagram is given in Fig. 1. As can be seen in this schematic, a single 6AK6 pentode tube is used. The tube heaters in the "Voyager" are supplied through a selenium rectifier with 12 volts d.c., that is, all but the 6X4 high voltage rectifier tube. Heater supply for the 6AK6 which draws only 0.15 ampere was, therefore, taken off the 6X4 filament winding of the transformer. No hum pickup was encountered.

Plate and screen voltage for the 6AK6 was obtained directly from the 6X4 rectifier output prior to the filter section. Taking this voltage off at the input side of the rectifier filter will prevent the additional current drain of the 6AK6 from passing through the filter with the resulting greater voltage drop which would occur across the filter resistor. A separate filter section is provided for the 6AK6 by the resistor R_3 and capacitor C_2 although the filtering requirements in this case are not too great.

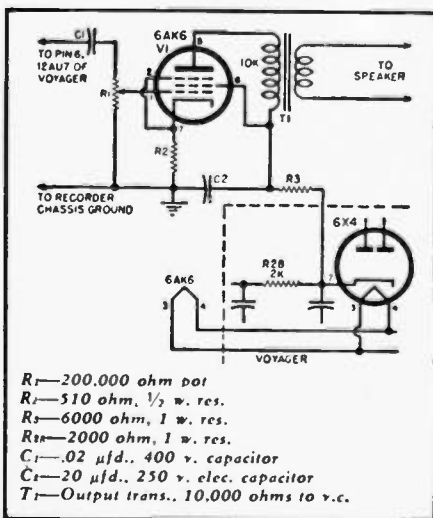
Signal voltage for the monitor is obtained at the plate of the 12AU7 output stage of the "Voyager" and coupled to the grid of the 6AK6 through a .02 μ f. capacitor and the 200,000 ohm potentiometer which serves as a volume control. This connection provides for loudspeaker monitoring during recording as well as playback operation. The output from the 6AK6 is coupled to a loudspeaker through T_1 , a small plate-to-voice coil transformer.

Frequency response and distortion measurements were made on the "Voyager" before and after the monitor was connected. With the 6AK6 monitor in operation no deterioration in the recorder characteristics was observed.

With normal recording and playback levels the power output of the monitor is approximately 1 watt. Setting the monitor volume control to obtain $\frac{1}{2}$ watt output gave ample loudspeaker output for average monitoring requirements. Although the only negative feedback available is obtained across the unbypassed cathode resistor, which contributes slightly more than 2 db, the distortion at $\frac{1}{2}$ watt output is just 2%, while at 1 watt output it increases to 3% at 1000 cycles.

Ample space was found available in the "Voyager" carrying case for both this small amplifier and the 4-inch loudspeaker.

Fig. 1. Schematic of tape recorder monitor.





a Musician Looks at Hi-Fi

The author subjects a disc to a rigorous "listening test" on the high-fidelity equipment in the studio of Custom Classics. See article for his critique of the recording.

By KURT LOEBEL*

A professional artist compares a symphonic concert-hall performance with today's high-fidelity reproduction.

IN THESE days of such extreme specialization it is not surprising that the source of musical sound has been overshadowed by scientific progress in the recording field. The musician's work is judged by a vast audience on the basis of an intricate array of knobs, tweeters, woofers, tubes, resonators, etc. To the artist-performer, orchestral player, and teacher the aim and ideal is the attainment of the "highest fidelity" in interpreting the composer's written instructions. While the musician's life is primarily directed toward the achievement of this goal, no realistic outlook on our present day musical scene can overlook what is commonly referred to as "hi-fi."

The estimate of retail sales of components for high-fidelity equipment for 1954 is \$50,000,000. The significance of this figure can not be brushed aside by even the most "non-commercial" musician. Any movement directed towards bringing finer musical reproduction and ultimately finer music into the American home, deserves wholehearted support on the part of the musician, providing it is based on the critical appraisal of the listening audience. At the same time, however, it must be understood that a considerable difference exists between a live performance and a recording session.

Aspects of a Recording Session

The limitations of the recording session are the first of many obstacles to be overcome in the reproduction of the concert hall sound by mechanical means. There are the unavoidable physical restrictions, preventing coughs, the squeaking of chairs, sneezes, hitting buttons, and the noise of turning pages. There is the psychological ten-

sion of aiming for technical perfection in view of the permanency of one's musical creation. The musician also seeks to maintain freshness and enthusiasm despite the repetition of the same passages or movements. With the use of tape and the possibility for splicing, great improvement has been made in this direction.

As far as the technical aspects of recording are concerned, at least in the author's experience, there is often a desire from the technical personnel that the performer do a minimum of modifying dynamically and balance-wise. The recording engineer's technical and musical understanding is then trusted to achieve the proper dynamic range, without destroying musical logic or stylistic traditions.

Fortunately in the recording of classical music there is no evidence that the method of recording is like that of recording crooners, where the singer uses no dynamic range and the expression is supplied entirely by the men who turn the knobs.

For the best playing results, the

* Graduate, Juilliard School of Music and Cleveland Institute of Music. Member of Cleveland Orchestra and on the faculty of the Cleveland Institute of Music.

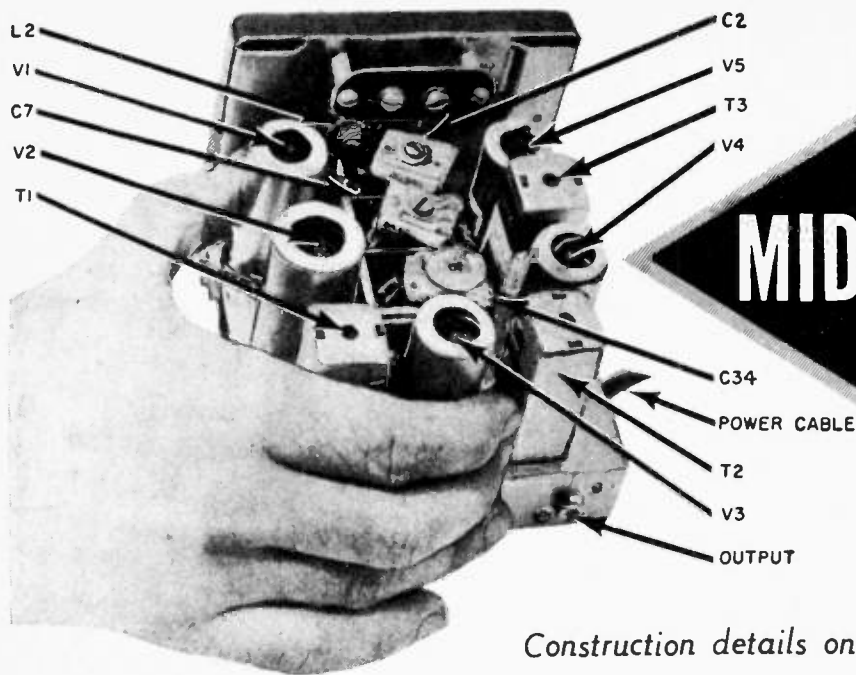
musician likes acoustical conditions which will give him freedom, ease, and power in tone production. He will favor, rather, the longer reverberation time of a good concert hall (about 1.7—2 seconds) than the comparatively dead studio with shorter reverberation time, to avoid the forcing of the string sound and overblowing of the brasses. A recent experience of the Cleveland Orchestra illustrates these problems. *Columbia's* engineers have had to adjust to the acoustics of the hall by moving the orchestra out and in front of the shell as far as possible. Then they supplemented the sound by reproducing and picking up live sound from the marble-lined foyer. Eventually a much more "live" recording sound was obtained.

Four factors would constitute a musician's recording paradise:

1. Absorbents (appropriate types of wall coverings) adjustable in relation to the reverberation time for various musical performances, such as orchestra, chamber music, and solo recital;
2. Air-conditioning controlling the moisture content of the air, which has such a vital effect on intonation;
3. Physical conditions which would enable players to hear each other well enough to achieve perfect ensemble.

4. Conditions under which perform-

(Continued on page 132)



MIDGET FM TUNER

By
ROBERT J. LARSON
 Design Engr., Jensen Mfg. Co.

Construction details on a unit which can be used with a car radio, AM or TV set, hi-fi system, or in p.a. work.

Top chassis view of the 4" x 4½" x 3½" tuner with parts identified.

RADIO listeners in most of the larger cities are missing a lot if they have no means of receiving some of the excellent programs to be found exclusively on our FM band. There is little need to mention here the noise-free, wide-range reception possible with this method; one has only to compare the relative fidelity of the AM and FM channels of some of the stereophonic musical programs being broadcast in many cities to hear the difference.

For those who would like to build a tuner for FM, here is a little unit which will be ideal for any of the following applications:

1. An FM tuner for use with presently owned high-fidelity amplifier and speaker systems.
2. FM reception in an automobile.
3. Use with a tape recorder as a source of wide-range programs and good music.
4. Connection to an existing AM-only radio receiver to permit FM reception.
5. Connection to a TV set for FM reception.
6. Use with a music system in a store or restaurant to provide background music.

Perhaps even more applications will occur to readers after perusing this article.

The entire tuner, shown in the photographs, measures only 4 by 4½ by 3½ inches high, enabling it to be tucked away within the cabinet of existing equipment. Construction is simple and no special mechanical or electrical parts are required. The total cost for parts should be about thirty dollars. The power supply is not included on this chassis, since, in many cases, power may be taken from the associated amplifier or radio. Power requirements are fairly modest as the tuner draws about 30 milliamperes at 150 volts d.c. and 1.65 amperes at 6.3

volts for the heaters. When adding this tuner to an AM receiver it is usually possible to switch off all but the audio tubes and use the additional power supply current thus made available to energize the FM tuner. This has been done successfully with the author's automobile receiver. More about this later in the article. Many hi-fi amplifiers have the extra power available for use with preamplifiers.

For those applications in which the extra power cannot be robbed from accessory equipment a suggested power supply is shown in the inset diagram of the schematic. The components for this supply will easily fit on another chassis the same size as that used for the tuner. The aluminum chassis used for the tuner measures 4 x 4½ x 1 inch and is made by ICA. Since two sides of the chassis are open the builder may decide to bend his own if materials and tools are available. The tuner uses five high gain tubes: a 6CB6 is used as a pentode r.f. amplifier, feeding a 6X8, which contains a pentode mixer and triode oscillator. Two more 6CB6's are used as 10.7 mc. i.f. stages, followed by a 6AL5 ratio detector. The tuner employs automatic gain control, causing the plate current to vary between 15 and 30 milliamperes, depending upon the signal strength of the station tuned in.

The cathode bias resistor in the 6CB6 r.f. stage is purposely left unbypassed in order to quell a tendency of the stage to oscillate. The r.f. stage is used mainly to isolate the oscillator from the antenna to minimize radiation, and to reduce the presence of images. In the 6X8 mixer, note that the suppressor grid and screen and plate bypass connections are made to the cathode. This causes the oscillator voltage to appear effectively only on

the signal grid, resulting in a highly efficient mixer stage. The use of a single tube here saves valuable space on the small chassis.

Midget 10.7 mc. i.f. transformers made by Miller are used in the i.f. stages. These little coils give adequate bandwidth without the use of swamping resistors across the transformer windings, resulting in a high gain per stage. The ratio detector provides, in addition to about one volt of audio, a negative voltage which is used for automatic gain control on all preceding stages of the tuner. About five volts are developed on local stations. R_{12} and C_{31} form a 68-microsecond roll-off network to equalize the pre-emphasis broadcast by all FM stations. Ideally, this time constant should be 75 microseconds, but it was desired to incorporate a slightly rising high-frequency response in the tuner to compensate for the attenuation caused by the shielded cable used to connect the tuner to the amplifier.

The parts layout, as shown in the top chassis photo, provides for short leads and allows the shaft of the tuning capacitor to be centered on the front panel. No volume control is included since in most installations this control is found on the associated equipment. The tuning dial used will depend on the space available. It is suggested that a dial with a large tuning ratio be used, as tuning is more critical than with AM tuners. When installing this little tuner in an AM receiver cabinet it may be possible to arrange the dial cable to turn both the AM and FM tuning capacitors simultaneously.

Shields should be used on all tubes to minimize interference of signals entering at the 10.7 mc. intermediate frequency. A grounding lug is used under one of the screws securing each tube

socket and should be soldered to the central grounding post on the tube socket; all grounds associated with each stage should then be made to this central connection. Mica-filled sockets were used on the first two stages of the tuner to increase gain and minimize drift.

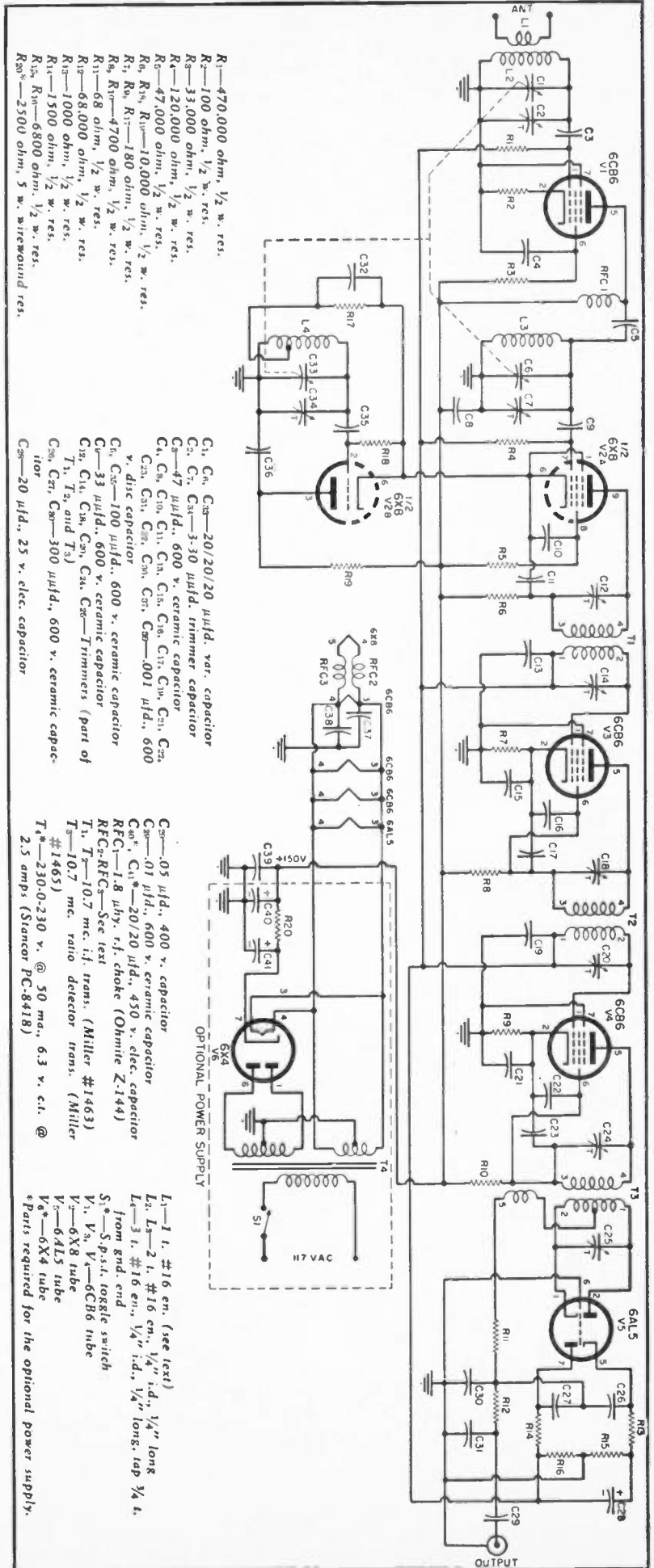
Underneath the chassis a sheet copper partition is used between the r.f. and oscillator/mixer sections of the tuner. One end of the partition is soldered to the grounding lug near the tube socket and the other end is soldered to one of the tuning capacitor ground lugs which projects through the chassis. In addition to shielding, this partition provides a low inductance connection between the capacitor and the coil, which is soldered directly to the copper partition. The r.f. coil, L_3 , and the three trimmer capacitors are mounted on top of the variable tuning capacitor. The antenna coil, L_1 , is placed adjacent to the grounded end of L_3 and is supported by a terminal strip fastened to the back of the tuning dial.

Like all equipment used at very high frequencies, this tuner requires some special wiring techniques. All leads should be as short as possible—point to point wiring is used. The new .001 μ fd. disc ceramic capacitors are excellent since the two leads are only about 3/16-inch apart and fit nicely onto the miniature tube sockets. None of these capacitors should have leads over 1/4-inch long. Buy a few extra disc ceramics because it is sometimes necessary to bypass both ends of certain long runs of wire on the chassis. This procedure is taken up again later.

The converter filament choke, RFC_2 and RFC_3 , is made by winding number 26 enameled wire over an Ohmite Z-144 choke. Notice the direction of winding on the choke and wind the number 26 wire over it in a single layer in the same direction. The extra winding is cemented in place with 912-B coil dope or Duco cement. Two Z-144 chokes may be used in the filament circuit; this bifilar winding saves space.

Probably the most interesting part of this project is the final alignment and adjustment. You will need a vacuum-tube voltmeter and a signal generator. The author used Heathkit instruments. For the initial tests, connect the tuner to a suitable power supply. A quick check should show about 2 volts at the cathode of each stage except the ratio detector. Then connect the v.t.v.m. to measure the a.g.c. voltage (across R_{10}). Inject a 10.7 mc. unmodulated signal to the grid of the mixer stage and peak the interstage i.f. transformers for a maximum v.t.v.m. reading. If a reading cannot be obtained, try connecting the signal generator to the grid of the first i.f. tube in order to peak up T_2 and T_3 . After these two transformers are aligned it will be possible to obtain a reading by reconnecting the signal generator to the mixer grid. The ratio detector is aligned by using the zero center feature of the v.t.v.m. and connecting to one end of R_{12} . The ground lead of the v.t.v.m. is connected to the tuner chassis. Adjust the top slug of T_3 so that the voltage across C_{30} (or C_{31}) is zero at 10.7 mc. and adjust the bottom slug so that an equal voltage swing is obtained on each side of 10.7 mc. as the signal generator dial is rocked back and forth across the intermediate frequency. These two adjustments interact with each other so this procedure will have to be repeated several times.

If the a.g.c. voltage remains at several volts when the signal generator is disconnected, one of the stages is oscillating. This occurrence is to be expected, and is nothing to become ex-



→ Schematic diagram of Midget FM Tuner. An optional power supply is shown within the dotted box.

cited about. Here is where those extra disc ceramic capacitors are useful. Experiment with placement of these bypasses. In the author's tuner, oscillation of the first i.f. amplifier was removed by placing a bypass at both ends of the lead running from the screen to terminal 3 on T_2 . Try bypassing both sides of the filament to ground. Try bypassing both ends of C_{20} to ground. It's safe to bypass anything except a grid or plate lead. If all else fails, try removing the capacitor between cathode and ground. This introduces degeneration and reduces the gain, however.

Alignment of the r.f. stages is easy if there is a station in your area at each end of the FM band. If not, the signal generator can be used. Since signal generator output in the 88 to 108 mc. range is usually a harmonic of some lower frequency signal, the dial calibration should be checked against some accurately calibrated receiver which tunes the 88 to 108 mc. FM band. Connect an amplifier system or earphones to the audio output jack and the antenna to L_1 . Connect the v.t.v.m. to read the a.g.c. voltage developed across R_{10} . If the coils have been wound correctly you should be able to tune in some stations. Adjustment of C_{21} should bring the stations to their proper place on the dial. If the FM band is crowded into too few degrees of tuning dial rotation, spread the turns of the oscillator coil, L_1 , apart and realign with C_{21} . Conversely, to decrease the bandwidth squeeze the turns of L_1 together and decrease the capacitance of C_{21} .

When a satisfactory spread of stations is obtained, the mixer and r.f. stages may be adjusted. The author has found the following method of tracking adjustment to be useful in many v.h.f. receivers. Tune in a signal at the high frequency end of the dial and adjust the mixer trimmer, C_7 , for maximum a.g.c. voltage. Then tune in a signal at the low end of the dial and readjust the trimmer for maximum signal. If it was necessary to increase the capacitance at the low end of the

band, squeeze the turns of the mixer coil, L_3 , together and try the high and low frequency signals again. A decrease in capacitance indicates that the coil turns should be spread apart. The process is repeated several times until maximum output is obtained at both ends of the dial with one setting of the trimmer. The same procedure is used in adjusting C_2 and L_2 in the r.f. stage.

After alignment, the tuner may be connected to the amplifier or radio in its permanent connection. When connecting to an AM radio, connect the output of the tuner across the volume control of the radio in order to be able to control the volume of the tuner as well. An automobile installation is complicated by the extra noise reducing procedures which are usually required. The antenna is somewhat of a problem also, as a horizontal FM dipole exhibits directional characteristics which sometimes cause the station to fade as the automobile turns a corner. After some experimentation the author found the best antenna to be the regular whip used by the car radio in spite of the fact that the transmitting and receiving antennas are cross polarized. Best reception occurred with the whip extended about two feet. Ignition noise, which was troublesome on some of the weaker FM stations, was considerably reduced by the use of resistor type spark plugs. If the tuner is to be used exclusively in an automobile, it is helpful to reduce the capacitance of C_{20} to 10 microfarads. This decreases the recovery time of the a.g.c. on the rapidly varying signals encountered when the automobile is in motion. The high fidelity FM transmissions may be heard to greater advantage by the use of a better speaker on the car radio. Although the usual oval automobile speaker is adequate for AM listening, better bass and treble response can be quite simply obtained by using one of the new compact loudspeaker cabinets which are now available. The author uses a *Jensen "Portable Duette"* in the rear seat of his car. The speaker is

provided with about forty feet of cable so that it may be used at a distance from the car at the beach or on picnics. This arrangement also provides entertainment while washing the car since it is not necessary to keep the windows open to hear the radio.

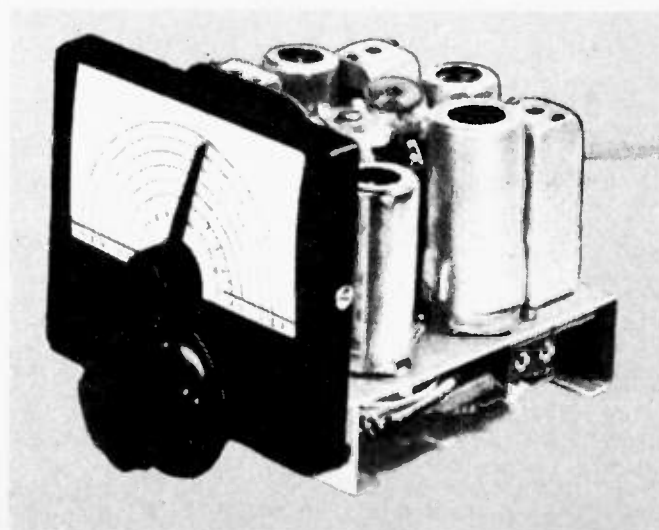
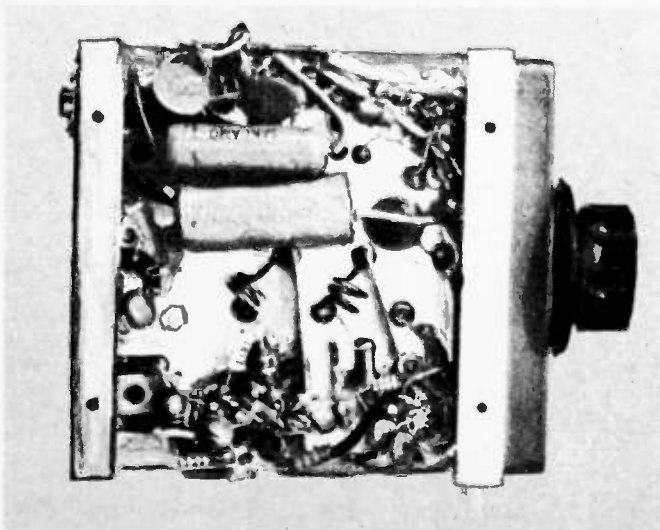
An interesting use of this tuner is made when it is used to receive the FM portion of stereophonic radio broadcasts. Several of our leading radio stations are now broadcasting musical programs in this manner. On WMAQ, in Chicago, two microphones are used in the studio, one being connected to their AM transmitter and the other to their FM. In the listener's home, two tuners, two amplifiers, and two speakers are used to recreate the original acoustical atmosphere. For those interested in receiving these broadcasts, it may be advantageous to include an AM tuner circuit on the same chassis with the midget tuner described here.

In spite of its low cost and small size, this little tuner will give many hours of enjoyment. The author's tuner has been in service for over a year, and stations over 50 miles away have been received with the use of a simple indoor folded dipole antenna. Local reception is fully as good as that normally provided by many commercial tuners.

Those builders who live more than 50 miles from an FM station may have better reception if an outdoor antenna is used. Usually, if a television signal is receivable in your area, successful FM reception is also possible. For real DX, one of the commercial yagi type FM antennas should be used. These antennas are installed and oriented in the same manner as TV antennas. If a special FM yagi is not readily available, a channel 6 TV antenna may be modified by cutting down all of the elements to 85 per-cent of their original length. Remove an equal amount of tubing from both ends of each element, of course, and flatten the ends with a hammer. This cut-down TV antenna will resonate at the middle of the FM band, but will be broad enough in its response to be useful over the entire band —50—

Under chassis view. Disc ceramics are used to conserve space.

Over-all view of the tuner which uses standard parts throughout.



RECORDED "ECHO" ADDS REALISM TO MUSIC

A HOME-TYPE tape recorder can open new fields for school and civic musical programs: cathedral-like acoustical effects from a small chorus . . . ethereal solo voices . . . dramatic orchestral performances. Professional "production numbers" can be staged with minimum sound equipment. Although the tape recorder brought inexpensive recording to groups and individuals, few realize the machine's possibilities in creating—not recordings—but special musical effects. Outlined here is one such effect which, with its many possible variations, will give the amateur producer something unique for stage, radio, or athletic field programs.

Basically the tape machine is used to cause additional controlled "liveness," or "brilliance." This may be slight, or may approach an actual echo or reverberation. Thus a small chorus in a small auditorium seems to be singing in a great cathedral. A solo voice or instrument can rise celestially above a chorus or orchestra whose music is heard in normal acoustical perspective. A narrator's voice can reverberate dramatically. Experiments will show other applications in musical shows as well as in vocal echo chamber effects for plays and programs.

Principally, you'll need a tape recorder and an amplifier. The amplifier may be a public address system unit or almost any other capable of operating a loudspeaker from microphones. It must have two separate microphone input channels; they may be either high or low impedance. The tape recorder must meet several requirements. Refer to Fig. 2 for an illustration of one general type machine. Note that the tape leaves the supply reel (1) at left. It passes over the erasing head (2) which removes the previous magnetic patterns. Next the tape crosses the recording head (3) then moves over a "capstan"—or wheel—(4) which pulls the tape along and determines its speed of travel. The take-up reel (5) stores the tape for re-winding and playback. On playback the tape travels the same path but with the erase head idle and the recording head becoming a playback pickup. (The erasing head and recording head may be mounted in a single housing, but a close examination of the surface over which the tape moves will show two separate heads contacting the tape.) The same amplifier within the machine serves in turn as recording driver and playback amplifier.

For the special effects to be described a tape recorder must have an additional head, making three in all: erase, recording, and a separate playback. Some of the more expensive machines already have this arrangement. If your machine is of this type you may bypass the next four paragraphs which deal



By JACK THORNTON

Fig. 1. The actual wiring has been replaced by white cords for purposes of illustration. Additional pickup is seen above regular heads. Steel spring under a head mounting bolt holds the extra head in place. See article.

Your home tape recorder can, with slight adjustment, be used to provide interesting and dramatic "sound effects."

with adding an auxiliary head to other machines.

Referring again to Fig. 2 you'll see the tape is exposed along its path of travel from the recording-playback head to the capstan and take-up reel. This is necessary in adding the extra head. In many machines the tape is readily accessible. In others it may be exposed by removing a metal housing. With a few machines the tape runs in a slot which is an integral part of the recorder's top surface. This arrangement is not adaptable unless the tape is accessible from below, from inside the machine.

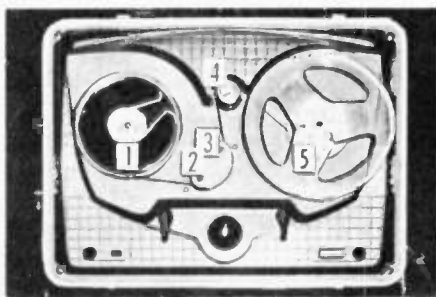
A further requirement demands that there be space between the machine's existing record-playback head and its drive capstan, or between the drive capstan and the take-up reel. About an inch-and-a-quarter to an inch-and-a-half space is needed, depending on the size of the extra head. Fig. 1 will give you an idea regarding the space required and the mounting of the additional head. It should not be more

than two inches from the existing head for best results.

If your machine fills these requirements you're ready to proceed. The extra head should be a *single* one—playback only. A full-track model is preferred and of the smallest possible physical size. Before ordering, check the amplifier you plan to use as a part of this setup. Note whether it has high-impedance microphone inputs, or low-impedance. (A radio repair shop can clear any doubt about this.) Obtain a playback head to match the amplifier's microphone input impedance. Many makes and models in pickup heads are available. It is suggested that you outline your needs to a local radio technician. Usually he will have catalogues describing heads in some detail and often picturing them. In high-impedance design the full track unit for a *Concertone* machine is a possibility. *Brush* manufactures a pickup in low-impedance rating.

Mount the extra head about one-and-one-half to two inches past the existing record-playback head if possible. The additional head of the machine in Fig. 1 is clamped down with a steel spring held under one of the original-equipment head's mounting screws. If space is available, the extra head may be mounted with wing-bolts through a slot cut in the machine's top surface. This will allow adjustment of distance for best effect. Take care that the pole-pieces on the working surface of the head contact the tape squarely. Then to the electrical terminals on the head attach a shielded wire of the type used to connect a phonograph pickup with its amplifier. On the other end of the

Fig. 2. A tape recorder of the type suitable for adaptation. The numbers on the photo are referred to in the text proper.



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wire attach a plug which will fit your amplifier's microphone input. Keep the wire short to avoid hum pickup.

If your machine already is equipped with the additional head you may connect it to the amplifier in either of two ways. You can detach the wires from the tape machine to the playback head, then attach the head to the auxiliary amplifier as described in the preceding paragraph. Or, you can wire from the tape machine's playback amplifier output to the auxiliary amplifier's input. With the second method it may be necessary to insert a transformer or resistor in the wire between the tape machine and amplifier to cut down the volume and prevent distortion.

Now to the circuits for the artificial brilliance effects. They are based on electrical feedback with time delay. The sound for echo is fed through an amplifier, recorded, picked up on the extra tape playback head, and fed back into a second channel of the original amplifier. Both the original sound and the recorded-played-back sound are heard in the loudspeaker. Thus, as the original sound picked up by the microphone is heard in the loudspeaker it is also being recorded. The tape takes a fraction of a second to travel from the recording head to the pickup head. When this delayed sound is fed into the amplifier it is heard in the speaker and also is re-recorded, but with diminished volume. As the sounds thus "chase their tails" the total effect reminds one of the acoustical conditions in a large stone hall.

Examine Fig. 1. Notice that the microphone feeds the vocal music to the public address amplifier's microphone channel one. The signal from the amplifier output is heard in the loudspeaker. The same signal is supplied to the tape recorder by a wire from the amplifier output to the tape machine input. (On many machines the proper input is marked "radio." Do not use the microphone input.) The tape machine is operated on "record." The output of the extra head is fed to the amplifier's microphone channel two, where it mixes with the original sound to form the final effect.

Channel two's volume control will control the amount of feedback or "echo." The time-delay involved is set by varying the space between the recording head and the playback pickup. The greater the distance the longer the delay. The speed of the machine would affect this too. Slower tape speed means longer delay. A speed of 7½ inches-per-second is suggested for heads spaced one to two inches apart.

If sufficient volume cannot be obtained from the extra head working into a microphone channel, a preamplifier can be inserted between the head and the amplifier.

Although this article has dealt with "live" performances, the same technique can be used for recordings. The same tape used in producing the effect will contain a recording of it. You'll find many uses for this setup in adding a bit of spice to musical and dramatic productions.

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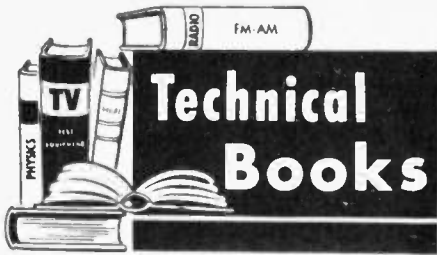
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"THE NEW HIGH FIDELITY HANDBOOK" by Irving Greene & James R. Radcliffe. Published by *Crown Publishers, Inc.*, New York, 185 pages. Price \$4.95.

The lay audiophile who has been staggered by the complexity and technical level of most texts on the subject will welcome the appearance of this authoritative yet easy-to-read treatise.

There are no prerequisites to an understanding and enjoyment of this book. Mathematical treatment has been completely eliminated in favor of explanatory text material. The book itself is divided into fifteen chapters, each a hard-hitting exposition of some phase of the subject. One of the most encouraging things to the lay reader is the authors' realism regarding hi-fi systems. While they set high standards for the assembling of the components of the ideal hi-fi system they are realistic enough to appreciate the fact that not all music lovers have unlimited funds to expend on systems for playing their records. To this end both moderately-priced and high-priced units are analyzed and described.

The budget-minded music lover will also appreciate the large and detailed section covering the construction of speaker enclosures and the techniques required to assemble such cabinets. Troubleshooting, servicing, and maintenance procedures for the owner are also covered in some detail, along with information for building simple test equipment with which to make the requisite checks.

One especially valuable feature of the book is a buyers' guide covering manufacturers and distributors of hi-fi equipment of all types.

This book meets the real need expressed by that large segment of the audio "fraternity" that enjoys its music but does not have a string of impressive engineering degrees to back up its interest.

* * *

"PRACTICAL ELECTROACOUSTICS" by M. Rettinger. Published by *The Chemical Publishing Co., Inc.*, New York, N. Y. 266 pages. Price \$10.00.

This is an engineering handbook for the serious audioman and fills a hiatus in the literature. Of necessity the treatment is mathematical but those with a working knowledge of advanced high school and college math could handle the formulas.

The text material is divided into eight chapters and covers microphones, loudspeakers, circuits, magnetic struc-

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tures, public address systems, vibrations, architectural acoustics, and magnetic recording. Four valuable appendices covering octaves; decibels, volume units, dbm *versus* watts; dbm *versus* voltage; and a bibliography complete the book.

The lavish use of graphs, charts, and schematic diagrams contributes to the practical value of this handbook. The engineer entrusted with the job of planning various types of sound installations—whether for a huge outdoor amphitheater or for a family living room—will find this book of great assistance in coping with his particular problems.

"YOUR TAPE RECORDER" by Robert and Mary Marshall. Published by Greenberg, Publisher, New York. 273 pages. Price \$4.95.

Subtitled "How to Select One and Get the Most Out of It," this volume is directed to the non-technical user of home recorders. Today tape equipment turns up in the hands of such diverse consumers as clergymen, Scout leaders, teachers, housewives, schoolchildren, college students, and club members. The authors of this volume acknowledge and recognize the fact that, for the most part, such users do not have the technical background or training of a recording engineer and that for them to get the maximum benefit and fun from their equipment they must have a non-technical handbook for guidance.

A wide assortment of commercial tape machines is illustrated and described along with the salient features of their operation. A directory of recorder and accessory equipment manufacturers is appended for the benefit of the person who is still "looking around" for the recorder to meet his needs.

The imaginative applications for tape recorders form an interesting and instructive part of this text and those who think of such devices only as a means of preserving Junior's first Sunday-School recitation will be amazed at the wide variety of uses for such gear.

"ELECTRONIC MUSICAL INSTRUMENTS" by Richard H. Dorf. Published by Radio Magazines, Inc., P. O. Box 629A, Mineola, N. Y. 321 pages. Price \$7.50.

For all those electronic musical instrument fans, here is a text which covers both commercially-available organs and the circuitry for two complete instruments that can be home-built.

There are four introductory chapters which cover the nature of music and musical instruments; basic musical facts; requirements of an instrument; and tone color.

Details and schematics are given for Hammond, Baldwin, Minshall, Connsonata, Lowrey, Hammond "Solovox", Wurlitzer, Hammond Chord Organ, Allen, and Strobocorn instruments. There is a chapter on installing and servicing

such instruments and then details for constructing the "Thyratone" and the "Electronorgan". Three additional chapters are devoted to a discussion of electron-tube tone generator circuits, non-tube tone generators, tone coloring, amplification and control. An appendix listing electronic music patents and a bibliography complete the text.

"THE FABULOUS PHONOGRAPH" by Roland Gelatt. Published by J. B. Lippincott Company, Philadelphia. 310 pages. Price \$4.95.

The subtitle "From Tin Foil to High Fidelity" is indicative of the scope of this volume. Covering the entire phonographic era from 1877 to the present, this book is somewhat of a *tour de force*. The author, who is New York Editor of *High Fidelity* magazine, spent years collecting his material and verifying his findings with many of the principals involved in the early experiments with recorded sound.

While most people are aware that all was not smooth sailing for Edison and his successors, few realize the downright "skullduggery" involved as well as the legal and commercial battles that shaped up before the industry was "tamed." In addition to the claims and counterclaims that were wafted around, the recording artists themselves added many colorful chapters to phono history.

Mr. Gelatt has told his story interestingly and well, drawing upon numerous anecdotes to point up his text. We believe that modern audiophiles will find this fascinating, as well as giving them a true appreciation of the fact that "they never had it so good." The text material is illustrated with a number of historic photographs and some reproductions of the original sketches of equipment as submitted by the inventors in connection with their patent applications.

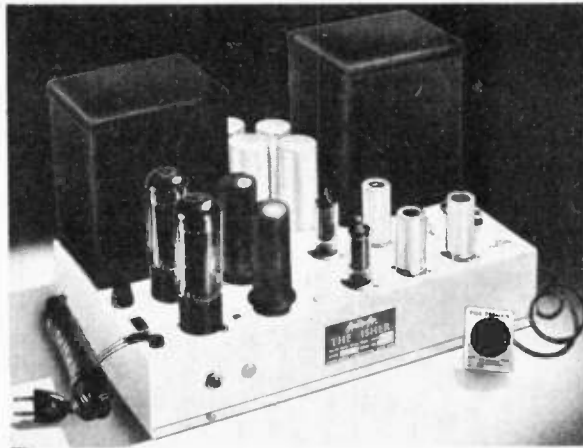
"AUDIO AMPLIFIERS AND ASSOCIATED EQUIPMENT" by Sams Staff. Published by *Howard W. Sams & Co., Inc.*, Indianapolis. \$3.95. Paper bound. Vol. 5 (AA-5).

This newest volume in the *Sams* "Amplifier" series covers 1953 and 1954 model audio amplifiers, preamps, and AM-FM custom tuners.

Each unit is pictured, controls are identified, tubes are listed and the power supply and rating given in tabular form. Under chassis and top chassis views are also provided along with complete parts lists and circuit diagrams.

Where pickup arms are part of the unit, complete details on the correct cartridges and needles to be used are also given.

This volume also contains a cumulative index covering all of the volumes in the series thus far for the speedy location of the desired schematic and service data. This listing is divided into amplifiers and tuners for further ease in spotting the correct diagram.



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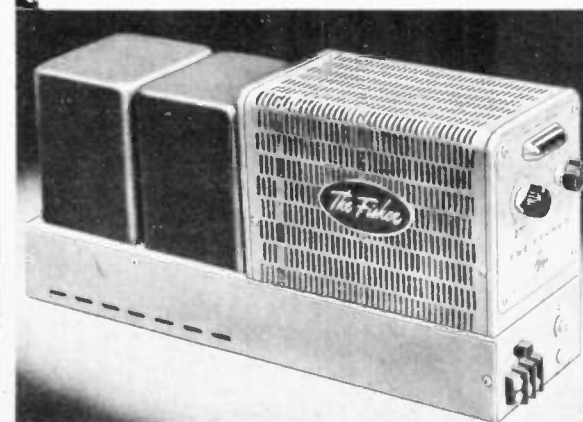
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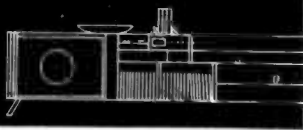


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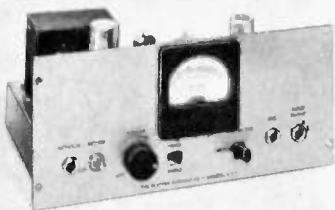


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- A Amplifier, preamps., etc.
 - B Disc (recorded)
 - C Pickup arms
 - D Record Player
 - E Tape (recorded)
 - F Tape Recorder, playback only, etc.
 - G Tuners
- (Italics apply only to trade name of recorded tapes.)

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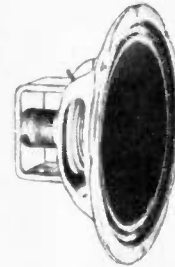
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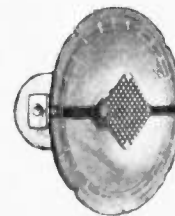


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DUOTONE CO., Locust St., Keyport, N. J. (C)
ELECTRON ENTERPRISES, 6917 W. Stanley Ave., Beverly, Ill. (B)
ELECTRO-SONIC LABS., 35-54 36th St., Long Island City, N. Y. (A, D)
ELECTRO-VOICE, INC., Buchanan, Mich. (A)
ELECTROVOX CO., INC. (WALCO PRODUCTS), 60 Franklin St., East Orange, N. J. (C)
FAIRCHILD RECORDING EQUIP. CO., Whitestone 57, N. Y. (A, B, D)
FANON ELECTRIC CO., INC., 150-09 South Rd., Jamaica 33, N. Y. (B, E)
FENTON CO., 15 Moore St., New York, N. Y. (A, B, E)
FERRANTI ELECTRIC, INC., 30 Rockefeller Plaza, New York 20, N. Y. (A, D)
GARRARD SALES CORP., 80 Shore Rd., Port Washington, N. Y. (B, E)
GENERAL ELECTRIC CO., Electronics Park, Syracuse, N. Y. (A, C, D)
GENERAL INDUSTRIES CO., THE, Olive and Taylor Sts., Elyria, Ohio. (B)
GRAYLINE CO., THE, 12243 S. Ave. "O," Chicago 33, Ill. (B)
GRAY RESEARCH & DEVELOPMENT CO., INC., 658 Hilliard St., Manchester, Conn. (D)
GROMMES DIV. OF PRECISION ELECTRONICS, INC., 9101 King Ave., Franklin Park, Ill. (E)
GROSSMAN RADIO & ELECTRIC CO., 81 Sprink St., New York, N. Y. (E)
JENSEN INDUSTRIES, INC., 7333 W. Harrison St., Forest Park, Ill. (C)
KINGDOM PRODUCTS, LTD., 23 Park Pl., New York 7, N. Y. (D)
KRAFT BROS., INC., 1299 Jerome Ave., New York 52, N. Y. (B)
KRAL-PRODUCTS, 1704 Walnut St., Philadelphia 2, Pa. (D)
LIPPS ENGINEERING, EDWIN A., 5485 W. Washington Blvd., Los Angeles 16, Calif. (C)
LIVINGSTON ELECTRONIC CORP., Livingston, N. J. (D)
MAJOR ELECTRONICS CO., 42 Meserole St., Brooklyn, N. Y. (B, E)
MARKEL ELECTRIC PRODUCTS, INC., 145 Seneca St., Buffalo, N. Y. (E)
MP ENGINEERING CO., Fairfield 3, Conn. (B, D)
NATIONAL COMPANY, INC., 61 Sherman St., Malden 48, Mass. (E)
PERMO, INC., 6415 N. Ravenswood Ave., Chicago 26, Ill. (C)
PFANSTIEHL CHEMICAL CORP., 104 Lake View Ave., Waukegan, Ill. (A, C, D)
PICKERING AND CO., INC., 309 Woods Ave., Ocean-side, L. I., N. Y. (A, C, D)
PRESTO RECORDING CORP., P. O. Box 500, Paramus, N. J. (B, C)
RADIO CORP. OF AMERICA, ENGINEERING PRODUCTS DIV., Front and Cooper Sts., Camden, N. J. (E)
RADIO MUSIC CORP., 84 S. Water St., Port Chester, N. Y. (A, D)
RECOTON CORP., 52-35 Barnett Ave., Long Island City 4, N. Y. (A, C)
REK-O-KUT CO., 38-01 Queens Blvd., Long Island City 3, N. Y. (B)
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SCOTT, HERMON HOSMER, INC., 385 Putnam Ave., Cambridge 39, Mass. (B)
SHURE BROTHERS, INC., 225 W. Huron St., Chicago, Ill. (A, C, D)
SIGHTMASTER CORP., 111 Cedar St., New Rochelle, N. Y. (A, D)
SONOTONE CORP., Elmsford, N. Y. (A, C)
TANNOY (AMERICA), LTD., 38 Pearl St., New York 4, N. Y. (A)
TELEX CO., THE, 46 Lakeview Ave., Yonkers, N. Y. (C)
TETRAD CO., 62 St. Mary St., Yonkers 2, N. Y. (C)
THORENS CO., New Hyde Park, N. Y. (B, E)
TURNER CO., THE, 909 17th St., N. E., Cedar Rapids, Iowa. (A)
UNITED AUDIO PRODUCTS DIV., UNITED OPT. MFG. CORP., 202-4 E. 19th St., New York 3, N. Y. (A, B, E)
U. S. RECORDING CO., 1121 Vermont Ave., N. W., Washington 5, D. C. (B)
WEATHERS INDUSTRIES, INC., 66 E. Gloucester Pike, Barrington, N. J. (A, D)
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B Without preamplifier
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- B & R ELECTRONICS, 1701 Boone Ave., New York 60, N. Y. (A)
- BEAM INSTRUMENTS CORP., 350 Fifth Ave., New York 1, N. Y. (A, B)
- BELL SOUND SYSTEMS, INC., 555 Marion Rd., Columbus 7, Ohio. (A, B)
- BIGG OF CALIFORNIA (BIGG RADIO CO.), 2506 W. Washington Blvd., Los Angeles, Calif. (A, B)
- BOGEN CO., INC., DAVID, 29 Ninth Ave., New York 14, N. Y. (A, B)
- BRAINARD ELECTRONICS, 8586 Santa Monica Blvd., Los Angeles 84, Calif. (D, E)
- BROCINER ELECTRONIC CORP., 344 E. 32nd St., New York 16, N. Y. (E)
- CLEGG LABORATORIES, INC., Hanover Rd. & Ridgedale Ave., Morris Township, N. J. (A, C, D, E)
- DICTOGRAPH PRODUCTS INC., 95-25 149th St., Jamaica 35, N. Y. (D, E)
- DUBROW DEVELOPMENT CO., 235 Penn St., Burlington, N. J. (C)
- DYNAMO MAGNETRONICS CORP., DIV. OF THE MAICO CO., INC., 21 N. 3rd St., Minneapolis 1, Minn. (E)
- ELECTRONIC INSTRUMENT CO., INC. (EICO), 84 Withers St., Brooklyn 11, N. Y. (E)
- ELECTROSONIC SPECIALTIES, 7230 Clinton Rd., Upper Darby, Pa. (E)
- ELECTRO-VOICE, INC., Buchanan, Mich. (E)
- EMC RECORDINGS CORP., 808 E. Seventh St., St. Paul 8, Minn. (E)
- EPCO ELECTRONICS, INC., 140 Liberty St., New York 8, N. Y. (D, E)
- FAIRCHILD RECORDING EQUIPMENT CO., White-stone 57, N. Y. (E)
- FANON ELECTRIC CO., INC., 150-09 South Rd., Jamaica 33, N. Y. (E)
- FEMCO, INC., Box 32, Irwin, Penna. (E)
- FISHER RADIO CORP., 21-21 44th Dr., Long Island City 1, N. Y. (A, B)
- GENERAL ELECTRIC CO., Electronics Park, Syracuse, N. Y. (B)
- GORDON ENTERPRISES, 5362 N. Cahuenga Blvd., N. Hollywood, Calif. (A, B)
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- GROMMES DIV. OF PRECISION ELECTRONICS, INC., 9101 King Ave., Franklin Park, Ill. (A, B)
- GROSSMAN RADIO & ELECTRIC CO., 81 Sprink St., New York, N. Y. (A, B)
- HAMILTON ELECTRONICS, 2726 Pratt Ave., Chicago 45, Ill. (A, B)
- HARMAN KARDON INC., 520 Main St., Westbury, L. I., N. Y. (A)
- HEATH CO., 305 Territorial Rd., Benton Harbor, Mich. (A, B)
- JAMES INSTRUMENT LABORATORY, 9110 S. 52nd Ct., Oak Lawn, Ill. (B)
- LOGE SOUND ENGINEERS, J. M., 2171 W. Washington Blvd., Los Angeles 18, Calif. (A, B)
- MAJOR ELECTRONICS CO., 42 Messerole St., Brooklyn, N. Y. (A, B)
- MARANTZ COMPANY, 44-15 Vernon Blvd., Long Island City 1, N. Y. (B)
- MARTIN AND CO., H. S., 1916 Greenleaf St., Evanston, Ill. (A)
- McINTOSH LAB., INC., 320 Water St., Binghamton, N. Y. (B)
- MP ENGINEERING CO., Fairfield 3, Conn. (A, B)
- NATIONAL COMPANY, INC., 61 Sherman St., Malden 48, Mass. (A, B)
- NEWCOMB AUDIO PRODUCTS CO., 6824 Lexington Ave., Hollywood 38, Calif. (A)
- PALMER, INC., M. V., 4002 Fruit Valley Rd., Van-couver, Wash. (A, B)
- PARAFLEX CORP., 10068 Franklin Ave., Franklin Park, Ill. (B)
- PILOT RADIO CORP., 37-06 36th St., Long Island City 1, N. Y. (A, B)
- RADIO CORP. OF AMERICA (ENGINEERING PRO-DUCTS DIV.), Front & Cooper Sts., Camden, N. J. (A, B)
- RADIO CRAFTSMEN, INC., THE, 4401 N. Ravens-wood Ave., Chicago 40, Ill. (A, B)
- RADIO MUSIC CORP., 84 S. Water St., Port Ches-ter, N. Y. (A, B)
- RAULAND-BORG CORP., 3515 W. Addison St., Chi-cago 18, Ill. (A)
- REGENCY DIVISION, I. D. E. A., INC., 7900 Pen-dleton Pike, Indianapolis 26, Ind. (A, B)
- SARGENT-RAYMENT CO., THE, 1401 Middle Harbor Rd., Oakland 20, Calif. (B)
- SCOTT, INC., HERMON HOSMER, 385 Putnam Ave., Cambridge 39, Mass. (A, B)
- SHERWOOD ELECTRONICS LABORATORIES, INC., 2802 V. Cullom Ave., Chicago 18, Ill. (A)
- SHRADER SOUND, INC., 2803 M St., N. W., Wash-ington 7, D. C. (A, B)
- SIGHTMASTER CORP., 111 Cedar St., New Rochelle, N. Y. (A, B)
- SONEX, INC., 245 Sansom St., Upper Darby, Penna. (A, B)
- SONOTONE CORP., Elmsford, N. Y. (B)
- SPECIAL PRODUCTS CO., P. O. Box 188, Rockville, Md. (A, B)
- STEPHENS MANUFACTURING CORP., 8538 Warner Dr., Culver City, Calif. (B)
- STROMBERG-CARLSON CO., SOUND DIV., 1225 Clif-ford Ave., Rochester, N. Y. (A, B)
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- THORDARSON-MEISSNER MANUFACTURING DIV., 7th & Belmont Ave., Mt. Carmel, Ill. (C, E)
- UNITED TRANSFORMER CO., 150 Varick St., New York 13, N. Y. (B)
- U. S. RECORDING CO., 1121 Vermont Ave., N. W., Washington 5, D. C. (A, B)
- WEBSTER ELECTRIC CO., 1900 Clark St., Racine, Wis. (B)
- WELLS-GARDNER & CO., 2701 N. Kildare Ave., Chi-cago 39, Ill. (A, B)
- ALTEC LANSING CORP., 9356 Santa Monica Blvd., Beverly Hills, Calif. (E)
- APPROVED ELECTRONIC INSTRUMENT CORP., 928 Broadway, New York 10, N. Y. (E)
- BEAM INSTRUMENTS CORP., 350 Fifth Ave., New York 1, N. Y. (E)
- BIGG OF CALIFORNIA (BIGG RADIO CO.), 2506 W. Washington Blvd., Los Angeles, Calif. (A, B, D, E)
- BOGEN CO., INC., DAVID, 29 Ninth Ave., New York 14, N. Y. (E)
- BRAINARD ELECTRONICS, 8586 Santa Monica Blvd., Los Angeles 84, Calif. (D, E)
- BROCINER ELECTRONIC CORP., 344 E. 32nd St., New York 16, N. Y. (E)
- CLEGG LABORATORIES, INC., Hanover Rd. & Ridgedale Ave., Morris Township, N. J. (A, C, D, E)
- DICTOGRAPH PRODUCTS INC., 95-25 149th St., Jamaica 35, N. Y. (D, E)
- DUBROW DEVELOPMENT CO., 235 Penn St., Bur-lington, N. J. (C)
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- ELECTRONIC INSTRUMENT CO., INC. (EICO), 84 Withers St., Brooklyn 11, N. Y. (E)
- ELECTROSONIC SPECIALTIES, 7230 Clinton Rd., Upper Darby, Pa. (E)
- ELECTRO-VOICE, INC., Buchanan, Mich. (E)
- EMC RECORDINGS CORP., 808 E. Seventh St., St. Paul 8, Minn. (E)
- EPCO ELECTRONICS, INC., 140 Liberty St., New York 8, N. Y. (D, E)
- FAIRCHILD RECORDING EQUIPMENT CO., White-stone 57, N. Y. (E)
- FANON ELECTRIC CO., INC., 150-09 South Rd., Jamaica 33, N. Y. (E)
- FEMCO, INC., Box 32, Irwin, Penna. (E)
- FISHER RADIO CORP., 21-21 44th Dr., Long Island City 1, N. Y. (A, C, D, E)
- GENERAL ELECTRIC CO., Electronics Park, Syracuse, N. Y. (C, E)
- GOTHAM TELEVISION CORP., 123 W. 64th St., New York, N. Y. (D, E)
- GROMMES DIV. OF PRECISION ELECTRONICS, INC., 9101 King Ave., Franklin Park, Ill. (D, E)
- GROSSMAN RADIO & ELECTRIC CO., 81 Sprink St., New York, N. Y. (E)
- HAMILTON ELECTRONICS, 2726 Pratt Ave., Chicago 45, Ill. (A)
- HARMAN KARDON INC., 520 Main St., Westbury, L. I., N. Y. (E)
- HEATH CO., 305 Territorial Rd., Benton Harbor, Mich. (E)
- LANGEVIN MFG. CORP., 47-37 Austell Pl., Long Island City 1, N. Y. (A, C, D, E)
- LOGE SOUND ENGINEERS, J. M., 2171 W. Washing-ton Blvd., Los Angeles 18, Calif. (A, E)
- MARANTZ CO., 44-15 Vernon Blvd., Long Island City 1, N. Y. (E)
- MARTIN AND CO., H. S., 1916 Greenleaf St., Evan-son, Ill. (E)
- McINTOSH LAB., INC., 320 Water St., Binghamton, N. Y. (A, D, E)
- MILLER CO., W., 5917 S. Main St., Los Angeles 3, Calif. (C)
- MP ENGINEERING CO., Fairfield 3, Conn. (A, D, E)
- NATIONAL COMPANY, INC., 61 Sherman St., Malden 48, Mass. (E)
- NEWCOMB AUDIO PRODUCTS CO., 6824 Lexington Ave., Hollywood 38, Calif. (E)
- ORTHO FILTER CORP., 196 Albion Ave., Paterson, N. J. (C)
- PALMER, INC., M. V., 4002 Fruit Valley Rd., Van-couver, Wash. (A, C)
- PENTRON CORP., THE, 777 S. Tripp, Chicago 24, Ill. (E)
- PFANSTIEL CHEMICAL CORP., 104 Lake View Ave., Waukegan, Ill. (D, E)
- PICKERING AND CO., INC., 309 Woods Ave., Ocean-side, L. I., N. Y. (D, E)
- PILOT RADIO CORP., 37-06 36th St., Long Island City 1, N. Y. (D, E)
- PRECISE DEVELOPMENT CORP., 2 Neil Court, Oceanside, N. Y. (E)
- PRESTO RECORDING CORP., P. O. Box 500, Paramus, N. J. (D, E)
- RADIO CORP. OF AMERICA (ENGINEERING PRO-DUCTS DIV.), Front and Cooper Sts., Camden, N. J. (E)
- RADIO CRAFTSMEN, INC., THE, 4401 N. Ravens-wood Ave., Chicago 40, Ill. (C, E)
- RADIO MUSIC CORP., 84 S. Water St., Port Ches-ter, N. Y. (A, C, D, E)
- R. AND B. ELECTRONICS CO., Box 6998, Washington 20, D. C. (E)
- REGENCY DIVISION, I. D. E. A., INC., 7900 Pen-dleton Pike, Indianapolis 26, Ind. (A, C)
- SCOTT, INC., HERMON HOSMER, 385 Putnam Ave., Cambridge 39, Mass. (B, D, E)
- SHRADER SOUND, INC., 2803 M St., N. W., Wash-ington 7, D. C. (A)
- SONEX, INC., 245 Sansom St., Upper Darby, Penna. (D, E)
- SONOTONE CORP., Elmsford, N. Y. (E)
- SPECIAL PRODUCTS CO., P. O. Box 188, Rockville, Md. (A, B)
- STROMBERG-CARLSON CO., SOUND DIV., 1225 Clif-ford Ave., Rochester, N. Y. (A, E)
- SUMMIT ELECTRONICS, INC., 7 Industrial Pl., Sum-mit, N. J. (A, B, E)
- SWITCHCRAFT, INC., 1328 N. Halsted St., Chicago 22, Ill. (A)
- TANNOY (AMERICA), LTD., 38 Pearl St., New York 4, N. Y. (E)
- TECH-MASTER CORP., 75 Front St., Brooklyn, N. Y. (E)
- TELECTROSONIC CORP., 35-18 37th St., Long Island City 1, N. Y. (A, E)
- THORDARSON-MEISSNER MANUFACTURING DIV., 7th & Belmont Ave., Mt. Carmel, Ill. (C, E)
- TRI-DEX CO., P. O. Box 1207, Lindsay, Calif. (E)
- UNITED TRANSFORMER CO., 150 Varick St., New York 13, N. Y. (C, D)
- U. S. RECORDING CO., 1121 Vermont Ave., N. W., Washington 5, D. C. (A, E)
- WEBSTER ELECTRIC CO., 1900 Clark St., Racine, Wis. (E)

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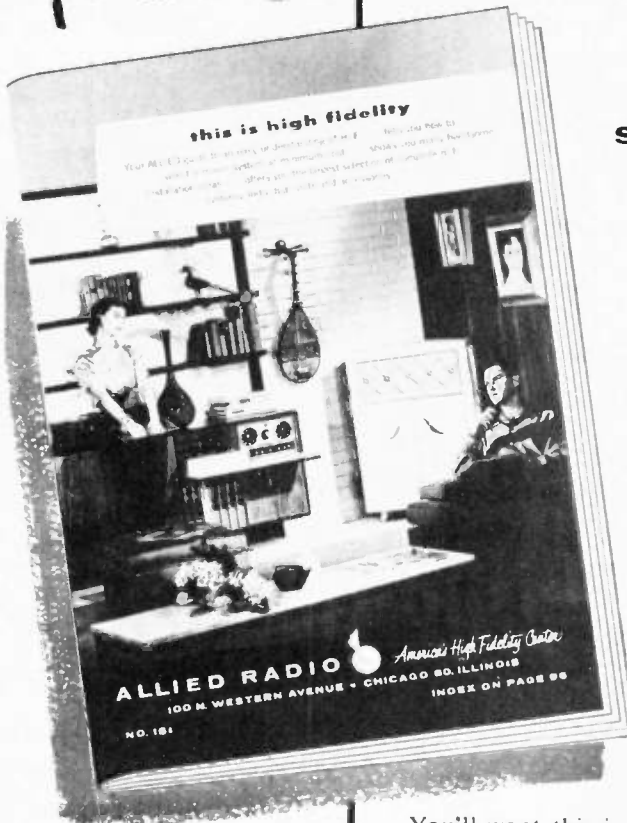
SPEAKERS AND EQUIPMENT

- A Crossover Networks
- B Enclosures—Baffles (speaker)
- C Mid-Range
- D Tweeters (cone type)
- E Tweeters (horn type)
- F Wide-Range
- G Woofers
- H Tweeters—Electrostatic

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 ALTEC LANSING CORP., 9356 Santa Monica Blvd., Beverly Hills, Calif. (A, B, E, F, G)
 AMI, INC., 1500 Union Ave., S. E., Grand Rapids, Mich. (A, B)
 ANGLE GENESSEE CORP., 107 Norris Dr., Rochester 10, N. Y. (B)
 ARGOS PRODUCTS, 310 Main St., Genoa, Ill. (B)
 ATLAS SOUND CORP., 1449 39th St., Brooklyn 18, N. Y. (A, C, E)
 BEAM INSTRUMENTS CORP., 350 Fifth Ave., New York 1, N. Y. (A, B, C, E, F, G)
 BENNETT & CO., PAUL, 230 E. Ohio St., Chicago 30, Ill. (D, E, F, G)
 BIGG OF CALIFORNIA (BIGG RADIO CO.), 2506 W. Washington Blvd., Los Angeles, Calif. (A, B)
 BOGEN CO., INC., DAVID, 29 Ninth Ave., New York 14, N. Y. (B)
 BOZAK SALES CO., THE R. T., P. O. Box 966, Darien, Conn. (A, B, C, D, G)
 BROCIER ELECTRONICS CORP., 344 E. 32nd St., New York 16, N. Y. (F)
 BUD RADIO, INC., 2118 E. 55th St., Cleveland 3, Ohio (B)
 BUTLER MFG. CO., INC., B. B., 3150 Randolph St., Bellwood, Ill. (B)
 CALIFORNIA CABINET CO., 522 S. San Pedro St., Los Angeles, Calif. (B)
 CARBONNEAU INDUSTRIES, 100 Lexington S. W., Grand Rapids 4, Mich. (D, F)
 CLEVELAND ELECTRONICS, INC., 1974 E. 61st St., Cleveland 3, Ohio. (C, D, F, G)
 COMPONENTS CORP., 106 Main St., Denville, N. J. (H)
 CONSOLIDATED RADIO PRODUCTS CO., 350 W. Erie St., Chicago 10, Ill. (D, E, F, G)
 D & R, LTD., 402 E. Gutierrez St., Santa Barbara, Calif. (A)
 DICTOGRAPH PRODUCTS INC., 95-25 149th St., Jamaica 35, N. Y. (B)
 DUKANE CORP., St. Charles, Ill. (C)
 ELECTRON ENTERPRISES, 6917 W. Stanley Ave., Berwyn, Ill. (B)
 ELECTRO-VOICE, INC., Buchanan, Mich. (A, B, C, E, F, G)
 GENERAL APPARATUS CO., 344 E. 32nd St., New York 16, N. Y. (A)
 GENERAL ELECTRIC CO., Electronics Park, Syracuse, N. Y. (B, D, F, G)
 G & H WOOD PRODUCTS CO., INC., 99 N. 11th St., Brooklyn 11, N. Y. (A, C)
 GROSSMAN RADIO & ELECTRIC CO., 81 Spring St., New York, N. Y. (C, D, F, G)
 HARTLEY PRODUCTS CO., 521 E. 162nd St., New York 51, N. Y. (B, F)
 INTERNATIONAL RADIO & ELECTRONICS CORP., Elkhart, Ind. (B)
 JAMES INSTRUMENT LABORATORY, 9110 S. 52nd Ct., Oak Lawn, Ill. (B)
 JENSEN MFG. CO., 6601 S. Laramie St., Chicago 3, Ill. (A, B, C, D, E, F, G)
 KARLSON ASSOCIATES, INC., 1483 Cone Island Ave., Brooklyn 30, N. Y. (B)
 KINGDON PRODUCTS, LTD., 23 Park Pl., New York 7, N. Y. (A, D, F, G, H)
 KLIPSCH & ASSOCIATES, P. O. Box 64, Hope, Ark. (A, B, C, F, G)
 LANSING SOUND, INC., JAMES B., 2439 Fletcher Dr., Los Angeles 39, Calif. (A, B, C, E, F, G)
 LOWELL MFG. CO., 3030 Laeclde Station Rd., St. Louis, Mo. (B)
 MANFREDI WOOD PRODUCTS CO., 226 New York Ave., Huntington, N. Y. (B)
 MARKELL ASSOCIATES, JEFF, 764 Sixth Ave., New York, N. Y. (B)
 MATTISON TELEVISION AND RADIO CORP., 10 W. 181 St., New York 53, N. Y. (B)
 MENDELS, INC., R. I., 41 E. 42nd St., New York 17, N. Y. (F)
 ORTHO FILTER CORP., 196 Albion Ave., Paterson, N. J. (A)
 OXFORD ELECTRIC CORP., 3911 S. Michigan Ave., Chicago 15, Ill. (D, F, G)
 PALMER, INC., M. V., 4002 Fruit Valley Rd., Vancouver, Wash. (B)
 PERMOFLUX CORP., DISTRIBUTOR DIV., 2835 N. Kedzie Ave., Chicago 18, Ill. (A, B, C, D, F, G)
 PILOT RADIO CORP., 37-06 36th St., Long Island City 1, N. Y. (B)
 QUAM-NICHOLS CO., 234 E. Marquette Rd., Chicago 37, Ill. (A, C, D, F, G)
 RACON ELECTRIC CO., INC., 1261 Broadway, New York, N. Y. (A, B, C, D, E, F, G)
 RADIO CORP. OF AMERICA (ENGINEERING PRODUCTS DIV.), Front and Cooper Sts., Camden, N. J. (A, B, D, E, F, G)
 RIVER EDGE INDUSTRIES, 5 River Edge Rd., River Edge, N. J. (B)
 RJ AUDIO PRODUCTS, 80 Shore Rd., Port Washington, L. I., N. Y. (B)
 ROCKBAR CORP. (AGENTS FOR COLLARO, LTD., & GOODMANS INDUSTRIES), 215 E. 37th St., New York 18, N. Y. (A, C, D, F, G)
 SHERWOOD ELECTRONIC LABORATORIES, INC., 2802 W. Cullom Ave., Chicago 18, Ill. (A, B, C, D, G)
 SIGHTMASTER CORP., 111 Cedar St., New Rochelle, N. Y. (A, B, D, F, G)
 SONOTONE CORP., Elmsford, N. Y. (D, F, G)
 SOUTHWESTERN INDUSTRIAL ELECTRONICS CO., 2831 Post Oak Rd., P. O. Box 13058, Houston 19, Texas (A)
 STANDARD WOOD PRODUCTS CORP., 47 W. 63rd St., New York 23, N. Y. (B)
 STEPHENS MANUFACTURING CORP., 8538 Warner Dr., Culver City, Calif. (A, B, C, D, E, F, G)
 STROMBERG-CARLSON CO., SOUND DIV., 1225 Clifford Ave., Rochester, N. Y. (B, F)
 SUMMIT ELECTRONICS, INC., 7 Industrial Pl., Summit, N. J. (A)
 TANNAY (AMERICA), LTD., 38 Pearl St., New York 4, N. Y. (A, B, E, F, G)
 TEXAS INSTRUMENTS, COMPONENTS DIV., 6000 Lemmon Ave., Dallas 9, Texas (A)
 ULTRA HI-FI CO., 709 Sip St., Union City, N. J. (B)
 UNITED AUDIO PRODUCTS DIV., UNITED OPT. MFG. CORP., 202-4 E. 19th St., New York 3, N. Y. (C, D, F, G)
 UNIVERSITY LOUDSPEAKERS, INC., 80 S. Kensico Ave., White Plains, N. Y. (A, B, C, E, F, G)
 UTAH RADIO PRODUCTS CO., INC., 1123 E. Franklin St., Huntington, Ind. (B, C, D, E, F, G)
 WHITE, INC., STAN, 726 S. La Salle St., Chicago, Ill. (B)
 WILCOX-GAY CORP., Charlotte, Mich. (B)
 WRIGHT, INC., 2283 University Ave., St. Paul 14, Minn. (B, F)

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The Dynakit uses a new bug-free circuit, designed by David Hafler. Complete reproducibility of operating characteristics is guaranteed by the use of a factory-wired printed circuit board. The Dynakit comes complete with all components including the super-fidelity Dynaco A-430 transformer.

Specifications:

Power Output: 50 watts continuous rating, 100 watts peak. Distortion: under 1% at 50 watts, less than 1% harmonic distortion at any frequency 20 cps to 20 kc within 1 db of maximum. Response: Plus or minus .5 db 6 cps to 60 kc. Plus or minus .1 db 20 cps to 20 kc. Square Wave Response: Essentially undistorted 20 cps to 20 kc. Sensitivity: 1.5 volts in for 50 watts out. Damping Factor: 15. Output Impedances: 8 and 16ohms. Tubes: 6CA7/EL-34 (2) (6550's can also be used) 6AN8, 5U4GB. Size: 9" x 9" 6 5/8" high.

69.75
net

Dynaco Output Transformers



Featuring para-coupled windings, a new design principle (patents pending). These transformers use advanced pulse techniques to insure superior square wave performance and undistorted reproduction of transients. Dynaco transformers handle full rated power over the entire audio spectrum from 20 cps to 20 kc, without sharp rise in distortion at the ends of the band which characterizes most transformers. Conservatively rated and guaranteed to handle double nominal power from 30 cps to 15 kc without loss of performance capabilities.

Specifications:

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MODELS

A-410	10 watts	6V6, EL-84	14.95
A-420	25 watts	KT-66, 5881, EL-34	19.95
A-430	50 watts	6550, EL-34 (6CA7)	29.95
A-440	100 watts	6550	39.95

(all with tapped primaries except A-440 which has tertiary for screen or cathode feedback)

Additional data on Dynakit and Dynaco components available on request including circuit data for modernization of Williamson-type amplifiers to 50 watts of output and other applications of Dynaco transformers.

DYNA COMPANY

5142 Master St.
Phila. 31, Pa.

WRIGHT ZIMMERMAN, INC., New Brighton, St. Paul
12, Minn. (A, B, C, D, F, G)

TAPE RECORDERS AND EQUIPMENT

- A Demagnetizers
 - B Heads
 - C Players
 - D Recorders
 - E Splicers
 - F Tape
 - G Tape (recorded)
 - H Tape (test)
- (Italics apply only to trade name of recorded tapes.)

- ALONGE PRODUCTS, INC., 163 West 23rd St., New York, N. Y. (E)
- AMPLIFIER CORP. OF AMERICA, 398 Broadway, New York 13, N. Y. (C, D)
- AMPRO CORP., 2835 N. Western Ave., Chicago 18, Ill. (D)
- AUDIO DEVICES, INC., 444 Madison Ave., New York 22, N. Y. (A, F)
- AUDIO-MASTER CORP. (MASTERTAPE), 17 E. 45th St., New York 17, N. Y. (C, D, F, G)
- A-V TAPE LIBRARIES, INC. (A-V RECORDED TAPES, NATIONAL MUSITIME), 730 Fifth Ave., New York 19, N. Y. (G)
- BELL & HOWELL, 7100 McCormick Rd., Chicago 45, Ill. (D)
- BELL SOUND SYSTEMS, INC., 555 Marion Rd., Columbus 7, Ohio (D)
- BERLANT-CONCERTONE (AUDIO DIV., AMERICAN ELECTRONIC, INC.), 4917 W. Jefferson, Los Angeles 16, Calif. (D)
- BROADCAST EQUIPMENT SPECIALTIES CORP., P. O. Box 149, Beacon, N. Y. (D)
- BRUSH ELECTRONICS CO., 3405 Perkins Ave., Cleveland 14, Ohio (B)
- BURDETT SOUND & RECORDING CO. (BURDETT), 3619 Henderson Blvd., Tampa 9, Fla. (G)
- CINEMA ENG. CO., DIV. AEROVOX CORP., 1100 Chestnut St., Burbank, Calif. (A)
- COLUMBIA RECORDS, 799 Seventh Ave., New York, N. Y. (D)
- CONCERTAPES, INC., 400 Jackson Ave., Glencoe, Ill. (C)
- COUSINO, INC., 2325 Madison Ave., Toledo 2, Ohio (E)
- CRESCENT INDUSTRIES, INC., 5900 W. Touhy Ave., Chicago 31, Ill. (D)
- DAYSTROM ELECTRIC CORP., 837 Main St., Poughkeepsie, N. Y. (D)
- DEJUR-AMSCO CORPORATION, 45-01 Northern Blvd., Long Island City, N. Y. (D)
- DUBBINGS SALES CORP., 41-10 45th St., Long Island City 4, N. Y. (G, H)
- DUBROW DEVELOPMENT CO., 236 Penn St., Burlington, N. J. (D)
- DUKANE CORP., St. Charles, Ill. (D)
- DYNAMU MAGNETRONICS CORP., DIV. OF THE MAICO CO., INC., 21 N. 3rd St., Minneapolis 1, Minn. (B, C)
- EICOR, INC., 1501 W. Congress St., Chicago 7, Ill. (D)
- ELECTROSONIC SPECIALTIES (FIDELIVOX), 7230 Clinton Rd., Upper Darby, Pa. (G)
- ELECTRO SOUND CORP., 115 Halleck St., Boston 20, Mass. (D, G)
- EMC RECORDINGS CORP., 806 E. Seventh St., St. Paul 5, Minn. (C, G)
- ERICSSON CORP., THE, 100 Park Ave., New York 17, N. Y. (D)
- ESS TEE DISTRIBUTORS, INC. (TAPE TOONS, SKATIN' TOONS), Box 397, Smithtown, N. Y. (G)
- FAIRCHILD RECORDING EQUIPMENT CO., White-stone 57, N. Y. (D)
- FEDERAL MFG. & ENG. CORP., 211 Steuben St., Brooklyn 5, N. Y. (D)
- FENTON CO., 15 Moore St., New York, N. Y. (B, D)
- GENERAL INDUSTRIES CO., THE, Olive and Taylor Sts., Elyria, Ohio (D)
- GORDON ENTERPRISES, 5362 N. Cahuenga Blvd., N. Hollywood, Calif. (C, D)
- HARTLEY PRODUCTS CO., 521 E. 162nd St., New York 51, N. Y. (D, F)
- INTERNATIONAL PACIFIC RECORDING CORP. (ALPHATAPE OMEGATAPE, JAZZTAPE), 858 N. Vine St., Hollywood 38, Calif. (A, G, H)
- INTERNATIONAL RADIO & ELECTRONICS CORP., Elkhart, Ind. (D)
- INTERNATIONAL SCIENTIFIC INDUSTRIES CORP., 3101 E. 42nd St., Minneapolis 6, Minn. (C, D)
- JAMES INSTRUMENT LAB., 9110 S. 52nd Ct., Oak Lawn, Ill. (D)
- LIPPS ENGINEERING, EDWIN A., 5485 W. Wash-ington Blvd., Los Angeles 16, Calif. (B)
- LIVINGSTON ELECTRONIC CORP. (CONNOISSEUR, ESOTERIC, HACK SWAIN MUSIKOV, OCEANIC, RIVERSIDE, SWAIN-A-FONIC), Livingston, N. J. (C, G)
- MAGNASYNC MFG. CO., LTD., 5623 Satsuma Ave., North Hollywood, Calif. (B, D)
- MAGNACORD INC., 1101 S. Kilbourn, Chicago, Ill. (B, C, D)
- MAGNE-TRONICS, INC. (MUSIC TAPES), 122 E. 42nd St., New York 17, N. Y. (G)
- MICHIGAN ELECTRONICS, INC., 854 N. Rockwell St., Chicago 22, Ill. (C, D, F)
- MILLER ELECTRONICS (MILLER), 2840 Naomi, Burbank, Calif. (D, G)
- MINNESOTA MINING AND MANUFACTURING CO., 900 Fauquier Ave., St. Paul 6, Minn. (F)
- MITCHELL MFG. CO., 2825 Clybourn Ave., Chicago 14, Ill. (D)
- MOHAWK BUSINESS MACHINES CORP., 944 Halsey St., Brooklyn 33, N. Y. (C, D)
- MORRISON RECORDING LABS. (MORRISONTAPE), 120 S. Batavia Ave., Batavia, Ill. (G)
- MUSIC TAPE SOCIETY, Box 661, Detroit 6, Mich. (G)
- MUSITAPES, INC. (MUSITAPE, TEMPOTAPE), 8540 Sunset Blvd., Hollywood 46, Calif. (G)
- ORRADIO INDUSTRIES, INC., Opelika, Ala. (F)
- PENTRON CORP., THE, 777 S. Tripp, Chicago 24, Ill. (A, B, C, D, G)
- PERMO, INC., 6415 N. Ravenswood Ave., Chicago 28, Ill. (F)
- PHONOTAPES, INC. (FOLKWAYS), 853 Ninth Ave., New York 19, N. Y. (G)
- PRESS WIRELESS LABORATORIES, INC., 25 Prospect Pl., West Newton 85, Mass. (D)
- PRESTO RECORDING CORP., P. O. Box 500, Paramus, N. J. (A, B, C, D)
- RADIO CORP. OF AMERICA (RCA VICTOR), 630 Fifth Ave., New York 20, N. Y. (G)
- RADIO CORP. OF AMERICA (ENGINEERING PRO-DUCTS DIV.), Front & Cooper Sts., Camden, N. J. (D, F)
- RANGERTONE, INC., 73 Winthrop St., Newark, N. J. (B, D, H)

- RECORDED TAPE OF THE MONTH CLUB, Box 195, Radio City P. O., N. Y. (G)
- REEVES SOUND-CRAFT CORP., 10 E. 52nd St., New York 22, N. Y. (F)
- REVERE CAMERA CO., 320 E. 21st St., Chicago 16, Ill. (D)
- ROBINS INDUSTRIES CORP., 41-08 Bell Blvd., Bay-side 61, N. Y. (E, H)
- SHURE BROTHERS, INC., 225 W. Huron St., Chi-cago, Ill. (B)
- STANCIL-HOFFMAN CORP., 921 N. Highland Ave., Hollywood 38, Calif. (D)
- SUMMIT ELECTRONICS, INC., 7 Industrial Pl., Sum-mit, N. J. (A, D)
- TECHNICAL TAPE CORP., ELECTRO-MAGNETIC PRODUCTS DIV., Morris Heights 53, N. Y. (F)
- TELESTRO INDUSTRIES CO., 35-16 37th St., Long Island City 1, N. Y. (C, D)
- TELESTROSONIC CORP., 35-18 37th St., Long Island City 1, N. Y. (C, D)
- VIKING OF MINNEAPOLIS, 3520 E. 43rd St., Min-neapolis 6, Minn. (C, D)
- WARREN CORP., J. C., 21 Hanse Ave., Freeport, N. Y. (D)
- WEB MUSIC PUBLISHING CO. (WEB), 149 W. 48th St., New York 36, N. Y. (G)
- WEBSTER-CHICAGO CORP. (WEBCOR), 5610 W. Bloomingdale Ave., Chicago, Ill. (D, F, G)
- WEBSTER ELECTRIC CO., 1900 Clark St., Racine Wis. (D)
- WILCOX-GAY CORP., Charlotte Mich. (C, D)

TRANSFORMERS

- A Input
- B Output

- ACRO PRODUCTS CO., 369 Shurs Lane, Philadelphia 28, Pa. (A, B)
- AIRCRAFT TRANSFORMER CORP., Corner West & Willow Ave., Long Branch, N. J. (A, B)
- BUTLER MFG. CO. INC., B. B., 3150 Randolph St., Bellwood, Ill. (B)
- CHICAGO STANDARD TRANSFORMER CORP., 3850 Elston, Chicago 18, Ill. (A, B)
- CONSOLIDATED RADIO PRODUCTS CO., 350 W. Erie St., Chicago 10, Ill. (A, B)
- DIETZ DESIGN & MFG. CO., Grandview, Md. (A, B)
- ELECTRO-SONIC LABS., 35-54 36th St., Long Island City, N. Y. (A, B)
- FAIRCHILD RECORDING EQUIPMENT CO., White-stone 57, N. Y. (A)
- FEMCO, INC., Box 32, Irwin, Penna. (A, B)
- FOREST ELECTRIC CO., THE, 7216 Circle Ave., Forest Park, Ill. (A, B)
- FRED TRANSFORMER CO., INC., 1718 Weirfield St., Brooklyn 27, N. Y. (A, B)
- GENERAL RADIO CO., 275 Massachusetts Ave., Cambridge 39, Mass. (B)
- GROSSMAN RADIO & ELECTRIC CO., 81 Spring St., New York, N. Y. (B)
- KENYON TRANSFORMER CO., 840 Barry St., New York 59, N. Y. (A, B)
- LANGEVIN MFG. CORP., 47-37 Austell Pl., Long Island City 1, N. Y. (A, B)
- MCINTOSH LAB., INC., 320 Water St., Binghamton, N. Y. (A)
- MERIT COIL & TRANSFORMER CORP., 4427 N. Clark St., Chicago 40, Ill. (B)
- MICROTRAN CO., DIV. CREST LABORATORIES, INC., 84-11 Blvd., Rockaway Beach 93, N. Y. (A, B)
- MORROW RADIO MFG. CO., 2794 Market St., Sa-lem, Ore. (A, B)
- NYT ELECTRONICS, INC., 2979 N. Ontario St., Bur-bank, Calif. (A, B)
- ORTHO FILTER CORP., 196 Albion Ave., Paterson, N. J. (A, B)
- PEERLESS ELECTRICAL PRODUCTS DIV., ALTEC LANSING CORP., 9356 Santa Monica Blvd., Beve-ly Hills, Calif. (A, B)
- QUAM-NICHOLS CO., 234 E. Marquette Rd., Chicago 37, Ill. (B)
- RADIO CORP. OF AMERICA (ENGINEERING PRO-DUCTS DIV.), Front & Cooper Sts., Camden, N. J. (A)
- RADIO MUSIC CORP., 84 S. Water St., Port Ches-ter, N. Y. (A, B)
- SKY SWEEPER, INC., McHenry, Ill. (A, B)
- SNC MANUFACTURING CO., INC., P. O. Box 277, Oshkosh, Wis. (A, B)
- SONEX, INC., 245 Sansom St., Upper Darby, Penna. (B)
- SOUTHWESTERN INDUSTRIAL ELECTRONICS CO., 2831 Post Oak Rd., P. O. Box 13058, Houston 19, Texas (A, B)
- STANDARD ELECTRICAL PRODUCTS CO., 2240 E. Third St., Dayton, Ohio (A, B)
- TELEX CO., THE, 46 Lakeview Ave., Yonkers, N. Y. (A)
- TEXAS INSTRUMENTS, COMPONENTS DIV., 6000 Lemmon Ave., Dallas 9, Texas (A, B)
- THORPSON-MEISSNER MANUFACTURING DIV., 70 Belmont Ave., Mt. Carmel, Ill. (A, B)
- TRIAO TRANSFORMER CORP., 4055 Redwood Ave., Venice, Calif. (A, B)
- UNITED TRANSFORMER CO., 150 Varick St., New York 13, N. Y. (A, B)
- WHEELER INSULATED WIRE CO., INC., 150 E. Aurora St., Waterbury 29, Conn. (A, B)
- WRIGHT, INC., 2233 University Ave., St. Paul 14, Minn. (B)

TUNERS

- A AM-FM
- B FM

- ALTEC LANSING CORP., 9356 Santa Monica Blvd., Beve-ly Hills, Calif. (A)
- APPROVED ELECTRONIC INSTRUMENT CORP., 928 Broadway, New York 10, N. Y. (A, B)
- BELL SOUND SYSTEMS, INC., 555 Marion Rd., Co-lumbus 7, Ohio (A, B)
- BIGG OF CALIFORNIA (BIGG RADIO CO.), 2506 W. Washington Blvd., Los Angeles, Calif. (A, B)
- BOGEN CO., INC., DAVID, 29 Ninth Ave., New York 14, N. Y. (B)
- BRAINARD ELECTRONICS, 8586 Santa Monica Blvd., Los Angeles 64, Calif. (A, B)
- BROWNING LABORATORIES, INC., 750 Main St., Win-chester, Mass. (A, B)
- COLLINS AUDIO PRODUCTS CO., INC., Route 29, Mount-ainside, N. J. (A, B) (Mailing Address: P. O. Box 368, Westfield, N. J.)
- COLUMBIA RECORDS, 799 Seventh Ave., New York, N. Y. (A)
- DeWALD RADIO MANUFACTURING CORP., 35-15 37th Ave., Long Island City 1, N. Y. (A)
- DICTOGRAPH PRODUCTS INC., 95-25 149th St., Jam-nica 35, N. Y. (A, B)
- ELECTON ENTERPRISES, 6917 W. Stanley Ave., Berwyn, Ill. (A)

ELECTRO-VOICE, INC., Buchanan, Mich. (A, B)
FISHER RADIO CORP., 21-21 44th Dr., Long Island City 1, N. Y. (A, B)
GOTHAM TELEVISION CORP., 123 W. 64th St., New York, N. Y. (A)
GRANCO PRODUCTS, INC., 36-07 20th Ave., Long Island City 5, N. Y. (B)
GROMMES DIV. OF PRECISION ELECTRONICS, INC., 9101 King Ave., Franklin Park, Ill. (A)
HARMAN-KARDON INC., 620 Main St., Westbury, L. I., N. Y. (A, B)
HEATH CO., 305 Territorial Rd., Benton Harbor, Mich. (B)
INTERNATIONAL RESEARCH ASSOCIATES, 2221 Warwick Ave., Santa Monica, Calif. (A)
NATIONAL COMPANY, INC., 61 Sherman St., Malden 48, Mass. (A)
NEWCOMB AUDIO PRODUCTS CO., 6824 Lexington Ave., Hollywood 38, Calif. (A)
PILOT RADIO CORP., 37-06 36th St., Long Island City 1, N. Y. (A, B)
RADIO CORP. OF AMERICA (ENGINEERING PRODUCTS DIV.), Front & Cooper Sts., Camden, N. J. (A)
RADIO CRAFTSMEN, INC., THE, 4401 N. Ravenswood Ave., Chicago 40, Ill. (A, B)
RADIO ENGINEERING LABS., INC., 36-40 37th St., Long Island City, N. Y. (B)
RAULAND-BORG CORP., 3515 W. Addison St., Chicago 18, Ill. (A)
REGENCY DIVISION, I. D. E. A., INC., 7900 Pendleton Pike, Indianapolis 26, Ind. (A, B)
SARGENT-RAYMENT CO., THE, 1401 Middle Harbor Rd., Oakland 20, Calif. (A)
SCOTT, INC., HERMON HOSMER, 385 Putnam Ave., Cambridge 39, Mass. (A, B)
SHERWOOD ELECTRONIC LABS., INC., 2802 W. Cullom Ave., Chicago 18, Ill. (A)
SIGHTMASTER CORP., 111 Cedar St., New Rochelle, N. Y. (A)
STROMBERG-CARLSON CO., SOUND DIV., 1226 Clifford Ave., Rochester, N. Y. (A)
TECH-MASTER CORP., 75 Front St., Brooklyn, N. Y. (B)
THORDARSON-MEISSNER MANUFACTURING DIV., 7th & Bellmont Ave., Mt. Carmel, Ill. (A, B)
WELLS-GARDNER & CO., 2701 N. Kildare Ave., Chicago 39, Ill. (A)
WILCOX-GAY CORP., Charlotte, Mich. (A, B)

WIRE RECORDERS AND EQUIPMENT

A Heads
 B Recorders
 C Wire
CRESCENT INDUSTRIES, INC., 5900 W. Touhy Ave., Chicago 31, Ill. (A, B)
ELECTRON ENTERPRISES, 6917 W. Stanley Ave., Berwyn, Ill. (B)
GENERAL CEMENT MFG. CO., 919 Taylor Ave., Rockford, Ill. (C)
GORDON ENTERPRISES, 5362 N. Cahuenga Blvd., N. Hollywood, Calif. (A, B, C)
INTERSTATE PHOTO SUPPLY CORP., 17 W. 17th St., New York 11, N. Y. (B)
MAGNASYNC MFG. CO., LTD., 5523 Satsuma Ave., North Hollywood, Calif. (A)
PERMO, INC., 6415 N. Ravenswood Ave., Chicago 28, Ill. (C)
SHURE BROTHERS, INC., 225 W. Huron St., Chicago, Ill. (A)
TELECTROSONIC CORP., 35-18 37th St., Long Island City 1, N. Y. (B)
WEBSTER-CHICAGO CORP. (WEBCOR), 5610 W. Bloomingdale Ave., Chicago, Ill. (B, C)

EDITOR'S NOTE: The above listing has been confined solely to manufacturers, both in this country and abroad, who replied to our questionnaire. Foreign suppliers are listed under their U. S. distributors.

We want to call the reader's attention to the fact that one major group is not included, i.e., the electronic parts distributor. Many of these companies market various audio components both in assembled and kit form under their own brand names. We would like to suggest that for further information on these components, you check your local parts distributor or consult the ads in this magazine.

SPEAKER SERVICE HINT

By WILLARD S. WEISS

HERE is one way to locate those elusive speaker troubles.

Occasionally, a loudspeaker may become noisy and rattle due to such things as a loose eyelet where the voice coil wires are attached to the cone, "unstuck" glue, etc.

A quick way to find such trouble is to energize the speaker with some frequency at which the noise can be heard. Then, by using a piece of rubber tubing held to your ear with the other end near the cone, the trouble can usually be isolated.

A medical stethoscope, if available, can be used for the same purpose as the tubing.



FOR THE AUDIO GOURMET

We have prepared a tempting new menu of truly incomparable High Fidelity Creations

Stan White PRESENTS

The following entrees to thrill the musical palates of audiophiles. Especially created for those who think they have heard the best.

ooo...

THE Mach I® A new aerodynamic Ring Speaker.
 The first truly new speaker development in twenty years.
 featuring

- 15 Inch Voice Coil • Piston Oriented Speaker
- Radically New Magnetic Material • Magnet Weight 30 Lbs.

A 15 Inch Speaker Only 3 Inches Deep.
 New Shape . . . New Materials . . . Ringside Fidelity
 Designed with the aid of wind tunnel data at Mach 1.
 To be released in March; price to be announced.

ohh...

THE Beta-Tron* A Motional Feed-Back Tweeter
 The new tweeter that actually reaches astronomical heights

- New Massless Tweeter with No Intermodulation Distortion.
- New High Frequency response passes 50,000 cycle square waves.

*So named because "Beta" is a feed-back factor and "Beta" is an atomic particle which actuates this device.
 Net Price, complete with amplifier.....99.50

umm

Stan White Exponential Horn Speakers
 featuring

the MIRACLE OF MULTI-FLARE available in Blonde Korina, Walnut, Mahogany and Ebony Cabinets.

The Opus I. Size: 30x20x15 inches.....99.50
 The perfect speaker for small rooms and apartments.

The Esquire. Size: 36x24x16 inches.....199.50
 Deluxe three way horn system of unusual quality.

And Le Sabre—69.50 . . . The Hi-Fi—339.50 . . . The Millennium—1000.00. Each one the outstanding speaker in its price class.

See your exclusive Stan White Distributor. Some of the above releases are so new there may be a slight delay in delivery. But . . . you'll be glad you waited!

Stan White INC.
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 Dept. N-3, 725 S. La Salle Street, Chicago 5, Illinois

for the finest in sound

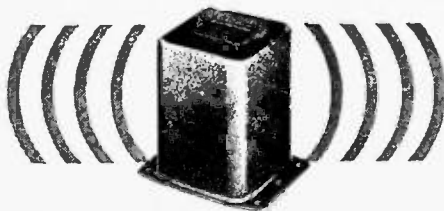


Acrosound

ULTRA-LINEAR 

OUTPUT TRANSFORMERS

- * The quality of Acrosound transformers is accepted throughout the world as being unequalled.
- * The finest amplifier quality comes from the optimum integration of the best output transformers and the best circuits.
- * Acro-developed Ultra-Linear circuits are likewise accepted as the best, and the combination of these factors make the finest sounding and finest performing amplifiers obtainable.
- * Excellence of Acrosound transformers and circuits is illustrated by specifications of the TO-330 model designed for high-power Ultra-Linear amplifiers using the new 6550 tubes.
Frequency response ± 1 DB, 10CPS to 100 KC
Power 50 watts 20 CPS—30 KC
rating— 100 watts 30 CPS—20 KC
Impedance—3300 ohms plate to plate to 4, 8, 16 ohms. Primary taps located at 40% for best U.L. performance.
Price: \$39.75 at leading distributors
- * Catalog available upon request
- * Distinctive features protected by patent!



ACRO PRODUCTS COMPANY
369 Shurs Lane, Philadelphia 28, Pa.

Musician Looks at Hi-Fi

(Continued from page 117)

ing or rehearsing could take place in the same hall.

Playback

Listening to the results of a recording session frequently proves a bit disappointing. The sound is often harsher and more edgy than the quality of sound perceived while playing. On occasion, an individual instrument or a certain passage clearly heard on stage seems to assume a different place within the over-all recorded balance. Certainly the type of playback equipment used and the lack of the final editing and recording refinements at that moment, play an important part. Since all musical and some technical improvements during a session are based on the playback, the more realism the playback can offer, the higher the degree of self-criticism which can be supplied by the player.

Home Reproduction

After the recording has been optimized and edited, what can we finally expect to hear at home, on a fine hi-fi system? We must first of all take into consideration the following facts; the most sensitive aural response lies within the range of 500-8000 cps (frequencies above 15,000 cps being inaudible to most people.) There is considerable variation in that people hear under the same conditions, particularly regarding music, as well as some deterioration in hearing with advancing age.

There are also distortions in the ear at low frequencies, and we must also allow for masking effects. Masking effects occur when one of two tones of different frequency is increased in volume and thereby renders the other tone inaudible. Generally masking comes about when frequencies are fairly close together. (2-3 octaves). Furthermore generalizations are difficult to arrive at, because of the differences in individual record pressings and listening-room acoustics.

In all the following listening tests these high fidelity components were used by the courtesy of *Custom Classics*, Cleveland, Ohio: *Weathers* pick-up arm; *Rek-O-Kut* T 12 turntable; *McIntosh* C8 preamplifier; *McIntosh* Mc 30 amplifier; and two *Bozak* B305 speakers and enclosures.

The listening room, 14 feet wide, 16 feet long, and 12 feet high, is fairly well damped, with carpeted floor. 64 square feet of the wall area is treated with acoustical tile. The remainder of the walls and ceiling is hard.

In comparing several of the leading labels it was observed that they could be distinguished by certain individual traits. None of them was entirely free from crackles and slight hissing noises. Even the factory-sealed records showed occasional scratches. In listening for accurate reproduction of the various

sections of the orchestra and comparing them with live sound, the string sections, especially violins, were the least natural. We discovered the speakers to be the most essential factor for improving or ruining the potential of a given recording.

Strong hi-fi responses seem to be stimulating to some people, while others prefer the more mellow tone quality and volume which one might hear in the first row or amidst the orchestra itself. The author's preference would be an objective position such as the center of a hall or the dress circle. Hi-fi addicts sometimes delight in such slogans as "sharp brasses, edgy strings, enormous percussion triumphs." These impressions would often prove musically incorrect when checked against the score, however effective these distortions.

Loudness

In order to test the effect of loudness on reproduction we used the clarinet solo at the opening of the 3rd movement of Rachmaninoff's "Symphony No. 2" (*Capitol*). The test was made without loudness compensation. The set was adjusted to correspond to a level which might be heard in the center of a concert hall. A reduction of 10 db showed no considerable change in the orchestral balance. A reduction of 20 db, however, distorted the musical balance beyond reason. The increase of 10 db above the original level caused considerable surface noise and the relation of the solo clarinet to its accompaniment became unbalanced.

Frequency Response

In trying to test the effects of a flat frequency response versus bass and treble boost, we had to favor the flat response, which offered greater musical clarity. Although more apparent "presence" was achieved with the treble boost, the resulting surface noise seemed to cancel any advantage gained. While bass boost was not as objectionable, the texture of the music was nevertheless obscured.

The author wishes to emphasize that these statements represent a personal reaction which, due to room acoustics and equipment variations, may call for individual modifications.

Discs Versus Tape

Discussing such controversial matters as the difference between tape and discs or the importance of overtones, the musician is caught in a web of contradictory statements by the experts. The absence of stylus noises, rumbles, changing of sides, and storage problems and the possibility of transferring discs to tape, all seem to be attractive features serving the cause of music well. The musician contends that it is the overtone structure which characterizes the instruments and supplies timbre and tone color. Tape suffers from some limitations in this regard. In comparing the frequency range of the fundamentals with that of the entire spectrum, it is clear that

the overtones extend the range by two or more octaves.

It would be difficult to deny the progress and accomplishments of the record industry. Although the musician realizes the need for further mechanical improvements, there are some other aspects connected with hi-fi recording which are most valuable to him. To name only a few, there is the advantage of repeated hearings of complex works, hi-fi productions of new music, American or otherwise, under the guidance or directly by the composer, educational series such as the various "appreciation" records, authentic recordings of world-famous organs and ancient instruments, and recent electronic research into the field of musical creation.

In the field of pop music hi-fi has actually created new styles. Here the engineer has taken over from the musician by adding fancy sound flourishes such as bird calls, the barking of a dog, and above all the echo chamber. Well known is the trick of a singer singing a duet with himself, which has as its counterpart in the classical recording literature in the Bach "Double Concerto for Two Violins," with Heifetz playing both parts. Although many of these effects are being overdone, we hear some interesting and sophisticated orchestrations on popular recordings nowadays. By the use of the oboe, the French horn, and even the harpsichord there has been somewhat of a reconciliation between the long-haired and the short-haired musician.

All this, however, has shifted the responsibility for maintaining our cultural level as far as music is concerned from a few individuals to the record buying and listening public. There is a danger in accepting a musical performance by repeated listening to the same recording of it, or by mechanical manipulations of one's hi-fi set. If we are to have participating and discriminating listeners, their judgment must be based on as much listening experience with actual live music as possible. If the highs and lows are exaggerated or overemphasized the sound is robbed of its natural musical quality. The hunt for the elimination of rumbles, scratches, and rattles often strips the music of its freshness and spirit. These technical alterations diminish the high fidelity of music. If hi-fi is not to become an end in itself, its interdependence with live music, the composer, the critic, performer, and educator must be realized. Frequently the performance of a new composition will be followed up with a recording of it. Just as often the reverse is the case, when a musician will perform a piece of music, which was initially heard on records. The reactions of a reputable critic to a live performance or a recording obviously have an important influence on the entire musical scene. Particularly from an economic point of view this interrelation is of utmost consequence.

-30-

Choosing a Phono Pickup (Continued from page 110)

correct normal relationship.

This means that, on the basis of our theoretical constant velocity recording, the record characteristic is as shown in Fig. 4A while the playback characteristic is complementary to it as shown in Fig. 4B. In practice there is a variety of equalization characteristics and those shown here are just typical of the general scheme.

There are other types of recording heads and pickups that employ a constant-amplitude principle, that is, the

output is proportional at all frequencies to the amplitude of stylus movement. In Fig. 4B the result of playing a constant-velocity type disc through a constant-amplitude type pickup is shown dotted. It will be seen that not too much equalization will be necessary to bring this to a level response, because the equalization required will be the difference between the solid curve and the dotted line. This equalization is shown at Fig. 4C.

Popular types of pickups, using the amplitude principle, are the piezoelectric and the electrostatic. The piezoelectric types fall into two groups, the crystal and ceramic. Both employ a piezoelectric principle in which move-



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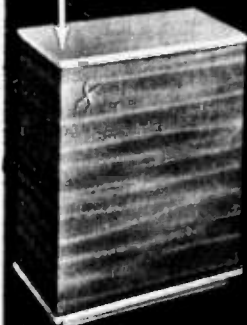
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ment of the stylus sets up a voltage due to distortion of the piezoelectric material.

In the crystal type of cartridge the piezoelectric material is a Rochelle salt crystal. These were very popular in the thirties and gave reproduction that was quite pleasing compared with the best competitive forms at that time. However a crystal structure is always apt to have mechanical resonances of rather high "Q", which means that any peaks or valleys the mechanical system of the pickup introduces are apt to be rather severe.

They can be damped out to some extent but this is rather difficult. In modern design the construction of the pickup is such as to remove these resonances beyond the audio range.

The crystal type of cartridge has the disadvantage that Rochelle salt is extremely susceptible to high temperatures and moisture. Consequently they can easily deteriorate, if subjected to high temperatures or humid conditions. The ceramic type of cartridge employs a variety of ceramic in which the molecules have been polarized during manufacture under a strong electrostatic field. This produces a piezoelectric material which is not subject to temperature and humidity in the same way that Rochelle salt is.

With this basic type of element there is a limit to the amount the crystal or ceramic element can be distorted before its elastic limit is reached. Beyond this limit, further stress permanently distorts the element, until a stress in the opposite direction restores it to its original form. So, as the elastic limit of the material is approached, the relation between movement of the stylus and electrical output will cease to be linear and it will begin to introduce distortion.

The final choice of pickup cartridge to be discussed is the electrostatic type. It does not employ the simple electrostatic principle of producing a voltage by varying the capacitance of the plates with a constant charge. In this cartridge the capacitance between the plates, which the movement of the stylus varies, is part of an oscillatory circuit, operating as an r.f. generator. Variation of the stylus position modulates the frequency of r.f. oscillation, which is then demodulated, in the same way an FM receiver demodulates an FM carrier, and the output is available as an audio signal.

While this technique very conveniently eliminates many of our mechanical problems, at the same time it really transfers them into the electrical circuit, because now our linearity is determined by the correct operation of the FM demodulator circuit.

Type of Stylus

So much for general transducer principles. Before we discuss the various characteristics of these pickups in general there are some features related to the contact between disc and stylus that require attention.

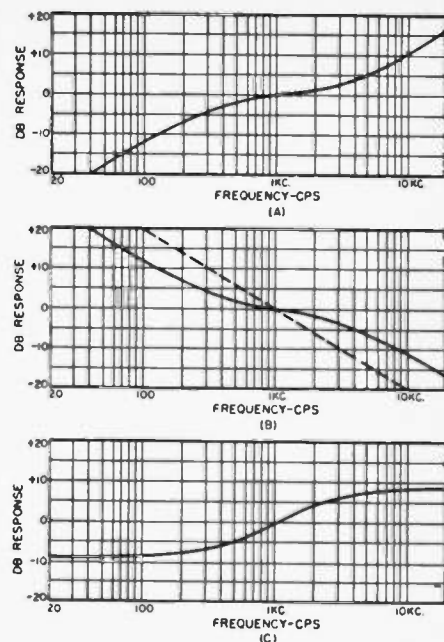


Fig. 4. Equalization characteristics associated with disc recording and playback. (A) Recording equalization for velocity-sensitive cutter head. (B) Playback (solid line) equalization for use with velocity-sensitive pickup and discs cut to the characteristic of (A). The dotted line is for a disc recorded at constant amplitude which would require this equalization for playback with velocity-sensitive pickup. (C) Playback equalization required for standard discs used with amplitude-sensitive pickup. This curve is the difference between solid and dotted lines of (B).

In the early days of phonograph pickups, steel needles were used and the record manufacturers introduced abrasive material into their discs, which served the purpose of keeping the steel needle sharpened to the correct shape during the playing of one disc. The single-play type needles wore down sufficiently during this playing to make them unsuitable for a second playing and so the needle had to be changed after every disc. Failure to do this resulted in serious damage to the record due to the needle becoming unduly enlarged at its point.

The use of successively harder grades of steel resulted in needles that would play for a longer time before needing to be changed. First, needles were developed that would last ten playings and then the so-called "permanent" type, which was sold in packets of ten instead of boxes of a hundred. Of course the fact that you bought ten in a packet belied the title "permanent."

These facts serve to illustrate a basic principle in the use of phonograph discs for transcription of program material. The recorded information has to be transferred from the disc groove to a stylus by mechanical friction between the groove of the disc and the point of the stylus. This inevitably involves wear. The early attempt was to make the needle suffer more than the record so as to preserve the quality of the record longer. However the wear was always considerable, because the needle was harder than the record material

and hence both had to wear to some extent.

The high frequencies invariably suffer more than the lower frequencies in the course of wear because they constitute much smaller deviations in the modulation of the groove. So the modern approach has been to reduce wear by reducing stylus pressure and using much harder stylus materials so as to minimize wear of both the disc and the stylus. Of course the degree of wear is still relative, because the mere fact that there is mechanical friction means that some wear must take place.

The successive development of sapphire and diamond styli has shown considerable improvement over the best hardened steel variety. The sapphire stylus will play much longer than the "permanent" needles just referred to, and the diamond stylus can be played over a thousand times without showing detectable wear.

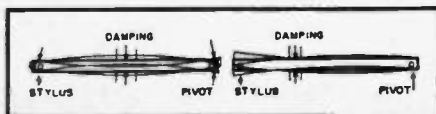
Stylus Force

This pressure is called stylus force and a feature in the design of all pickups is the stylus force necessary to maintain tracking. The word "tracking" is applied to two different things in phonograph reproduction. The meaning intended here concerns the ability of the stylus to stay in the groove at all the modulation frequencies in the audio band. The other meaning has to do with correct tone arm mounting, and will be discussed in another article.

If the stylus point tends to ride up the walls of the groove at any modulation frequency in the audio band, it is said to be failing to track at that frequency. This means that a greater pressure is required on the stylus to force it down into the groove and avoid this cause of distortion.

The principal way in which a pickup can be made to follow the groove more readily with less stylus force is by increasing the compliance in the stylus movement. Compliance is that quality which is opposed to the quality known as stiffness, thus, as the movement becomes less stiff it becomes more compliant. Obviously compliance is a matter that must be considered in conjunction with the ruggedness of the pickup, because the stylus has to maintain its correct operating position in the middle of its travel, otherwise it will run into distortion troubles.

Fig. 5. Effects of poor damping distribution along stylus arm. At (A) damping concentrated midway between the stylus and pivot, accentuates a resonance in the mode shown. The pivot vibrates more than it rotates at this frequency, so the effect will cause a dip in the frequency response. At (B) damping concentrated about one-third of the distance from stylus to pivot will accentuate resonance in a different mode, giving rise to a peak at this frequency, because the damping does not have to move with the stylus arm.



This means that every part of a pickup, including the arm in which it is mounted, must have extremely free movement if the best use is to be made of a high compliance. Thus we might say that the higher the stylus compliance, the less will be the record wear that the pickup produces. While this is true in principle, there are other factors that go along with compliance, which we will discuss in a moment.

Undoubtedly the best pickup for minimizing record wear is the *Weathers* which was mentioned earlier. It has a lateral compliance of 14×10^{-6} centimeters per dyne, which is much higher than for any other pickup on the market, and it will track with an effective stylus force of 1 gram, which is not more than half that required for the next best type of pickup.

Its disadvantage, as already mentioned, is that performance depends on accurate maintenance of the oscillator/FM part of the equipment. Most people prefer a pickup which gives an audio output direct, so that, if the pickup is mechanically sound, there is nothing else to worry about. However, if you are a perfectionist, and are prepared to go to some trouble with maintenance to keep your record wear to an absolute minimum and distortion low, you may well choose this pickup.

Design Features

A few words here about the general principles in the mechanical design of pickup movements. The best design is a compact one, in which the whole moving mechanism is of the simplest and most "streamlined" shape possible. Try to think of the moving mechanism vibrating in as many different modes as you can imagine. The more complicated its shape, the more complicated the possible modes in which the mechanism could vibrate, and each of these will produce variations in the frequency response in the form of resonances.

The stylus must be rigidly mounted to the armature, or moving coil, or whatever element forms the transducer and any damping, using blocks of rubber or other special material, must be mounted in such a manner that the damping controls the resonance of the system and does not encourage it to vibrate in some more complicated mode. For instance, damping applied half way back along the stylus arm, in a concentrated lump between the stylus point and the pivot, will tend to hold the point still where the damping is applied, and make the stylus vibrate at its tip and again at the pivot. As output from the pickup is dependent upon the moving mechanism *rotating* about the pivot, rather than vibrating there, this kind of movement will not result in output and will cause a hole in the response. This is illustrated at Fig. 5. Other kinds of undesired vibration can produce peaks in the response.

A sudden change in the cross-section can also result in undesirable characteristics in the response curve. The cross-section from the stylus tip back to the pivot should increase gradually, and

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not in any sudden step, and similarly the junction of the stylus with the transducer element should maintain a smooth change in shape and not one that gives sudden steps to the cross-section. All these things can add irregularities to the frequency response.

Assuming that these irregularities are reasonably well eliminated, there are two basic mass components and two compliance components that contribute to the over-all performance: the mass of the tone arm, and the mass of the stylus with its arm, the compliance of the stylus movements, and the compliance of the disc material. Three of these elements are under the control of the pickup and tone arm designers, while the fourth is liable to vary from disc to disc.

The mass of the tone arm in conjunction with the compliance of the cartridge tends to produce a low-frequency resonance. This should be adequately damped, either by damping in the tone arm movement itself, or by care in its construction, or the mass should be adjusted so that the resonance is beyond the low frequency end of the range and not at some frequency where it will emphasize rumble, and other undesirable effects. This will be more fully discussed when we come to the question of choosing a tone arm.

The mass of the stylus and its arm referred to the stylus point, in conjunction with the compliance of the stylus and the compliance of the disc material, should produce a resonant frequency at least up at the top end of the audio spectrum and preferably beyond it. The damping of the compliance should also prevent the resonance from producing a peak, that is, it should be critically damped, so as to produce a smooth roll-off.

There is the problem of selecting a suitable damping material for a pickup cartridge. Most manufacturers have solved it by using a material that suits their particular design, but the qualities of the material have a time limitation, which means the cartridge has a limited life before its performance begins to deteriorate.

Of course the final judgment of performance will be based on the frequency response of the cartridge and a listening test of the whole equipment. But a general look at the construction provides a good performance criterion.

Matching Problems

After considering the various pickups on their basic merits, we are still left with the problem as to whether we can match it successfully to our preamplifier.

All ribbon-type pickups and many moving-coil types require a transformer to step the output voltage up to something suitable for amplifying at the grid of the preamplifier input stage, because the impedance of the pickup is so low and its output voltage would be down in the noise unless a transformer were used. In most in-

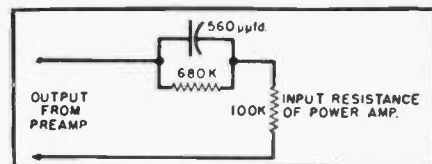


Fig. 6. A simple equalizer circuit for amplitude-sensitive types of pickup that can be placed between preamplifier and the power amplifier. It is calculated on the basis of the power amplifier having an input resistance of 100,000 ohms. For other values, the resistance should be multiplied and the capacitor divided by the factor by which the input resistance differs. For example, with an input resistance of 500,000 ohms, the resistance should be 3.3 megohm and the capacitor 120 μ fd. (nearest value). The arrangement should be carefully shielded, because of high impedance.

stances the manufacturer either supplies or recommends a suitable transformer which will serve this purpose. It is essential to consider where the transformer may be placed, as failure to pay careful attention to this can result in serious hum pickup. The transformer must be placed well away from any supply transformers or motor in the equipment and will probably need careful orientation to get a minimum of hum pickup from these components.

All pickups using a transformer will need careful attention to the loading on the secondary of the transformer to avoid a high-frequency resonance due to the transformer itself. This means that the grid resistance of the input stage will need adjusting to get the best over-all response.

The magnetic, or variable reluctance, or moving-iron type cartridge—which ever you prefer to call it—comes in an impedance suitable for the input of the average preamplifier and requires attention only to the necessary loading resistance. If the preamplifier has a resistance across the input of higher value than that specified for the cartridge used, there will be a rise in high frequency. If the value is lower the high frequencies will roll off somewhat. Maybe this can be corrected in the equalization of the preamplifier or maybe the resistor should be changed to suit the cartridge.

All of the foregoing cartridges require a low-level or high-gain input at the preamplifier to get sufficient gain to supply the power amplifier.

The piezoelectric pickups and the Weathers FM pickup each provide a high-level output that does not need a low-level input on the preamplifier. On the contrary, a high-level input, such as that normally used for input from a tuner, is adequate for this purpose.

The input should be high impedance. This is essential for the piezoelectric type, otherwise there will be a low-frequency loss due to the fact that the pickup has a capacitance source impedance instead of a resistance.

Quite pleasing results can be obtained with these pickups without any equalization, but this really assumes that the discs have a constant-ampli-

tude instead of constant-velocity recording characteristic. A little equalization, using the response shown in Fig. 4C, produces an over-all flat response from this type of cartridge. If this is not provided in the preamplifier, a useful circuit for adding it between the preamplifier and power amplifier, assuming that the input resistance of the power amplifier is 100,000 ohms, is shown in Fig. 6. Take care to shield the entire circuit to avoid hum pickup.

The life of any pickup being limited, and it being a vital link in phono reproduction, if you have any perfectionist leanings, you will probably end up trying at least one of each type in the course of time.

To summarize then: for a low-cost system, requiring a pickup with reasonable fidelity and a large output, the ceramic and crystal types are ideal; but if you want to get absolutely the best from your records, in terms of both reproduction and number of playings before wear begins to show up, you need a type that will reduce stylus force necessary and give you the lowest distortion. Suitable types are: the higher quality variable reluctance, the moving coil, and the ribbon. Taken in this order, however, they will require progressively more gain from your preamplifier. Another high-quality possibility is the Weathers FM pickup, which needs the FM demodulator that comes with it, but has a high-level output.

EDITOR'S NOTE: *The general comments with regard to the crystal cartridges are, of course, the opinions of the author and are generally well-known. However, we would like to call to the reader's attention that there are crystal cartridges on the market today that, according to the manufacturers, are extremely good to a point where they compare favorably with magnetic units. These reports have not been verified as yet.*

One of the most important things about the performance of a pickup is its frequency response. But published responses are of rather limited meaning, particularly at the high end; this is because the record material compliance contributes toward the response measured and different test records use different pressing compositions, and so do not give consistent results with different pickups. This means that pickups will not give consistently different performance either, when compared on program records using different pressing composition. For this reason, published characteristics can best be assessed by listening to the differences they represent, which means the ear should be the final arbiter in making a choice. After making your preliminary choice from the published data, it is advisable to listen to the pickup played through your own equipment, or one as similar to it as possible, to judge whether or not it will give satisfaction.

Happy hunting!

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Like all Miller products this #595 tuner is guaranteed to perform to your satisfaction. Buy with confidence from your Radio and TV Parts Distributor, Audio Distributor, or High Fidelity Dealer.

It is one of the lowest priced quality tuners on the market. The #595 tuner is complete in every detail, ready to plug into your audio system. Net weight 1 lb. 8 ozs. Shipping weight gross 2 lbs. Net price only \$19.50 plus excise tax.

NO POWER REQUIREMENTS
SELECTIVITY • SENSITIVITY • NO NOISE • DOES NOT ATTENUATE HIGH FREQUENCY RESPONSE • NO BIRDIES
LOW COST • LIFETIME TROUBLE-FREE OPERATION • NOTHING TO CAUSE DISTORTION • GAIN CONTROL AND VERNIER TUNING DIAL

The audio output of the tuner is proportional to the input signal and will vary from .07V to .7V for stations within a 20-25 mile radius. We recommend a good antenna and a high quality audio system for best results.

A beautiful ebony black bakelite cabinet houses the tuner with a Richlow brass etched panel for the escutcheon. Both combine to complement any decor.

J. W. MILLER COMPANY

5917 South Main Street • Los Angeles 3, California

Canadian Representative: Atlas Radio Corp., Ltd., 50 Wingold Ave., Toronto 10, Ont., Canada



MUSCLES DON'T MAKE MUSIC

Great music, superbly rendered, draws life and warm beauty from the teamwork of the true artist's heart and hands. Precise coordination of mind and body — not muscular power — is the secret of artistic accomplishment.

Great performance is faithfully recreated in the full brilliance of original rendition by electronic teamwork in high fidelity sound systems.

Newcomb compact amplifiers and tuners give you that all-important electronic teamwork—designed-in. They have no overpowered, "inflated" parts nor trick gadgets "hung on" for sales-talk purposes. From circuit through cabinet, Newcomb hi-fi system components are tailored to true balance, to inter-relationship and interaction as precise and coordinated as the fingers and feelings of the musical artist. Only through such thorough integration can you be assured the utmost in listening pleasure.

for the best in home hi-fi, look to Newcomb

Ask your dealer to demonstrate the new Compact Series of amplifiers and tuners

NEWCOMB
the Sound of Quality Since 1937

NEWCOMB, Dept. F-A
6824 Lexington Ave., Hollywood 38, Calif.

Enclosed is 25c for booklet "Hi-Fi Is for Everybody."

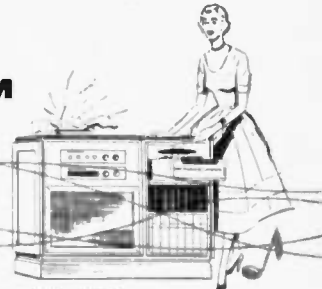
Send name nearest Newcomb dealer and details on Newcomb Compact.

name _____
address _____
city _____ state _____

"BUILD-IT-YOURSELF" AND ENJOY

high fidelity at its finest

IN KIT FORM



1 Heathkit FM TUNER KIT

Features brand new circuit and physical design. Matches WA-P2 Preamplifier. Modern tube line-up provides better than 10 uv. sensitivity for 20 db of quieting. Built-in power supply.

Incorporates automatic gain control—highly stabilized oscillator—illuminated tuning dial—pre-aligned IF and ratio transformers and front end tuning unit. Uses 6BQ7A Cascode RF stage. 6U8 oscillator—mixer, two 6CB6 IF amplifiers, 6AL5 ratio detector. 6C4 audio amplifier, and 6X4 rectifier. **MODEL FM-3**
\$2450
Shpg. Wt. 7 Lbs.

2 Heathkit 25-Watt HIGH FIDELITY AMPLIFIER KIT

Features a new-design Peerless output transformer and KT66 output tubes. Frequency response within ± 1 db from 5 cps to 160 Kc at 1 watt. Harmonic distortion only 1% at 25 watts, 20-20,000 cps. IM distortion only 1% at 20 watts. 4, 8, or 16 ohms output. Hum and noise, 99 db below rated output. Uses 2-12AU7's, 2-KT66's and 5R4GY. Attractive physical appearance harmonizes with WA-P2 Preamplifier. Kit combinations:

W-5M AMPLIFIER KIT: Consists of main amplifier and power supply, all on one chassis. Shpg. Wt. 31 Lbs. Express only. **\$5975**

W-5 COMBINATION AMPLIFIER KIT: Consists of W-5M amplifier kit plus Heathkit Model WA-P2 Preamplifier kit. Shpg. **\$7950** wt. 38 Lbs. Express only.

3 Heathkit HIGH FIDELITY PREAMPLIFIER KIT

Designed specifically for use with the Williamson Type Amplifiers, the WA-P2 features 5 separate switch-selected input channels, each with its own input control—full record equalization with turnover and rolloff controls—separate bass and treble tone controls—and many other desirable features. Frequency response is within ± 1 db from 25 to 30,000 cps. Beautiful satin-gold finish. Power requirements from the Heathkit Williamson Type Amplifier. **MODEL WA-P2**
\$1975
Shpg. Wt. 7 Lbs.

4 Heathkit Williamson Type HIGH FIDELITY AMPLIFIER KIT

This amplifier employs the famous Acrosound TO-300 "Ultra Linear" output transformer, and has a frequency response within ± 1 db from 6 cps to 150 Kc at 1 watt. Harmonic distortion only 1% at 21 watts. IM distortion at 20 watts only 1.3%. Power output 20 watts. 4, 8, or 16 ohms output. Hum and noise, 88 db below 20 watts. Uses 2-6SN7's, 2-5881's and 5V4G. Kit combinations:

W-3M AMPLIFIER KIT: Consists of main amplifier and power supply for separate chassis construction. Shpg. Wt. 29 lbs. Express only. **\$4975**

W-3 COMBINATION AMPLIFIER KIT: Consists of W-3M amplifier kit plus Heathkit Model WA-P2 Preamplifier kit. Shpg. **\$6950** Wt. 37 lbs. Express only.

5 Heathkit Williamson Type HIGH FIDELITY AMPLIFIER KIT

This is the lowest price Williamson type amplifier ever offered in kit form, and yet it retains all the usual Williamson features. Employs Chicago output transformer. Frequency response, within ± 1 db from 10 cps to 100 Kc at 1 watt. Harmonic distortion only 1.5% at 20 watts. IM distortion at rated output 2.7%. Power output 20 watts. 4, 8, or 16 ohms output. Hum and noise, 95 db below 20 watts, uses 2-6SN7's, 2-5881's, and 5V4G. An exceptional dollar value by any standard. Kit combinations:

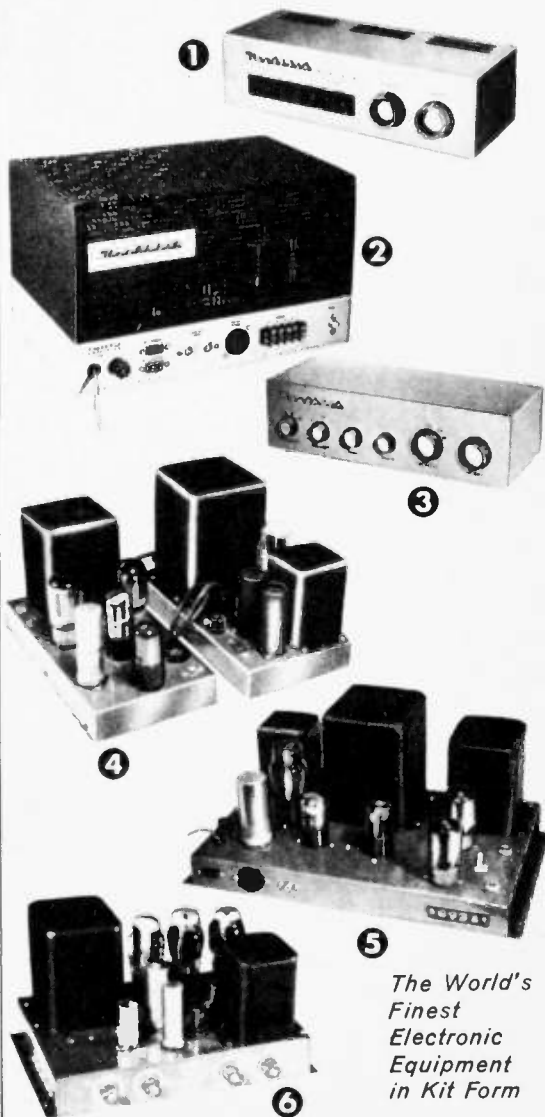
W-4AM AMPLIFIER KIT: Consists of main amplifier and power supply for single chassis construction. Shpg. Wt. 28 lbs. Express only. **\$3975**

W-4A COMBINATION AMPLIFIER KIT: Consists of W-4AM amplifier kit plus Heathkit Model WA-P2 Preamplifier kit. Shpg. **\$5950** Wt. 35 lbs. Express only.

6 Heathkit 20-Watt HIGH FIDELITY AMPLIFIER KIT

This model represents the least expensive route to high fidelity performance. Frequency response is ± 1 db from 20-20,000 cps. Features full 20 watt output using push-pull 6L6's and has separate bass and treble tone controls. Preamplifier and main amplifier on same chassis. Four switch-selected inputs, and separate bass and treble tone controls provided. Employs miniature tube types for low hum and noise. Excellent for home or PA applications. **MODEL A-9B**
\$3550
Shpg. Wt. 23 Lbs.

Heathkits®



The World's Finest Electronic Equipment in Kit Form

Heathkits are easy to build..

Heathkit construction manuals are full of big, clear pictorial diagrams that show the placement of each lead and part in the circuit. In addition, the step-by-step procedure describes each phase of the construction very carefully, and supplies all the information you need to assemble the kit properly. Includes information on resistor color-codes, tips on soldering, and information on the tools you need. Even a beginner can build high quality Heathkits and enjoy their wonderful performance.

HEATH COMPANY

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**BENTON HARBOR 15,
MICHIGAN**

HI-FI ANNUAL & AUDIO HANDBOOK

Only the PRESTO SR-27 Tape Recorder

gives such

Professional Performance

at such a popular price!



The top performance features of the finest PRESTO units are yours in this 2-unit combination—R-27 tape recorder and A-920B amplifier. *Check the features of the recording unit:* 3-motor drive; separated record, erase and playback heads; fast forward and rewind. There's no take-up reel clutch

or idler pulley. *Compare the features of the companion amplifier:* 10 watt output at 16 ohms, self-contained power supply, separate preamps and VU meter and *two* playback speakers. You can monitor the tape during recording. This professional performer is yours for just \$588 complete.



PRESTO

Pirouette

T-18 TURNTABLE

... finest low-priced professional turntable has world-famous flick shift—one sideway flick selects any speed.

There is no finer, smoother-running or easier-to-operate instrument for control room, studio or hi-fi system than the *Pirouette* T-18—the PRESTO achievement that sets a new high in turntable design. You select any speed—33 $\frac{1}{3}$, 45 or 78 rpm—with a sideway flick of the exclusive PRESTO 3-speed shift. You engage the proper idler for the speed you want with one motion. There are no bothersome arms or shift cams. And check the extras you get: extra heavy-weight wide-beveled table, deep-well bearing, sleek styling in telephone black and brushed chrome. \$53.50. With hysteresis motor, \$108.

Watch for
the

NEW

PRESTO

K-11 DISC RECORDER

... improved professional version of PRESTO's all-time favorite K-10 — incorporates revolutionary T-18H turntable*

K-11 is the smart new disc recorder you can fit into your sound system — or use as an on-location recorder. It's featherweight with a completely new pick-up arm, fully encased hi-fi speaker, smart new panel design with push button controls and the dependable PRESTO cutting head. You get excellent broadcast-quality fidelity. For those times when a disc recorder is preferable, the K-11 is your best bet. Cuts discs up to 13 $\frac{1}{4}$ inches in diameter. Three-speed operation.

**with hysteresis motor*



PRESTO

RECORDING CORPORATION

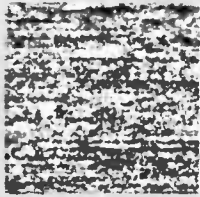
PARAMUS, NEW JERSEY

Export Division: 25 Warren Street, New York 7, N. Y.
Canadian Division: Instantaneous Recording Service, 42 Lombard St., Toronto

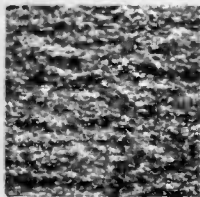
WORLD'S FOREMOST MANUFACTURER OF PRECISION RECORDING EQUIPMENT AND DISCS

It Won't Come Off!

These actual unretouched photomicrographs (50X) of FERRO-SHEEN and ordinary tape, taken under identical lighting conditions, emphasize surface irregularities of both tapes. See how irregular the eggshell surface of the ordinary tape appears in comparison with that of FERRO-SHEEN. See how much smaller are the shadows and highlights of the FERRO-SHEEN process tape, indicating a greater uniformity of oxide coating and a smoother surface.



**FERRO-SHEEN
PROCESS TAPE**



**CONVENTIONAL
TAPE**



ALL RECORDING TAPE is coated with magnetic oxide. On ordinary tapes this coating rubs off in use and forms a harmful deposit of abrasive dust on the recording head. Unless the head is constantly cleaned, the collection of abrasive dust eventually wears it out. A further disadvantage of oxide-shedding, common to all ordinary tapes, is that after a few playings, the tape loses enough coating to alter its original frequency response characteristic.

Make Your Own Test In Twenty Seconds!

Send for free Comparator Card, which has strips of all the leading tapes mounted side by side for your own visual comparison. You will SEE the difference at a glance! The difference you will see is responsible for the improvement you will HEAR... the mirror-smooth surface of the FERRO-SHEEN tape results in vastly improved contact between recording head and tape and gives you optimum magnetic performance... all the highs, all the lows.



WITH TAPE MADE BY THE EXCLUSIVE

irish

FERRO-SHEEN PROCESS

The Oxide Will Not Rub Off!

FERRO-SHEEN IS NOW AVAILABLE IN THESE THREE QUALITY irish TAPES:

SHAMROCK #300: The ultimate in premium, professional tape for broadcast and studio use. 1.5 mil plastic base. Comes with 5" Mylar leader in dust-proof polyethylene bag.
1200 feet _____ on 7" reel

LONG-PLAY #600: Extends playing time 50% over conventional tapes on same size reel. Same premium qualities as Shamrock #300, but on Dupont's 1 mil new miracle film Mylar.
1800 feet _____ on 7" reel

DOUBLE-PLAY: Doubles your playing time with 2400 feet on standard 7" reel. For all applications requiring uninterrupted operation. Dupont Mylar base.
2400 feet _____ on 7" reel

If not available at your local dealer, write: ➔

The **irish FERRO-SHEEN** process of tape manufacture anchors the oxide coating to the base permanently, inseparably and much more smoothly. The obvious advantage of the homogeneous bond is that *the entire vicious cycle of shedding and abrasion of recording head and tape is eliminated, resulting in longer life for the tape, longer life for the head and flat frequency response over a wider range.*

ORRADIO INDUSTRIES, INC., Opelika, Alabama

World's Largest Exclusive Magnetic Tape Manufacturer

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