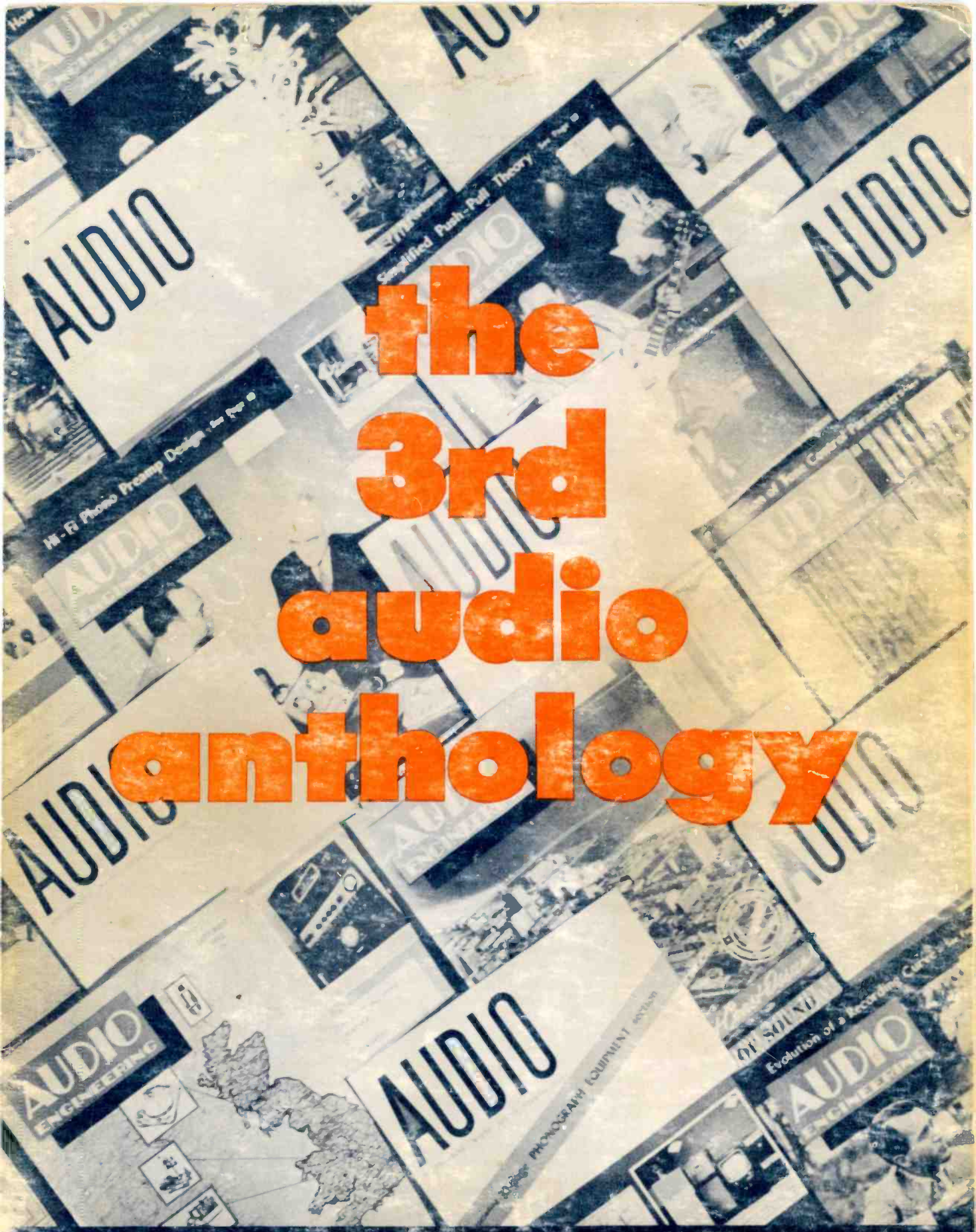
the 3rd audio anthology



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the 3rd audio anthology

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by

C. G. McProud
Editor

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Foreword

Continuing the practice of providing the reader with a single volume that contains most of the material from issues of the magazine of greatest interest to the serious audio experimenter, the *3rd audio anthology* is offered in the sincere hope that its pages will continue to provide a wealth of useful reference material without the necessary additional pages that go to make up a monthly magazine.

The original *audio anthology* was created to bring the early articles that appeared in AUDIO ENGINEERING into the hands of those who did not learn of the magazine's existence until some time after it was introduced. The articles that were in the first thirty issues of AUDIO ENGINEERING were among the most valuable that had been published in magazines up to that time, and once the early copies of the magazine were exhausted, there was not any way in which the reader could secure them for his own use. Thus the *audio anthology* was born.

As months went on, more and more readers—both of the magazine and of the anthology—came to expect that there would be another compilation of the more popular articles, and so *the 2nd audio anthology* was introduced. With a precedent thus created, it was natural to expect that a third would be along soon, and here it is.

We are grateful to our readers for their continued interest, and it is to them that this *3rd audio anthology* is appreciatively dedicated.

C. G. McPROUD, *Editor*
AUDIO

Mineola, N. Y.
September, 1950

A Simple High-Quality Phono Amplifier

R. D. MIDDLEBROOK*

An inexpensive audio amplifier with sufficient gain to drive a loudspeaker from a magnetic pickup and which uses an unusually large amount of negative feedback over the output tube and transformer.

THE BEST AUDIO AMPLIFIER is wasted if the associated equipment can not take advantage of it. Many high-quality amplifiers are available commercially, all of them using push-pull output stages. Their capabilities are impressive—and so are their prices. Of the thousands of non-technical people who have bought these amplifiers, how many have spent a *proportionate* amount of money on their loudspeaker systems?

There is room for a cheaper and simpler audio amplifier which can still be considered high-quality. It is not generally believed that a single-ended output stage is capable of producing results in keeping with present-day standards; yet it can be done. This article will describe how excellent listening quality can be obtained from a circuit which, at first glance, doesn't look as though it would sound any better than a small table radio.

The secret of this transformation is negative feedback. It is the writer's belief that most of the unpleasantness of the reproduction from an ordinary single tetrode or pentode output is due to the low speaker damping, and not to the harmonic distortion. Without entering into the fierce discussion on the merits or otherwise of negative output impedance of amplifiers, it can be stated with some confidence that an output resistance somewhat less than the speaker resistance is desirable in order to obtain satisfactory damping. By using enough negative feedback this can be achieved with a tetrode output tube, with all the attendant advantages of reduced distortion and extended frequency response. The amplifier to be described uses a single 6V6 beam tetrode tube, with an ordinary universal output transformer. The performance figures show that the results obtained are definitely in the "high-quality" class:

Maximum output:	3 watts
Total distortion:	1 per cent at 3 watts (1000 cps)
Frequency range:	3 db down at 20 cps and 25,000 cps
Output resistance:	0.1 ohm

The maximum power output of 3 watts is obtained at middle frequencies; the maximum power obtainable at extreme low and high frequencies depends on the quality of the output transformer. No amount of negative feedback can effect any improvement in this respect. Using the Thordarson Type T22S58 universal output transformer, the maximum available power output falls to 3 db below 3 watts at 80 cps and at 8000 cps.

Since most present-day amplifiers professing to be "high fidelity" are capable of at least 10 watts output, a word in defense of 3 watts may not be out of place. 10 or 20 watts certainly give a considerable margin of safety, so that even the peaks of the signal give rise to distortion less than the figures quoted for maximum output. What the loudspeaker does to the transient is, of course, another story. In this amplifier, owing to the large amount of negative feedback, the signal contains very small percentages of distortion right up to the maximum output of 3 watts, and if the signal is increased further, the peaks are clipped. This effect is clearly visible on an oscilloscope, not only for a sine wave but also for the complex waves of music or speech. As mentioned previously, the guiding principle in the design of the amplifier is economy in construction while maintaining adequate performance for ordinary home listening: the acid test for success is therefore a listening test. If an oscilloscope is connected across the loudspeaker, while reproducing heavily-recorded music, it will be seen that the gain can be increased until the volume is sufficient to drown all conversation before the peaks start to clip. In other words, all activity must cease while the music is at full volume—a condition which must surely warm the heart of any hi-fi enthusiast. Need more be said about the adequacy of 3 watts?

The complete amplifier has sufficient gain to produce full output from a magnetic pickup, such as the General Electric variable reluctance type. *Figure* 1 is a block diagram of the various stages, and it is seen that the circuit can be divided into two distinct parts, each embodying heavy overall negative feedback.

* 1928 Cooley Ave., East Palo Alto, Calif.

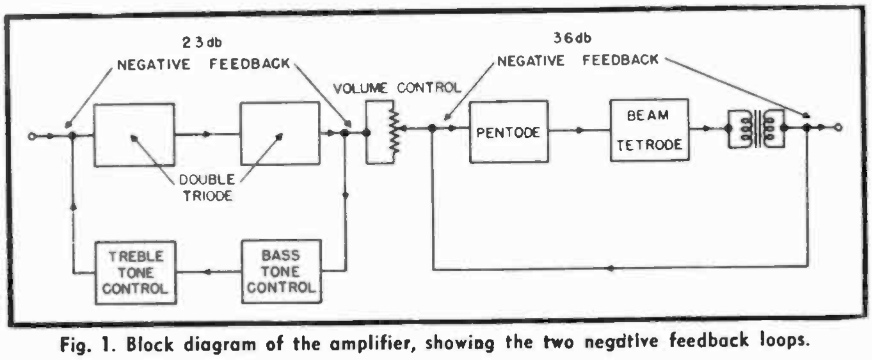

Fig. 1. Block diagram of the amplifier, showing the two negative feedback loops.

Circuit Description

Figure 2 shows the complete circuit in detail. The grid resistor R_1 is chosen to be a suitable load for the pickup. The first two triode sections form a straight cascaded amplifier with a gain of 1400 times, which is reduced to an over-all gain of 100 times by the negative feedback from the plate of V_2 to the cathode of V_1. This link contains the tone control circuits. Various tone control arrangements are possible, and can be chosen to suit the individual: however, two networks which have been found adequate are described here.

Figure 3 is the bass control, which gives a maximum of 12 db bass boost, the magnitude of the rise being controlled by the potentiometer. The point at which the rise begins is selected by the switch. These turnover frequencies are 80, 140, 200, 300, 400, and 500 cps. The bass boost is sufficient to compensate for the recording characteristic of LP records, and in practise it will be found that usually the maximum amount of boost is unnecessary.

Figure 4 shows the treble control, which has one position giving 3 db rise, one flat position, and four positions giving treble cut. The potentiometer determines the degree of cut, and the switch selects the turnover frequency. These turnover frequencies are 1500, 2000, 4000, and 7000 cps. The 0.1 megohm resistor is merely to prevent a thump at the speaker when switching from the flat to the lift position.

Raised in level by 40 db by V_1 and V_2, the signal is fed to the volume control and then to the grid of V_3. This tube has the unusually high plate load, R_{10}, of 2.2 megohms, and is direct-coupled to the grid of V_4. Using a 2.2 megohm load enables a gain of 340 times to be realized; if a.c. coupling were used, the grid resistor of V_4 would have to be at least 10 megohms to avoid shunting the load of V_3, and this exceeds the tube manufacturer's maximum permissible value of 1.0 megohm. Using direct-coupling also has the advantage that no low-frequency phase shift is introduced.

For an 8-ohm loudspeaker load, an output transformer turns ratio of 32 to 1 should be used. The voltage "gain" from the grid of V_4 to the transformer secondary is approximately 0.76, thereby giving a gain from the grid of V_3 to the transformer secondary of 340 × 0.76 = 260 times. Since the feedback voltage returned to the cathode of V_3 is one fourth of the output voltage, by virtue of the 470- and 1500-ohm resistors forming a voltage divider, the over-all gain is reduced from 260 times to 4 times: that is we have $20 \log(260/4)$, or 36 db of negative feedback.

V_3 obtains its bias voltage from the very small grid current flowing through the 4.7 megohm grid resistor; this is commonly described as "contact potential." Bias amounts to about minus 0.75 volt. The voltage developed across the 470-ohm cathode resistor is only some 0.07 volt, and is therefore negligible. The reason for using this method of biasing is that it avoids the low-frequency phase shift which would be introduced by the conventional cathode resistor and bypass capacitor.

One of the troubles inherent in direct-coupled stages is the drift of operating point. This is avoided here by deriving the screen voltage of V_3 from the cathode of V_4, through the 0.47-meg resistor. The operation is as follows: suppose the plate current of V_3 decreases, then its plate voltage increases, carrying the grid of V_4 with it. The plate current of V_4 then rises, and its cathode voltage increases. This also raises the screen voltage of V_3, tending to increase the plate current of V_3 and restore the operating point. Thus we have in effect an internal negative feedback loop which stabilizes the working points of V_3 and V_4. This feedback loop is made inoperative for signal voltages by decoupling the screen of V_3.

The resistor R_{13} and capacitor C_7 between the plate of V_4 and the cathode of V_3 are to suppress high-frequency oscillations. The capacitor should be a high-quality mica type rated at 1000 volts. If different taps are used on the output transformer, or if a different transformer is used, the values of these two components may need to be adjusted. The series grid resistor of V_3 and the screen resistor of V_4 are included to guard against parasitic oscillations.

The total current drain of the complete amplifier is only about 40 milliamperes, and hence the power pack requirements are modest. Using a 350-0-350 volt high voltage winding on the power transformer, sufficient voltage is available to dispense with a choke in favor of the much less expensive resistor for smoothing purposes. The less efficient smoothing thus obtained is nevertheless quite adequate, because the large amount of negative feedback reduces the 120-cycle hum from the B+ line to complete inaudibility. It is desirable to use one or more of the standard methods for reducing 60-cycle hum from the heater of V_1, such as tapping a ground connection across the heater winding, (as shown) or raising the heater winding to some positive potential with respect to ground.

Only one other point need be mentioned in connection with practical construction. The cathode resistor, both grid resistors, and grid capacitor of V_3 should be mounted as close as possible to the tube socket. This is because the input impedance to V_3 is extremely high and the grid leads are very sensitive to hum pickup.

All figures quoted above are measured values, not calculated. Aural results are excellent: the cleanness of the bass response is particularly pleasing. The entire amplifier can be built on a chassis 8 by 6 inches, and the quality of its reproduction bears comparison with that of amplifiers many times more expensive.

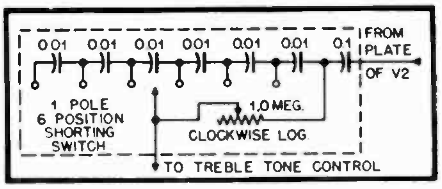

Fig. 3. A suggested bass-boost circuit. This section corresponds to box "A" in Fig. 2.

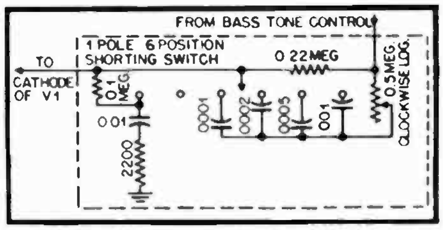

Fig. 4. A suggested treble tone-control circuit. This corresponds to box "B" in Fig. 2.

PARTS LIST—AMPLIFIER

C_1	0.1 µf, 600 v, paper
C_2	50 µf, 6 v, electrolytic
C_3	.01 µf, 600 v, paper
C_4	.001 µf, 500 v, mica
C_5	0.5 µf, 200 v, paper
C_6	500 µf, 50 v, electrolytic
C_7	100 µµf, 1000 v, mica
$C_{8a, b}$	10-10 µf, 450 v, electrolytic
C_9, C_{10}	20 µf, 500 v, electrolytic
R_1	pickup load resistor (see text)
R_2, R_5	0.12 meg, 1 watt
R_3, R_6	2200 ohms, ½ watt
R_4, R_{14}	0.47 meg, ½ watt
R_7	0.5 meg audio taper potentiometer
R_8	4.7 meg, ½ watt
R_9, R_{21}	27,000 ohms, ½ watt
R_{10}	2.2 meg, 1 watt
R_{11}	470 ohms, ½ watt
R_{12}	56,000 ohms, ½ watt
R_{13}	330 ohms, ½ watt
R_{15}	1250 ohms, 5 watts
R_{16}	1500 ohms, 1 watt
R_{17}	47,000 ohms, 1 watt
R_{18}	10,000 ohms, 1 watt
R_{19}	1000 ohms, 5 watts
R_{20}	0.27 megs, ½ watt
T_1	Output transformer, "Universal" type, 32:1 ratio for 8-ohm loudspeaker
T_2	Power transformer, 350-0-350 v at 50 ma; 6.3 v at 2.0 amps

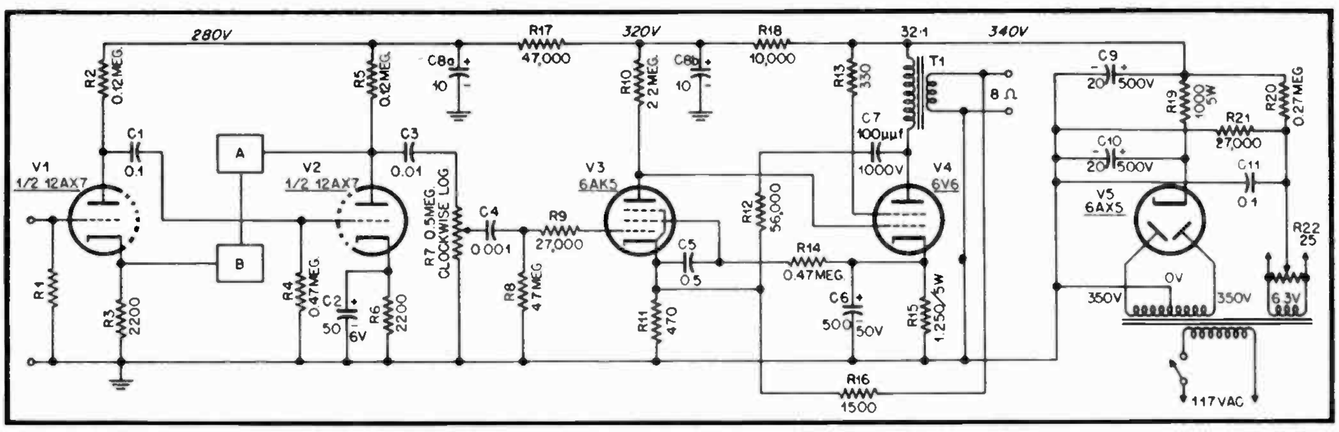

Fig. 2. The complete schematic of the amplifier.

A Quality Amplifier For The Home

ALBERT PREISMAN*

Mothers and wives can operate this unit, which has a built-in preamplifier and an absolute minimum of confusing knobs and dials. Even the selector switch is cleverly eliminated; manufacturers take note!

THE HI-FI ENTHUSIAST has been the butt of innumerable jokes for many years. The most fanatic have been accused of listening, not to music, but only to white noise, and the somewhat more reasonable members of this irascible fraternity make a fetish of "systems" with a bewildering multiplicity of knobs and switches, so as to be able to control every conceivable characteristic of the reproduced sound (except, apparently, its loudness).

The writer was accused by his good frau of never having a radio or phonograph that was simple in operation, presentable in appearance, and satisfying in performance, and the thing that rankled most in his mind was, of course, the truth of the accusation. So he sat down to figure out a circuit that could accommodate a phonograph pickup, an FM tuner, and a magnetic tape recorder, to be housed in a presentable cabinet, and arranged to be operated with a minimum of switches and controls.

The pickup chosen was an ordinary GE variable-reluctance dual type having a 1-mil diamond stylus and a 3-mil sapphire stylus. The necessary preamplifier was built into the main amplifier rather than on a separate chassis. The power supply, however, was built on a separate chassis to minimize hum pickup.

The FM tuner happened to be a Meissner; of course any other make can be used. The important thing is that the output level is much higher than that of the phonograph pickup, hence the tuner can be connected into the system at a point immediately following the preamplifier.

The tape recorder happened to be a Magnacordette. This has its own preamplifier, which functions both for recording and reproducing, and in the latter case has an output level comparable to that of the FM tuner. Hence it would be only natural to connect the recorder into the system at about the same point as the tuner.

This, then, suggested the layout that is actually employed. Following the preamplifier is an amplifier stage consisting of two sections of a 12AX7 whose plates are connected in parallel to a common plate load resistor. The grid of one section connects to the preamplifier, but the grid of the other section is fed, through a pair of separate series 75,000-ohm resistors, from the FM and tape-recorder outputs. The two series resistors merely serve to isolate the tuner and recorder from one another to a satisfactory degree.

In other words, the stage following the phono preamplifier is an electronic mixer stage, and the grid of one section is in itself arranged to mix two inputs. Hence a total of three inputs can be accommodated, and any one, two, or all three sources can be used to feed the loudspeaker. It requires no particular technical savvy to operate: if you want to play the phonograph, you turn on the amplifier, start the turnable, place the pickup on the record, and there you are. If you want to play the FM tuner, turn on the tuner as well as the main amplifier; if your wish is the tape recorder, turn it on as well as the main amplifier. Should you—perish forbid—wish to play two at once, or all three, merely turn on those you want. This is all admittedly very obvious, hence it is a desirable characteristic for the family instrument.

Now, as to tone controls: Women are not interested in engineering; that's why engineers have to work overtime designing engineering gadgets that do the technical work for them. Witness automatic transmissions, power steering, automatic washing machines, pre-cooked oatmeal, etc. Hence no equalizer was employed in this system; it has a fixed amount of bass boost, and a flat high-frequency response. The only control is a tone control of the "chopper-offer" type; it provides a rather sharp cutoff at 10 kc, 8 kc, or 5 kc, as desired, and is used (only by me, of course) to cut down the excessive surface noise on some old 78-r.p.m records that I have.

Perhaps the rest of the family will be enticed in time to use it, but since they play only LP's and 45's, they appear to have little need for it. It is desirable to have this control act on all three inputs, hence it is arranged to follow the mixer stage, and so requires a separate tube section for this purpose.

The present horsepower race among manufacturers of the horseless carriage had its origin in the power output race that began in the 20's, when the UX 112-A supplanted WD-11 or UV 201-A tubes in push pull. There is a group of diehard moderates who claim one or two watts output is sufficient. However, audio power is relatively cheap, and so this amplifier employs two 1622 tubes in push-pull, with a power output of 16 watts. Such power output may not be necessary for the home, but then one is under no compulsion to turn the gain up to a maximum. And if it should be desired to employ this amplifier for a larger room, the power is there to use.

However, the amplifier described here is readily modified to suit the specific needs of the builder. The tone control can be eliminated if not desired; the bass boost can be made variable, and smaller power tubes can be employed. Other possible variations may occur to the reader as he persues the description that follows.

The Phono Preamplifier

The complete amplifier diagram is shown in *Fig.* 1. The preamplifier consists of a 12AX7, with its two sections connected in cascade and employing feedback. But first we note the R-C input networks to the first grid; the use of these tends to cut down turntable rumble, although the best way to minimize this is to use a turntable as free of this characteristic as possible, and then to install it in a rock vault 10 feet underground.

Incidentally, the ordinary three-speed changer that I am using had excessive rumble until I slipped some rubber snubbers between the metal chassis and the wooden frame on which it is supported by three conical springs. It seems that the springs are too free and permit the whole assembly to quiver like a dowager's third chin at a gabfest; the rubber snubbers hold it in check like a whalebone corset.

The input resistor of 10,000 ohms seems to give the most satisfactory high-frequency response. An initial value of 20,000 ohms permitted the GE pickup to develop a double-humped peak at 6,000 and 10,000 cps, with a minimum at 8,000; the present value produces a reasonably smooth response.

Low-frequency boost is obtained by feeding back from the plate of the second tube to the cathode of the first via a .04-μf capacitor in series with a 47,000-ohm resistor. By varying the values of these two components, the low-frequency peak may be varied to a certain extent, both in magnitude and frequency. However, an additional boost is obtained by the .011-μf capacitor and 270,000-ohm resistor shunting the 50,000-ohm volume

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control, together with the 75,000-ohm resistor above this combination.

This latter network produces a reduction in gain, but the gain is more than sufficient anyway, and the additional bass boost of 4.2 db raises the total to 19 db. Of course more can be had, but the response from a test record using the RCA New Orthophonic curve seems quite satisfactory for the values given.

One advantage of employing two separate circuits for bass boost is that their peaks may be staggered and thus used to shape the curve to a certain degree. The curve as obtained for the values shown is exhibited in *Fig.* 2; the curve marked "Phono Input" shows a peak centered at about 45 cps. This is to be compared with the response marked "FM Input," which is that of the amplifier stages following the preamplifier. The slight dip below 50 cps in the latter curve is mainly due to the output transformer.

The 75,000-ohm resistor in the bass-boost network can be made variable in order to vary the amount of bass boost. The 540-ohm cathode resistor of the first stage may be changed to a potentiometer, with the 47,000 ohm resistor connected to the arm. In this way the amount of bass boost can be varied in this network without varying the position of the peak in the frequency spectrum.

A switch in series with the feedback network can be used to disconnect it when a flat response is desired, such as when it is desired to connect a microphone. The concomitant increase in gain is usually desirable in such a case because of the normally lower output of the microphone. However, I prefer to use the preamplifier in the Magnecordette for this purpose, hence I have omitted the switch.

One further point to note is the .015-μf cathode bypass capacitor employed in the second stage. It does not take effect until about 4,000 cps, at which point it begins to decrease the inverse feedback in this stage and thus raise the high-frequency response. This appears necessary in the case of triodes because of the appreciable input capacitance of a triode stage owing to the Miller effect. The resulting attenuation of the higher frequencies by several such stages is counteracted to an appreciable degree by this small bypass capacitor. By varying its value, the point at which compensation begins to take effect can be varied; a smaller value causes the compensation to set in at a higher frequency.

When the amplifier was completed, it was found that the hum was appreciable, owing in part to the bass-boost. The solution was to feed the heater of the first 12AX7 (whose two sections constitute the preamplifier) with the plate current of the two 1622 tubes operating in push-pull. Since these operate essentially class A, the even-harmonic components that are present in the combined cathode current are relatively small, and produce no noticeable effects in the preamplifier. The reduction in hum is very marked; most of the residual hum now present is very small, and is mainly due to stray magnetic fields acting on the phono pickup, FM tuner, or Magnecordette. The latter is particularly sensitive to stray 60-cps magnetic fields.

The 12AX7 heater is placed in series with a 22-ohm resistor and 25-ohm potentiometer. This increases the voltage drop to a value adequate to serve as self bias for the push-pull output stage. The 25-ohm potentiometer serves further to vary the bias of one 1622 tube relative to the other; in this way two 1622 tubes can be equalized as to d.c. plate current flow in the output stage. The 10,000-ohm resistor is required to draw additional bleeder current because the 12AX7 heater requires more d.c. than do the plates of the 6L6 tubes at an operating potential of 250 volts. The 10,000-ohm resistor also serves as a bleeder for the power supply.

Mixer and Equalizer Stages

The 50,000-ohm volume control following the preamplifier stages is for the phono pickup only, since it is assumed that the tuner and tape recorder have their own volume controls. In this way each volume control can be preset to its average position, so that there is no sudden blare of sound as the system is switched from one source to another.

The third stage employs the two sections of a 12AX7 tube in the form of an electronic mixer, as mentioned previously. The common plate resistor of 39,000 ohms was found to be the desirable value to use in conjunction with the bridged-T network following it. The latter employs a 2-henry Hycor toroidal inductor, which can be switched by two sections of a 3-gang switch to bridge various combinations of series capacitors.

The bridged-T configuration gives a sharp attenuation dip at the frequency at which the two capacitors and the in-

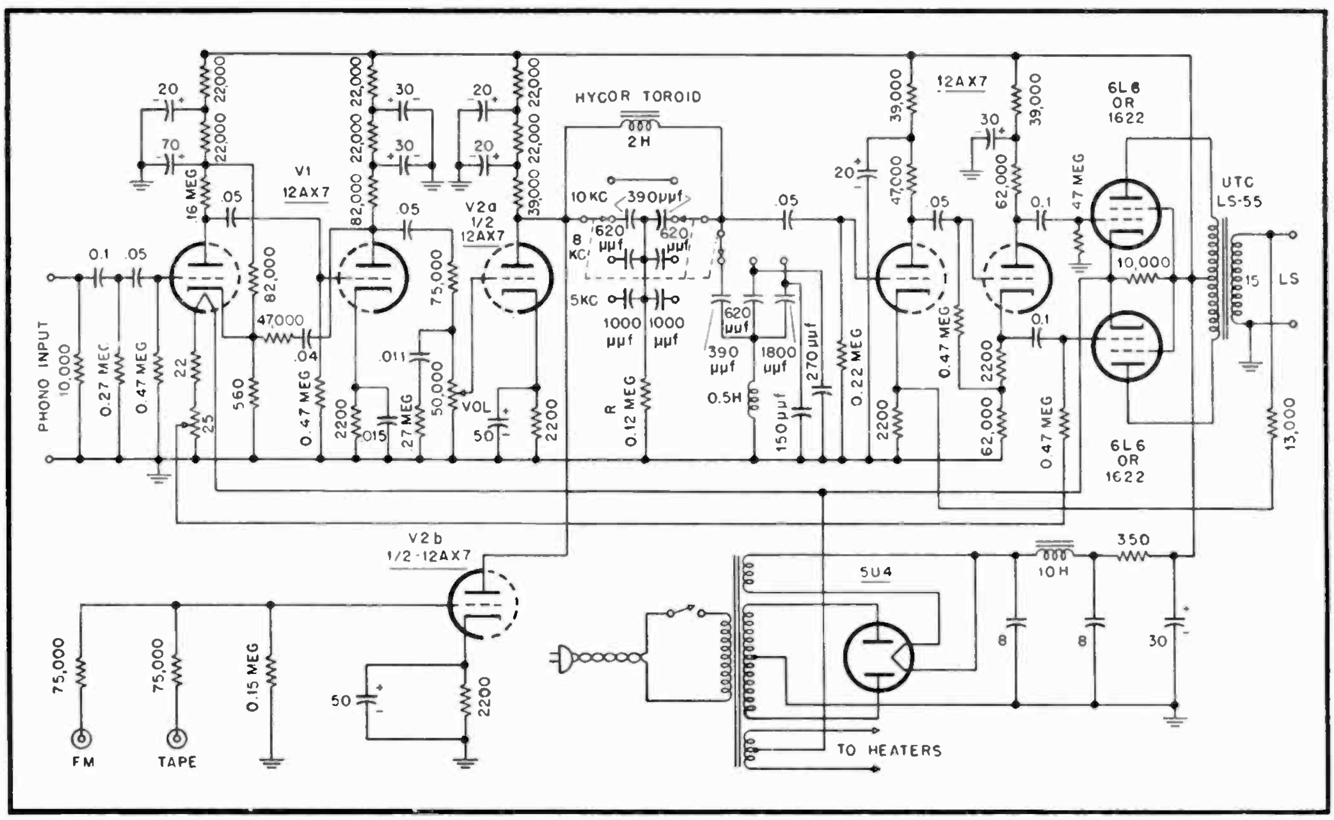

Fig. 1. Complete schematic of the amplifier. The input switch is eliminated by mixing all three sources so that it is only necessary to turn on tuner, phono, or tape machine as desired.

ductance are in series resonance. The value of R must be adjusted to the proper value to produce this sharp selective attenuation effect, but its value is not particularly critical. The magnitude indicated is 120,000 ohms, but this value would differ with different inductors.

Approximate values of the capacitors used are indicated, but exact values are best determined experimentally by running a frequency response curve on the amplifier and adjusting the capacitor values until the proper response is obtained. Actually, odd values may be required to hit the desired frequency, and in this case a practical dodge is to use slightly unequal values of capacitors. A little patience and a number of capacitors will result in the response dropping off fairly sharply at the desired cutoff frequency.

However, the bridged-T network attenuates only in the vicinity of one frequency; above that frequency its transmission approaches the normal value once more. To hold down the response beyond the attenuation frequency, the series-resonant network following the bridged-T is employed. This involves a 0.5-henry choke and suitable series capacitors, mounted on the third section of the 3-gang switch, to operate in conjunction with the bridged-T.

Approximate values are shown for these capacitors also, but here again the capacitors should be chosen experimentally to resonate at a suitable higher frequency than the corresponding frequency of maximum attenuation of the bridged-T, and thus hold down the response to a suitably low value at the higher frequencies.

However, for the 8000- and 5000-cps attenuation frequencies, the simple series resonant circuits permit the response to rise again beyond its resonant frequency, because the amplifier still has adequate gain in this region. Hence, two additional shunting capacitors are employed from the corresponding two switch contacts to ground, and these hold the response down to an acceptably low value.

The response curves of the filters are shown in *Fig.* 3. The peaks shown are due to a tendency for the bridged-T choke to resonate with the following

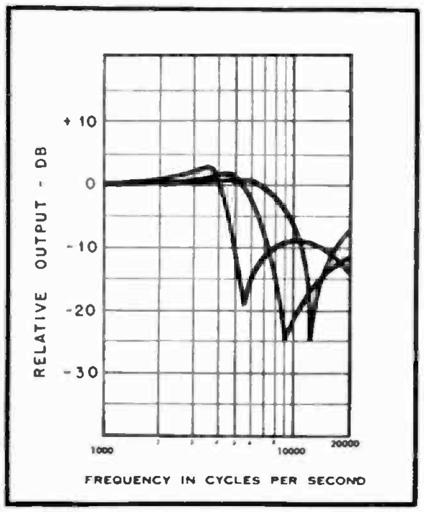

Fig. 3. This graph shows the characteristic with each of the three filters switched in.

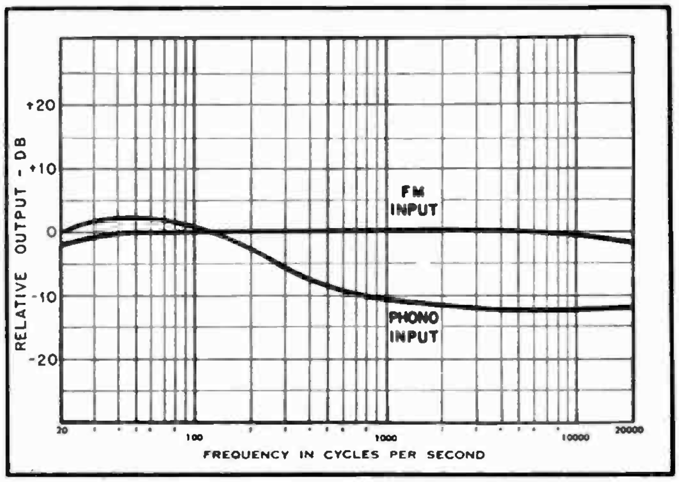

Fig. 2. Amplifier response from the pickup and tuner inputs to output.

series-resonant circuit; the rise in gain after the frequency of maximum attenuation is due to the inability of the series-resonant circuit to keep the response down. The results are quite satisfactory despite these peaks, and in spite of the grave admonitions to avoid peaks, the "ringing" of the system is not noticeable to the listener.

The main reason for employing such a rather involved network is to obtain as wide a frequency response as possible consistent with the surface noise of the record or other source. For LP's and even good 78's I use the amplifier wide open; I can take a certain amount of hiss. Incidentally, I seemed to get a sharper attenuation curve using the bridged-T insead of a parallel twin-T, which otherwise would have been more attractive because it requires only resistors and capacitors.

Power Output Section

The next two stages are sections of a 12AX7 tube, as indicated. The reason two sections are employed is to obtain feedback from the secondary of the output transformer to the cathode of the first section, and then to use the second section as the phase-splitter stage.

The amount of feedback that can be employed is limited; for the UTC LS-55 output transformer the values of 13,000 ohms for the feedback resistor, and 2200 ohms for the cathode resistor seem to permit the greatest allowable amount of feedback. Tests made by measuring the regulation of the output voltage as the load resistance is varied indicate that the internal output impedance, looking into the 15-ohm secondary terminals, is reduced to 4 ohms.

The split cathode resistor employed in the phase-splitter stage affords the correct grid bias for the stage and at the same time the correct drive voltage for the connected 1622 grid.

The output stage in itself is conventional. The 1622 tubes are operated at values recommended by RCA, 250 volts on the plates and screen grids, and −18 volts self bias. The output, as measured at the secondary of the output transformer, is 16 watts, and was determined by observing the output wave shape on a scope, and noting where it just began to flatten. The exact percentage is not important, as this is more power output than is required in the home.

The maximum gain was found to be about 102 db at 1000 cps. This is not only very high, considering that no input step-up transformer is employed, but it is also in excess of that required by any phonograph pickup; indeed, it reaches down into the thermal and microphonic noise levels of the first stage. This is particularly the case since the gain at low frequencies is an additional 18 db. A 12AU7 can be successfully substituted for the first 12AX7, with about 7 db reduction in gain. Hence, one might just as well use the 12AX7 and run the volume control a little lower.

The power supply is of conventional

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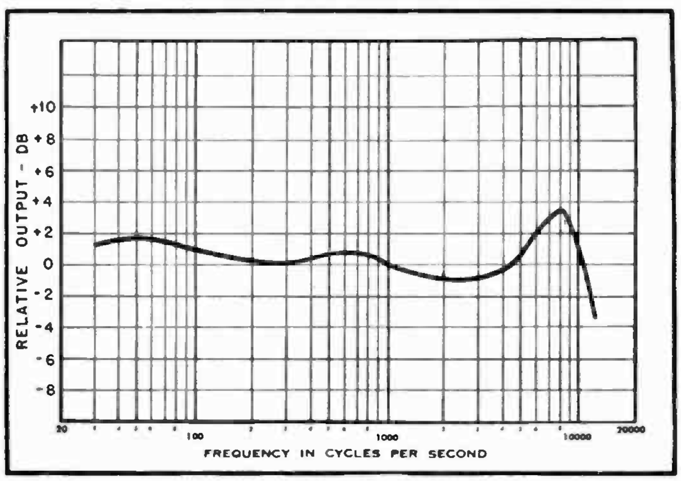

Fig. 4. Over-all response from the grooves of a test disc to a dummy load resistor following the output transformer is satisfactorily flat.

A Transformerless 25-Watt Amplifier for Conventional Loudspeakers

D. P. DICKIE, JR.* AND A. MACOVSKI**

A Low-Cost, High-Quality Amplifier Using No Iron-Cored Components

During the past several years audio amplifier design has progressed at a rate second only to that of the transducers associated with it in typical sound reproduction systems. A decade ago one could justifiably point to phonograph pickups and loudspeakers as the quality-determining links in the average home audio system. This disparity has been steadily narrowed and at the present time no one can generalize as to weak links. Suffice it to say that innovations in amplifiers have been less startling, since they had an initial jump on the rest of the elements involved in the sound reproduction process. Such advances as have been made center primarily about the output transformer. With minor exceptions, no basic circuit changes have found their way into commercially available amplifiers.

This attack on the problem was a logical one since only the transformers and the vacuum tubes in an amplifier can operate in a nonlinear fashion and thereby produce harmonic and intermodulation distortion. Furthermore, the fundamental limitations on bandwidth or frequency response have generally been due to the output transformer. One of the most practical ways to minimize harmonic distortion is to employ inverse feedback around those circuit elements that are responsible for the generation of distortion products. Again, the stumbling block has been the output transformer, for its high-frequency attenuation and phase shift characteristics have thus far limited the amount of inverse feedback which could be stably employed. This vicious circle has stimulated many inherent improvements in transformer design, but the fundamental problems still exist, although mitigated in magnitude. It is unfortunate but true that the region of most serious distortion in a transformer is in the low- and very-low-frequency range. At these frequencies the magnetizing current may become sufficiently high to produce saturation flux densities in the core. Although inverse feedback can substantially reduce the distortion in the near-saturation region, its application is dependent upon the high- as well as the low-frequency characteristics of the transformer. There is no simple solution to the problem and careful attention must be paid to the sometimes conflicting demands of good high- and low-frequency performance. It should be pointed out here that the problem is not merely confined to frequency response of the transformer. Most good output transformers exhibit a frequency response far wider than that needed for sound reproduction. The problem of being able to transfer large amounts of power without distortion, particularly at the low-frequency end of the range, is another issue altogether.

Still another limitation imposed by most output transformers in high-quality systems is inability to operate well in class-B and AB power-amplifier stages. Unless there is a very high co-efficient of coupling between the two halves of the primary, transient signals are generated by the nonsinusoidal currents which flow in the half-primaries. Class-B and AB operation can contribute greatly to the power handling capabilities of an amplifier stage, but unfortunately these classes of operation have become associated with higher distortion. While this is fundamentally true, the amount of distortion is not serious and if sufficient inverse feedback is employed the output signal will be a good replica of the input. Full realization of these more efficient operating conditions must await the practical application of large amounts of inverse feedback.

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The Transformerless Amplifier

With these problems of output transformers in mind many have envisaged transformerless amplifiers. While many of the problems associated with transformerless design seem overwhelming, at least one manufacturer has licked the biggest problem by winding a 500-ohm voice coil for his loudspeakers. Performance is almost unbelievable in those regions where transformer-type amplifiers fall down. It was felt by the writers that if outstanding performance could be obtained in a transformerless amplifier which could drive loudspeakers of conventional impedances, a very practical unit might be the result.

At the outset of study of the problem it was determined that any design should be a practical one. The use of transmitting-type tubes or inordinate quantities of receiving-type tubes was not justifiable. Plate efficiencies comparable to existing high-quality amplifiers should be achieved. Size, cost and weight should not exceed those of comparable amplifiers. Furthermore, it was felt that for a real contribution to be made, very exceptional operation should be the rule not only in the usual respects but particularly in those respects where transformer-type operation has its weaknesses. Since most high-quality loudspeakers are available in 16-ohm impedances, this amplifier was designed for that impedance. Following standard practice, the entire unit was designed for use with preamplifiers suitably equalized for the particular program source and capable of delivering 1 volt of signal.

Preliminary study yielded some startling results. It seemed that the unconventionality of the goal—that of driving a low-impedance speaker directly—ac-

[1] Chai Yeh, "Analysis of a single-ended push-pull audio amplifier," *Proc. IRE*, June 1953.

QUALITY AMPLIFIER

(Continued from previous page)

design and needs little comment. The reason for the dropping resistor as a second filter is that a spare television power transformer was used, which developed an unexpectedly high voltage. This was reduced to meet the rated plate and screen voltages of the 1622 tubes; perhaps the transformer you may have available will not require the same treatment.

Oil-filled filter capacitors of relatively low values were employed rather than electrolytics; however, the writer is not particularly prejudiced against them. There is, however, a satisfaction in knowing that these components will stand up indefinitely, although, admittedly, large electrolytic capacitors were used for decoupling purposes in the various stages, and hence in time trouble may occur at these points. Although smaller decoupling capacitors could be used, the values indicated do stabilize the amplifier to a large extent against line surges and similar "bumps."

This amplifier is operated into an LC1A speaker installed in an infinite baffle of about 15 cubic feet volume. The results are generally approved by people who have listened to the system, and it appears to be very suitable for the home.

As a matter of final interest, *Fig.* 4 shows the response of the system from the G-E pickup to a 15-ohm load resistor, employing a Dubbings D-101A record for the RCA New Orthophonic curve. The results, it will be noted, are fairly flat, particularly at the low-frequency end of the spectrum, and confirm the satisfactory conclusions drawn from listening tests.

tually set off a chain reaction of further innovations. While most of the features ultimately used are tried and true procedures in the electronic industry, this particular combination of them is new to audio. The infusion of new blood seems to produce a healthy new approach to an old problem.

Theory Of Operation

The output stage of the amplifier is the single-ended, push-pull type as shown in *Fig.* 1.[1] The quiescent current is equal in both sets of triodes, with no d.c. flowing through the load. The tubes are driven out of phase with the difference in current between the tubes flowing through the speaker load.

The most efficient method of utilizing this system is to bias the output tubes close to cutoff with the operation approaching that of class B. The usual objections to this mode of operation, such as switching transients and distortion, do not apply if no output transformer is used. Class-B operation provides maximum power output with minimum plate dissipation so that the peak current capabilites of tubes can effectively be used. A low-impedance power supply is necessary with this arrangement so that the supply voltage will not drop excessively when the maximum current is drawn and thereby reduce the maximum power output.

The power supplies used are half-wave selenium rectifier circuits developing + 140 v. and − 140 v. with respect to ground. These use large capacitors, with no additional filtering which would raise the power-supply impedance. To get higher voltages for low-level stages, additional selenium rectifiers are used in voltage adding arrangements to obtain 250 volts as shown in *Fig.* 1.

The absence of an output transformer allows 40 db of feedback to be applied by connecting the voice coil directly to the cathode of the phase splitter driver. Besides its distortion reduction characteristic, the application of feedback serves to reduce drastically the hum voltage which would otherwise be present. Since the gain within the feedback loop is essentially unity, an additional voltage amplifier is used, with separate feedback, to build up the input voltage to the voice-coil level.

Circuit Details

Figure 1 shows the final circuit of the amplifier. Three 6082 double triodes are used. They are 26.5-volt versions of the popular 6AS7G. These tubes are capable of 700 ma of peak current per triode section at the plate voltages used. The 6082's allow the use of a series heater string, eliminating a filament transformer. The tubes cut off at about 70 volts, and a 60-volt fixed bias is used on each. The bias on one side is made adjustable so as to equalize the d.c. in the two sets of triodes and insure that no d.c. flows in the voice coil. To protect the tubes, a low d.c. impedance of 56,000 ohms is used in the grid circuit.

It should be noted that the 6082 is not intended for use with fixed bias unless a limiting resistor is added in either the plate or the cathode circuit of the tube. Although this circuit does not use such resistors, their omission is feasible because the tubes are used under quiescent conditions well below maximum ratings.

The voltage-amplifier stages are all operated class A with conventional circuitry. A separate driver is needed for each side of the output tubes, since insufficient output is obtained from the phase splitter to drive the output tubes directly. One side of the phase splitter has a larger load than the other, since the input to the lower group of output tubes has the speaker impedance in the cathode. This causes degeneration and necessitates higher input than the upper group. In the first voltage amplifier, bias is obtained using unbypassed cathode resistors since the loss in gain can easily be tolerated. The phase-splitter driver, however, has fixed bias applied to its grid by dividing down from the − 140-volt bus, since maximum gain is desired within the main feedback loop.

The high-current power supplies use 300-μf capacitors and 5-ohm protective resistors (each made up of paralleled 10-ohm units). A voltage adder circuit is used for the − 250-volt supply, which supplies only bias. Another voltage adder circuit is used to supply + 250 volts for the driver stages. Additional R-C decoupling is used to minimize hum in the low-level stages.

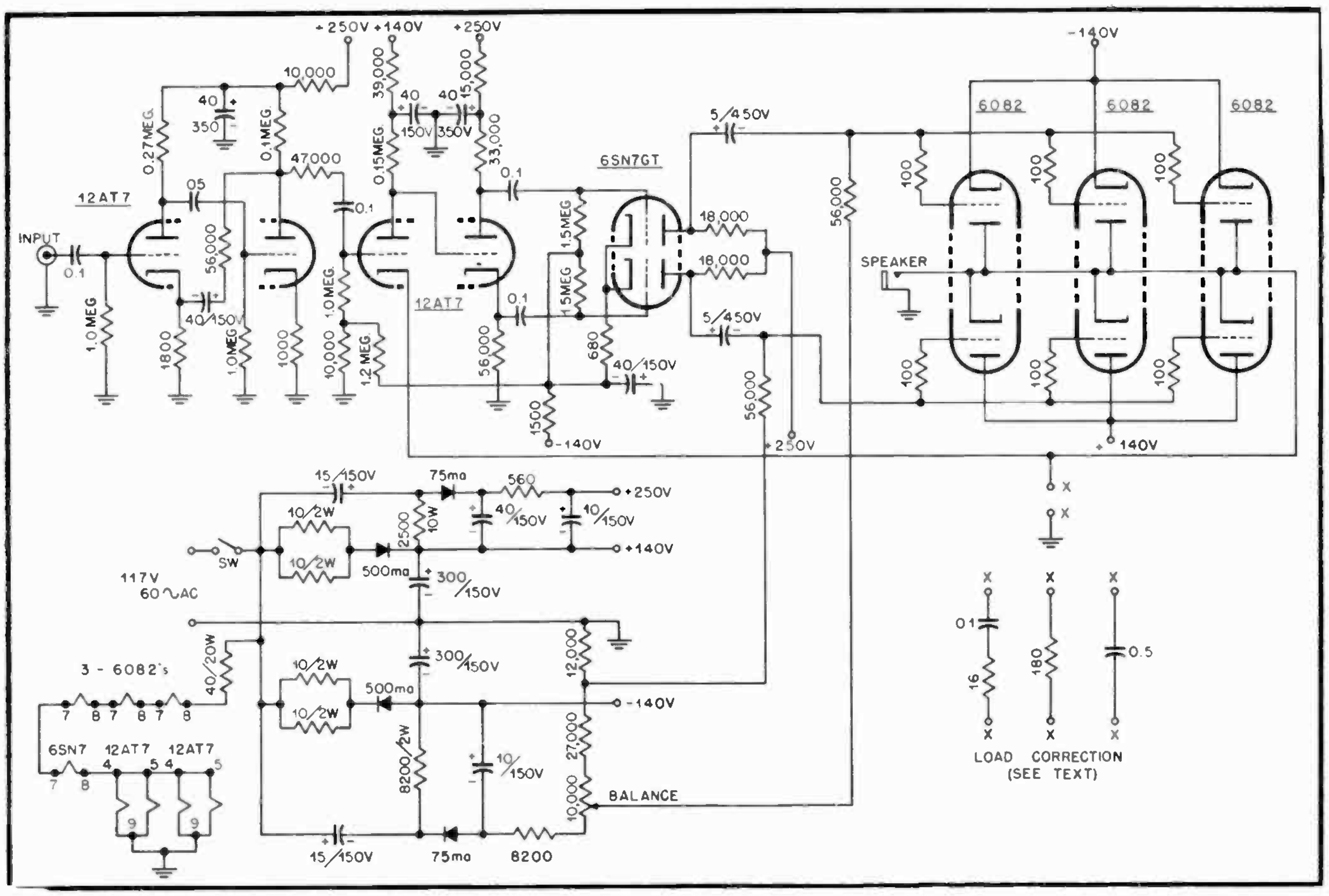

Fig. 1. This is the complete schematic diagram of the transformerless amplifier. Its total cost is comparable to that of a single high-quality output transformer. For safety, use of a 1-to-1 isolation transformer is recommended between the a.c. line and the power input terminals.

Operational Details

As with all power-transformerless equipment, care must be used when connecting to other pieces of equipment to see that the cold side of the line is connected to the chassis. Although this is readily achieved, the use of a small power line isolation transformer would eliminate the need for caution.

As would be expected, 40 db of feedback can be applied only within a loop having a minimum of phase shifts to avoid instability. It is therefore necessary to modify the arrangement when using speakers which present other than a resistive load to the amplifier. This situation is not generally encountered in transformer-type amplifiers since the output transformer itself becomes the main impedance at high frequencies. That is, the speaker high-frequency impedance is not reflected through the transformer. *Figure* 1 shows three alternative ways to deal with instability due to an inductive speaker load, which not only causes additional phase shift, but causes higher amounts of feedback due to the increased load impedance. The 180-ohm resistor merely limits the maximum impedance of the speaker and thus prevents excessive feedback. The 0.5-μf capacitor is a low impedance at high frequencies, shorting the inductive load. The series 16-ohm resistor and .01-μf capacitor places a 16-ohm resistor across the speaker at high frequencies and an open circuit at low frequencies. This serves to provide constant impedance and feedback over the frequency range. If any instability is noted with a given speaker, try one or the other for best operation.

The balance adjustment for zero d.c. in the voice coil can be made with a milliammeter in series with the voice coil (a closed-circuit jack might be added for the purpose). It should be repeated when tubes are changed or after many months of operation.

Performance

The operational results of the prototype model of this amplifier are shown by the curves and photographs.

The frequency response, which is shown in *Fig*. 2, is flat over a very wide range. Since resistive-capacitance-coupled circuits are used throughout, there is no serious limitation on response. To keep circuit complexity down, and to achieve the best feedback stability, a reasonable amount of gain per stage is desired. This, of course, will determine the high-frequency limitations, while the interstage coupling networks determine the low-frequencies limitations. While the bandwidth without feedback would be wide in this design, the use of 40 db of inverse feedback extends the ends of the range manyfold. Most good amplifiers exhibit a flat response well beyond the limits of audibilty, and this unit probably ranks as one of the widest-band designs intended for audio use.

As shown in *Fig*. 3, the harmonic distortion even at full rated output is exceptionally low and virtually independent of frequency. The ability to deliver 25 watts at 20 cycles and below with negligible distortion is practically impossible in a transformer-type amplifier of similar mid-frequency power rating. The low-frequency performance is directly attributable to the use of circuit components whose nonlinear properties are in no way dependent upon

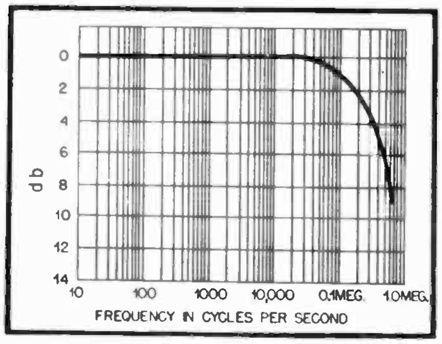

Fig. 2. Frequency response of this amplifier is flat at the low end and less than 3 db down at 200,000 cps, making it one of the widest-band designs in use today for audio.

frequency. Thus the 40 db of feedback remains at that value over the entire usable frequency range and satisfactorily reduces the distortion without regard to the frequencies involved. Even if the distortion in the amplifier without feedback were as high as 10 per cent, the 40-db figure, corresponding to a voltage ratio of 100 to one, would reduce this to 0.1 per cent.

While dealing with the tested results it is worthwhile to mention the subject of intermodulation distortion. IM is a very good and rapid means of evaluating distortion in an amplifier. It is, however, at its greatest value in testing systems where there is apt to be a marked dissimilarity in the nonlinear performance of the amplifier at the particular pair of frequencies used in the test. In an amplifier such as this where there is no frequency-sensitive distortion characteristic, IM testing would yield little to the total fund of information.

The efficiency with which a 16-ohm loudspeaker may be driven directly to produce these large power outputs is due not only to the extremely high perveance of the output tubes but also to operation aproaching class B. Sufficient quiescent current flows in each tube to ensure good small-signal linearity. That is, the operating bias is sufficiently low to ensure that the no-signal operating point is outside the curved region of the tube characteristic near plate-current cutoff. The resulting efficiency is about the same as a transformer-type class-A amplifier.

The square-wave performance of an amplifier is an indication of its ability to reproduce signals of a transient nature. The low-frequency square-wave response is a measure of the ability to reproduce extremely low frequencies. The amount of "droop" in the square-wave response is the important feature in this respect. A negligible droop at a particular square-wave frequency means that the amplifier will reproduce well down to frequencies which are only a fraction of the fundamental. The loudspeaker damping, however, will probably be the ultimate factor in low-frequency transient performance. While it is impossible to secure unlimited amounts of damping through reduction of the output impedance of the amplifier, it is im-

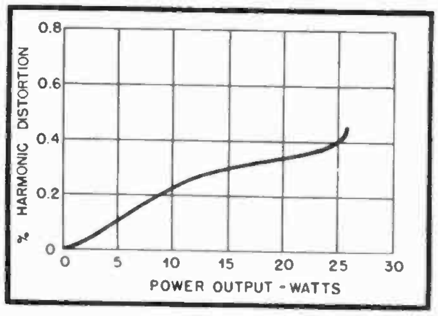

Fig. 3. Harmonic distortion is less than 0.4 per cent at the related output of 20 watts into a 16-ohm speaker. Intermodulation measurements would mean little since there are no frequency-sensitive nonlinear elements in the design.

portant that this internal impedance be at least several times lower than the speaker's nominal impedance. Reductions of amplifier internal impedance beyond this point are not necessary but can do no harm. The internal impedance of this amplifier, due to the large amount of feedback, is only a fraction of an ohm. This provides excellent electrical damping.

The high-frequency square-wave performance is a good indication, not only of transient response, but also of the stability of the feedback system. A

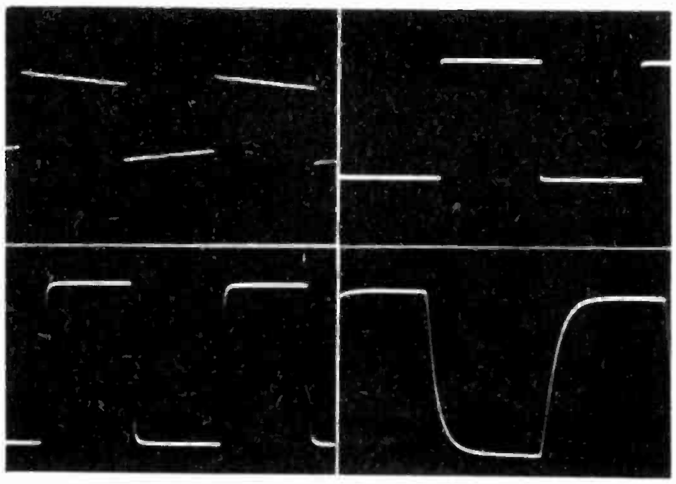

Fig. 4. These photos of oscilloscope traces show square-wave response at four frequencies—top row, left to right, 20 and 1000 cps; bottom row, 10,000 and 50,000 cps. Note the unusually short rise time even at 10,000 cps and the complete absence of evidences of instability despite a full 40 db of feedback. Effective output impedance is a fraction of an ohm, eliminating any tendency toward overshoot on transients.

tendency towards instability (not necessarily oscillation) will manifest itself as a decaying train of oscillations following the rapid rise and fall of the square wave. Of course, if these oscillating overshoots do not decay or die out, the system will oscillate continuously. It is important to have ample stability in the feedback system to prevent any form of "ringing" or overshoot on a rapidly rising square wave. The actual rise time of the square wave itself in the output is largely a measure of the high-frequency response.

Several square-wave frequencies are shown in *Fig.* 4. Although the highest frequency shown (50 kc) is only of academic interest, it was included to indicate that the performance was not greatly deteriorated at a high frequency ordinarily used to test video amplifiers.

The extremely light weight and small size as well as the low cost of this amplifier stem principally from the absence of heavy and costly components. The only items of major expense are the tubes, the four selenium rectifiers, and the two large electrolytic capacitors. The remaining components are small resistors and capacitors of the sort encountered in most amplifiers. The three output tubes are considerably more expensive than the normal receiving type, but are still reasonably priced. The other tubes are of widely used variety and inexpensive. Only two of the selenium rectifiers are the 500-ma type, while the other two are the small 75-ma variety. The total cost, computed from the catalog of a large parts supplier, is approximately the same as that of a single high-quality output transformer of the type originally designed for a currently popular amplifier.

The hum level in this amplifier is about .02 volt across the load. This is 60 db below the rated power output. This value was found to be exceeded by the extraneous hum of the supplied source material from the preamplifier, which in itself was acceptably low.

The qualitative results of an amplifier are generally the results of listening tests. While it is impossible to attribute specific attributes of good reproduction to particular links in the system merely by listening, some indication of performance may be had by comparative methods. As one would suspect from the features, the way in which this design excels is when handling large amounts of power at low frequencies. Even at moderate listening levels, an exceptional clearness of reproduction was noted on organ music. An RCA LC-1A in its standard studio console phase-inverter cabinet was used for listening tests. It was neither feasible nor desirable to employ anything near the full output of this amplifier with this loudspeaker, but at reasonably high room levels, the low-frequency reproduction seemed exceptionally smooth and realistic. The use of a horn-type low-frequency loudspeaker, which would more efficiently load the cone, would permit the use of higher and more realistic levels of, say, a pipe organ. While this has not been tried, it is thought that here is where the amplifier would excel. Since it is capable of delivering large amounts of low-frequency power, low-frequency signals such as those developed when tuning through an FM signal may cause excessive cone excursions in speakers which are inadequately coupled to the air. If such is the case, the coupling capacitor between the first and second 12AT7's may be reduced to attenuate these effects.

Parts List

Capacitors

1 .05 µf, 600 v. paper
4 0.1 µf, 600 v. paper
2 10 µf, 150 v. electrolytic
2 15 µf, 150 v. electrolytic
3 40 µf, 150 v. electrolytic
2 300 µf, 150 v. electrolytic
3 40 µf, 350 v. electrolytic
2 5 µf, 450 v. electrolytic

Resistors

1-watt

6 100 ohms	1 39,000 ohms
1 680 ohms	1 47,000 ohms
1 1000 ohms	4 56,000 ohms
1 1500 ohms	1 0.1 meg
1 1800 ohms	1 0.15 meg
2 10,000 ohms	1 0.27 meg
1 15,000 ohms	3 1.0 meg
2 18,000 ohms	1 1.2 meg
1 33,000 ohms	2 1.5 meg

2-watt

4 10 ohms	1 12,000 ohms
1 560 ohms	1 27,000 ohms
2 8200 ohms	

10-watt	20-watt
1 2500 ohms	1 40 ohms

1 10,000-ohm wirewound potentiometer
2 75-ma, 130-volt selenium rectifiers
2 500-ma, 130-volt selenium rectifiers
1 6SN7
2 12AT7's
3 6082's

The White POWRTRON Amplifier

STANLEY WHITE*

A discussion of one possible cause of power distortion and a description of a circuit developed to eliminate it. The author also describes his method of dividing the frequency spectrum ahead of the power amplifier. This unit has been popular with listeners at recent demonstrations.

Most amplifiers are developed and tested using pure resistive load impedances across the secondary of the output transformer. Determination of intermodulation distortion, harmonic distortion, and power performance are based upon results obtained using these resistive loads although it is well recognized that speakers do not present a constant load impedance over the entire frequency spectrum. However, for want of a better method, resistive loads have been retained as a standard procedure in determining the performance and operating characteristics of amplifiers.

This paper proposes a basic change in amplifier circuitry that is inevitable if amplifiers are to perform their basic function—that of presenting an electrical power waveform to a speaker in such a manner that the acoustical wave radiated from the surface of the speaker is a transformed replica of the voltage waveform at the input to the audio amplifier. It will be shown that present audio amplifiers create a power distortion of a magnitude of 6 to 8 db, and this type of distortion is not discernible by present day testing procedures.

Definition 1. Power distortion: A power waveform generated by an audio amplifier that deviates in any manner whatsoever from the form of the input voltage waveform is distorted with respect to power to the extent of the deviation.

From this definition, it can be seen that any distortion measurement of an audio power amplifier is, in fact, a measurement of power distortion. That is, power distortion is a generalized form covering intermodulation distortion, harmonic distortion, and so on. Any amplifier that changes its power output with changing load impedance suffers from power distortion to the extent that the power output is altered. It is recognized that the relationship

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between power, voltage, and impedance can be expressed by the formula

$$P=\frac{E^2}{Z} \qquad (1)$$

where P = power output,
E = voltage, and
Z = load impedance.

In test procedures using resistive loads, it can be seen that if E remains constant, the power output will remain constant. However, with variable load impedances the power output will bear an inverse relationship to the impedance.

From transducer theory, there are certain relationships between the electrical and acoustical characteristics of any speaker, and such factors as the resistance of the suspension system, the resistance of the air load, the reactance of the voice coil and cone, the reactance of the air load, and the reactance of the suspension system must be considered as affecting the total impedance of the speaker, in addition to the pure electrical impedance of the voice coil itself.

Effect of Feedback

The majority of hi-fi amplifiers employ some form of voltage feedback, but a study of equation (*1*) will show that if voltage remains constant there will be considerable power distortion, and it is agreed that voltage feedback tends to hold the voltage constant regardless of the load across the amplifier terminals. Thus any change in load impedance results simultaneously in an inverse power change. If electrical impedance characteristics and acoustical output characteristics of a given speaker were related in such a manner that electrical impedance peaks occurred simultaneously with acoustical peaks, the decrease in power response at the point of maximum acoustical output would be beneficial. However, in real speakers this condition seldom occurs.

The Powrtron circuit, *Fig.* 3, differs from conventional amplifiers in that it adds a small amount of negative current feedback to a usual amount of negative voltage feedback, with the result that over a reasonable range of load variations the power distortion is held to 1 db, whereas without the Powrtron feature the same amplifier shows a distortion of as much as 8 db.

Careful consideration of this will show that it is useless to attempt to control the behavior of a loudspeaker by means of a device that will sense impedance changes in the speaker, and this is exactly what is done with voltage feedback. Many other effects of voltage feedback are definitely beneficial, as is well known, but the effect on power distortion is to increase instead of decrease it.

Negative power feedback results in much less power change over a range of output loads than the other methods of operation. Positive current feedback reduces the internal impedance of power amplifiers to zero, but by so doing it increases power distortion.

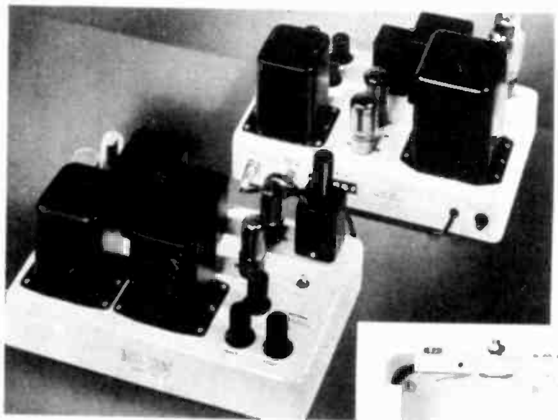

Fig. 1 (left). Top view of 10- and 20-watt White amplifiers with filter network plugged into the 10-watt unit. Fig. 3 provides for network to be plugged into the 20-watt low-frequency amplifier.

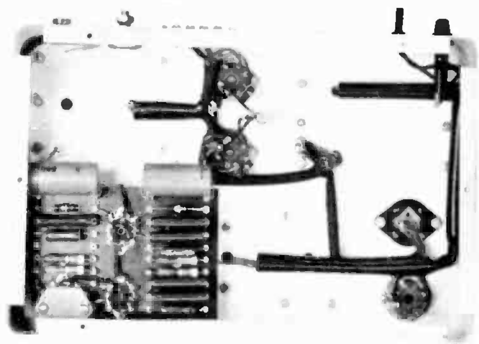

Fig. 2 (right). Underside view of the 20-watt amplifier. Large mica capacitor at lower left is C_1.

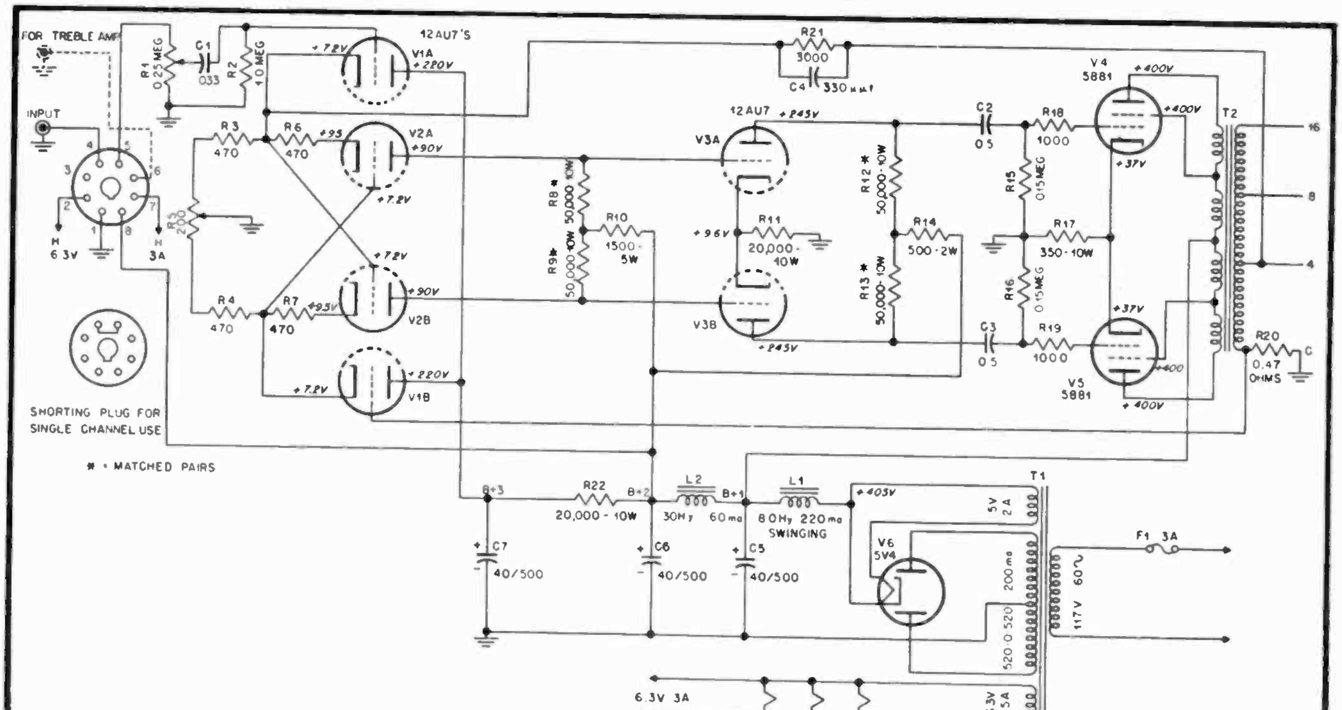

Fig. 3. Complete schematic of the 20-watt White amplifier arranged for plugging in the electronic filter network. 10-watt model is identical except for output tubes, which are 6V6's, and the output transformer.

The Complete Circuit

While the Powrtron circuit refers only to the addition of a single resistor in the output circuit and the connection back to a suitable point for the introduction of feedback, there are some advantages to the complete White amplifier and the method of introducing two separate kinds of feedback is simplified greatly. In *Fig.* 3 it will be noted that R_{21} and C_4 constitute a usual form of negative voltage feedback. The negative current feedback is obtained from R_{20} in the return leg of the secondary of the output transformer. The cross-coupled phase inverter, together with the direct-coupled driver stage make it possible to introduce the two different types of feedback with considerable ease. Furthermore, if a direct A-B test is desired, it is only necessary to short out R_{20}.

Since the circuit is somewhat unique, it may bear explanation. The input is fed into a level-adjusting potentiometer and thence to the grid of V_{1a} through C_1 and the grid resistor R_2. (The use of the octal socket will be described later.) C_1 and R_1 may appear unnecessary, but the slightest amount of d.c. on the grid of V_{1a} is sufficient to unbalance the operation of the entire system so C_1 is a mica capacitor—.033 μf or larger—which has been found to be completely free from leakage. The cathode of V_{1a} is directly coupled to the grid of V_{2b} and a tap on the cathode resistor string of V_{2a}. R_5 provides for a balance of d.c. voltages throughout the first three tubes—the method of adjustment being to set R_5 at a point where the voltage between the plates of V_{3a} and V_{3b} is zero. The negative current feedback is connected to the grid of V_{1b}—directly out of phase with the input section—and the output of V_{1b} is fed into the phase splitter in a manner similar to that from V_{1a}. The direct coupling between the phase splitter section and the driver is made possible by the use of a very large cathode resistor for V_3. It will be noted that these cathodes are about 96 volts above ground, resulting in a potential of approximately 90 volts on the plates of V_2—this same voltage being applied to the grids of V_3 which results in a bias of around 6 volts.

The output stage is the Ultra-Linear, which has been described heretofore.[1] In the 20-watt White amplifier, 5881's are used; in a very similar design for 10 watts output, 6V6's are used—this latter amplifier being used with the 20-watt model to make the two-way amplifier system to be described.

The current feedback is developed across R_{20}, shown as 0.47 ohms. In construction, it is suggested that this value be obtained by the use of a 1-ohm 10-watt adjustable resistor. Slight variations in the power response characteristics may be had by changing the value of this resistor, with corresponding changes in the tonal quality of the output.

Figure 2 shows the underside of the White amplifier. Note that most of the components ahead of the output stage are located on the terminal board, which is laid out as in *Fig.* 4. The parts list indicates the wattages of the various resistors, as well as the types recommended.

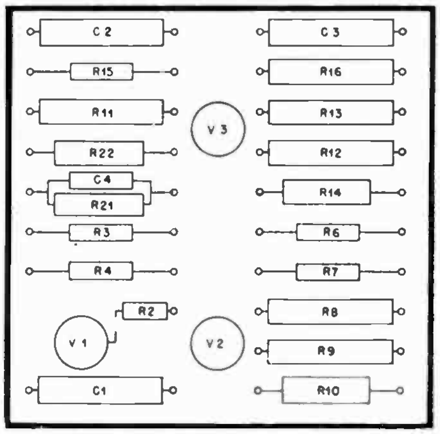

Fig. 4. Arrangement of parts on terminal board shown in Fig. 2.

In construction, it is suggested that the amplifier be assembled with semi-permanent connections between the driver stage and the output-tube grids, and with the negative-voltage feedback circuit—R_{21}-C_4—disconnected. Then pass a signal through the amplifier and note whether the signal increases or decreases when R_{20} is shorted. If the signal decreases, the leads to the two output grids should be reversed, since the feedback voltage developed across R_{20} should reduce the gain, and shorting the resistor eliminates the feedback. After the correct polarity is determined, the voltage-feedback circuit R_{21}-C_4 may be connected.

The Octal Socket

The octal socket previously mentioned provides for the insertion of an electronic dividing network ahead of the power amplifiers. With the shorting plug in place, the amplifier functions normally, and may be used to feed a single speaker, or to feed a two- or three-way system with a conventional dividing network. However, one of the advantages of the White system is that the dividing network is used ahead of the amplifiers, providing the advantage of low source impedance for the speakers. The principal disadvantage is the need for two power amplifiers, it being quite usual to use the 20-watt model for low frequencies and the 10-watt model for high frequencies.

The shorting plug simply connects the incoming signal to the input of the amplifier. However, when it is desired to use two amplifiers, the shorting plug is removed and an electronic filter unit is inserted in the socket. *Figure* 5 is the schematic of the filter network. which consists of a dual triode connected as two cathode followers. Each follower feeds a filter circuit—one of low-pass configuration, and one of high-pass configuration. The low-pass output is fed to the associated amplifier, and the other output is fed to the treble amplifier. In the commercially available model, the treble output is fed through a pigtail cable, which is plugged into the second amplifier. As shown in *Fig.* 3 the treble output is channeled to another phono jack, which is connected by a jumper to the second amplifier. The terminals shown are not those used in the commercial version, but are indicated for study of the circuit.

Filter-Network Advantages

The most recent trend in amplifier design has been toward increased negative feedback, using output transformers of wider and wider range and placing more and more stages inside the feedback loop. For optimum operation, all of the push-pull stages should be balanced, and maximum phase shift must be kept to less than 180 deg. inside the feedback loop if oscillation is to be avoided.

The two regions in which phase shift will occur and oscillation becomes a problem are at the extreme ends of the audio spectrum. The ideal way to design an amplifier is to keep the phase shift through the electronic section of the amplifier limited to less than 5 deg. and allow the electrical characteristics of the output transformer determine the operating frequency of the amplifier. Unfortunately this ideal is seldom achieved.

It is well known that the reactive

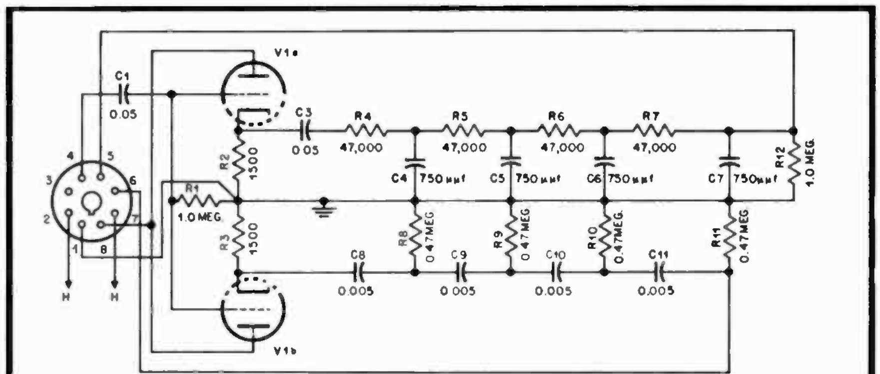

Fig. 5. Electronic filter network using a single 12AU7 as two cathode-follower sections to drive the R-C filters which comprise the dividing network ahead of the power amplifiers.

[1] David Hafler and Herbert I. Keroes, "The Ultra-Linear amplifier." AUDIO ENGINEERING, Nov. 1951.

The Maestro—a POWER Amplifier

DAVID SARSER[*] and MELVIN C. SPRINKLE[**]

A new version of the now famous Musician's Amplifier which should satisfy anyone's desires for more power—and which uses a newly developed tube type with modest plate supply requirements.

Noah Webster, in his book of words, defines maestro as "a master in any art, especially music." This name is particularly appropriate to this amplifier, shown in *Fig.* 1, for it combines the best properties of the now famous Musician's Amplifier with a prodigious increase in power output. It is truly the master of the art of recreating music by electronic means.

The success of the Musician's Amplifier[1] is too well known to require repeating, but certain specialized applications have been encountered in which it did not fill the bill. We have in mind its power output, for its response, low distortion, and low noise level leave little to be desired for home music listening.

One application for which it is not entirely adequate is as a driver for a disc recording head. The low distortion makes the Musician's Amplifier attractive, but it falls short on power, especially when making LP discs where pre-emphasis is required. The considerations on power for disc recording are well known and have been mentioned by these writers previously.[2]

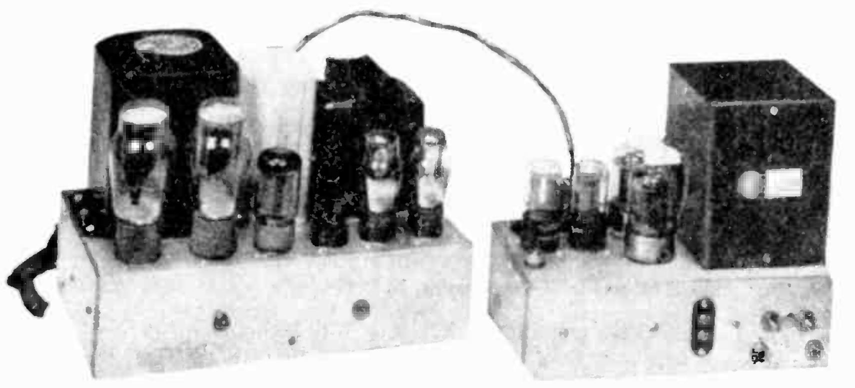

Fig. 1. The Maestro amplifier—a new contender for high-quality sound reproduction in the home, or for disc-recording cutter driving, or for any application where up to 90 watts is required.

The development of FM broadcasting, modern LP records, and tape equipment has set new standards for dynamic range in reproduced music. It is now necessary to re-appraise the power required for critical listening. In the past, the program material was compressed to a 35- or 40-db range and maximum power could be handled easily by the conventional "15-watt" amplifier. Today's trend is toward elimination of compression. Therefore it is necessary to increase the power delivery of the amplifying system.

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** Sales Engineer, Ampex Corporation, Silver Spring, Md.

[1] Sarser and Sprinkle, "Musican's amplifier," Audio Engineering, Nov. 1949.

[2] Sarser and Sprinkle, "Musician's amplifier senior," Audio Engineering, Jan. 1951.

The POWRTRON

(*Continued from previous page*)

filter networks cause substantial distortion in the process of sound reproduction. However, the manner in which it is caused is not nearly as well known. The design of filter sections of constant-valued elements of resistance, capacitance, and inductance is standard engineering practice. However, the design of filter sections capable of dealing with the variable impedance presented by a speaker is a problem of serious magnitude. The use of a dual-channel amplifier using electronic filter sections at the input of the system is deemed the best solution.

The crossover filter is constructed in a standard Vector CO-10-N turret can, making it readily interchangeable. Thus the experimenter can construct several different filter networks to determine the best operating crossover frequency for the speakers used, or by removing the network can restore the amplifier to normal operation with a minimum of effort. For the constants shown, the crossover frequency is approximately 560 cps.

In the hi-fi field the final judgement is always that of the listening test. In the case of amplifiers it is difficult to achieve a distinct improvement, but it is felt that a listening test with the crossover amplifier will give the listener just such a distinct improvement.

Parts List (Fig. 3)

C_1	.033 μf, 1200 v. mica
C_2, C_3	0.5 μf, 600 v. paper
C_4	330 μμf, 500 v. mica
C_5, C_6, C_7	40 μf, 500 v. elect.
L_1	8 H, 220 ma, swinging choke
L_2	30 H, 60 ma, smoothing choke
R_1	0.25 meg potentiometer, audio taper
R_2	1.0 meg, ½-watt, deposited carbon
R_3, R_4, R_5, R_7	470 ohms, 1-watt, wirewound
R_6	200 ohms, 4-watt potentiometer, linear
R_8, R_9	50,000 ohms, 10-watt, wirewound, matched pair
R_{10}	1500 ohms, 5-watt, wirewound
R_{11}	20,000 ohms, 10-watt, wirewound
R_{12}, R_{13}	50,000 ohms, 10-watt, wirewound, matched pair
R_{14}	500 ohms, 2-watt, wirewound
R_{15}, R_{16}	0.15 meg, 1-watt, deposited carbon
R_{17}	350 ohms, 10-watt, wirewound
R_{18}, R_{19}	1000 ohms, ½-watt, deposited carbon
R_{20}	1.0 ohms, 10-watt, adjustable, wirewound
R_{21}	3000 ohms, 1-watt, wirewound
R_{22}	20,000 ohms, 10-watt, wirewound
T_1	Power transformer, White Sound or Chicago PCR-200. 520-0-520 v at 200 ma; 5.0 v at 2.0 a; 6.3 v at 4.5 a; potted.
T_2	Ultra-Linear output transformer, Acro TO-300, or White Sound
V_1, V_2, V_3	12AU7
V_4, V_5	5881 or KT66
V_6	5V4

Parts List (Fig. 5)

C_1, C_3	.05 μf, 600 v. paper
C_4, C_5, C_6, C_7	750 μμf, 500 v. mica
C_8, C_9, C_{10}, C_{11}	.005 μf, 500 v. mica
R_1, R_{12}	1.0 meg, ½-watt, deposited carbon
R_2, R_3	1500 ohms, 10-watt, wirewound
R_4, R_5, R_6, R_7	47,000 ohms, ½-watt, deposited carbon
R_8, R_9, R_{10}, R_{11}	0.47 meg, ½-watt, deposited carbon
V_1	12AU7

A typical example is in a recent recording of Ponchielli's "Dance of the Hours." In this selection, the pianissimo 'cello solo passage is repeatedly interrupted by a crashing chord played by the entire orchestra. With the usual 10- to 15-watt amplifier, the chord is heard, but without sufficient definition to suit the fastidious listener. In order to distinguish between the various choirs of the orchestra playing this chord, which the trained ear can do in a concert hall, it is necessary that considerable power be available. A measurement of the peak produced by the chord shows around 22 db of change in instantaneous power. This is not, however, a true measure of the peak but is an integrated reading. This means that an amplifier of around 100 watts is required. Since this chord contains fundamental frequencies between 30 and 4000 cps, it may be seen that full power is required at these frequencies. In addition to power over this range, "clean" power is required up to at least 15,000 cps for disc recording as considered previously. Hence, we have looked toward the development of an amplifier which would combine the low distortion, low noise, and wide range of the Musician's Amplifier, with substantially increased power output.

While the Musician's Amplifier Senior[2] was a step in the right direction, it had several shortcomings: it is large in size; it requires a power supply much like a transmitter, and which can be lethal; it requires a power amplifier as a driver; and it is like all Class A amplifiers—inefficient. And in high-power amplifiers, efficiency becomes important.

New Tube Gives Clue

The recent announcement of the type 6146 by RCA pointed toward a solution of the need for more power with comparatively simple circuit design. This tube is a beam-power amplifier tube primarily intended for transmitter use. As shown in *Fig.* 2 in comparison with the 5881 and the KT-66, it is small in size, sturdily constructed; and it has a high power sensitivity. It can be used in a number of transmitter applications, but RCA's data sheet indicates that it will also serve as an audio power amplifier or modulator, Class AB. This data sheet recommends—under ideal conditions such as perfectly regulated power supplies—that a pair of 6146's be operated with a plate voltage of 750 and a screen voltage of 200. This requires a fixed bias of 50 volts and a plate-to-plate load of 8000 ohms. Under these conditions, the power output is approximately 120 watts into a plate-to-plate resistor. As a practical matter, we have departed slightly from these conditions and obtained a sine-wave power output of 90 watts from 25 to 30,000 cps. All this and Class AB, too, with no driver and no grid-current problems. The 6146 can be operated readily with resistance coupling from a voltage amplifier—and thus may be said to be a "jolly good bottle."

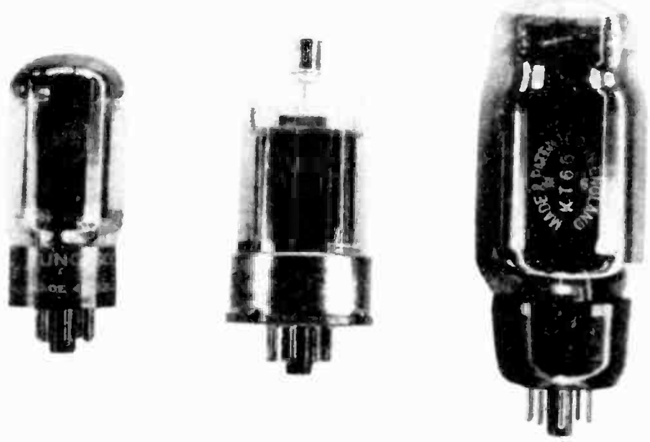

Fig. 2. Comparative size of the new RCA 6146 alongside the 5881 and the KT-66.

Having found a satisfactory tube type, the next problem was to find a suitable output transformer. Search of transformer catalogs failed to reveal one which would meet all requirements, so a conference was held with E. B. Harrison, of Peerless. On hearing the problem he said, "I think I can do it." Subsequently he has admitted it was a tough one. However, Harrison designed and built an output transformer for the 6146, and although originally built especially for this first amplifier, it is now in the Peerless line as type S-268-Q. When tested in a matched network, the response is within 1 db from 10 cps to 100,000 cps. Primary impedance is 8000 ohms, and it will handle 50 watts at 20 cps, and at least 80 watts mid-range. When used in a feedback amplifier where the source impedance is 10 per cent or less of the reflected primary impedance, the transformer will deliver close to 80 watts with no visible distortion at 20 cps. Primary inductance at 5 volts, 60 cps, is greater than 200 henries, while at 80 watts the inductance is aproximpately 800 henries, yet the leakage inductance referred to the primary is around 7 mh. The d.c. resistance of the primary is 115 ohms, and the insertion loss around 7 per cent. Small in size for its power rating this transformer has proved to be excellent in performance, and will pass a 30,000 cps square wave with a vertical rise and a flat top.

The Voltage Amplifier

Large triodes like the 845 have a high bias, and transformer coupling is almost a necessity. A power amplifier of some size is also required to produce the necessary voltage. The 6146, in common with other beam tubes, operates at a reasonable bias of 50 volts. It requires around 35 volts r.m.s. per tube, or 70 volts for a push-pull pair for grid excitation, and this is quite in line with the 807 or 5881 drive requirements in the Musician's Amplifier. Thus, the voltage amplifier of the earlier amplifier was adopted without change, as is observed from the schematic, *Fig.* 3.

Design of the power supply proved to be a bigger job. In the Musician's Amplifier Senior, the power supply resembled that of a small transmitter, and the problem was current capacity and high voltage. In the Maestro amplifier, the problem is regulation, since operation is Class AB. According to the data sheet, the plate current for a pair of tubes goes from a quiescent 57 ma to a peak of 227 ma, while the screen current changes from 1 ma quiescent to 27 ma at 120 watts. Another problem was to obtain the 750 volts with the choke input that good regulation dictates. One

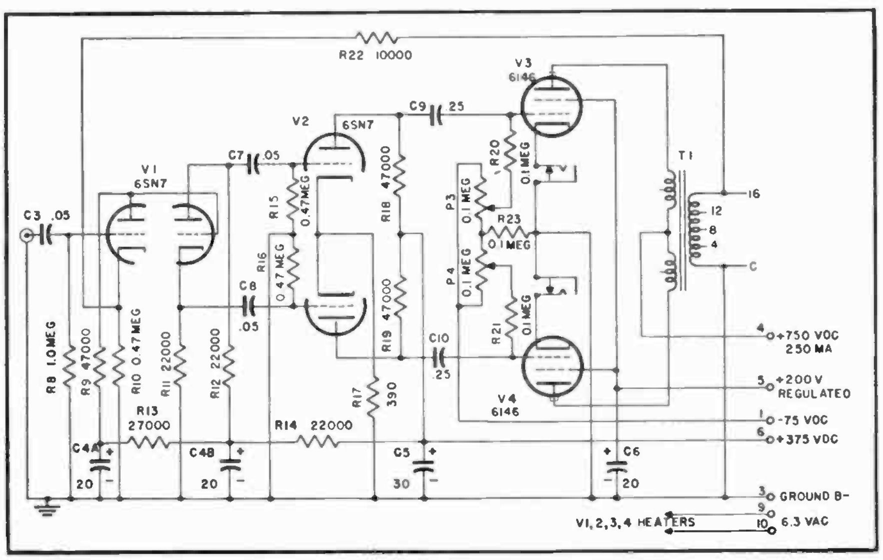

Fig. 3. Complete schematic of the Maestro. Note similarity to the Musician's amplifier.

solution was found by using two receiver-type transformers with the high-voltage windings in series. The primaries are paralleled across 117 volts a.c. and the secondaries are phased so as to obtain 1600 volts r.m.s. from rectifier plate to plate. The two-transformer scheme also provides the several 6.3-volt heater windings which are required.

The rectifiers employed are the high-vacuum, high-voltage 5R4GY, ideal for heavy-duty use. Two are used in parallel. In preliminary work, a swinging choke was used as input to the filter but it was found that a conventional smoothing choke works just as well. The requirements of sufficient minimum inductance and low d.c. resistance are met by the unit selected. The single high-voltage filter capacitor is oil filled.

One of the important requirements in obtaining high quality from beam tubes is regulation of screen voltage. This is not always mentioned in connection with amplifier construction articles and so does not receive the recognition it deserves. In our preliminary work we used VR tubes to regulate the screen voltage but had poor luck. By the time the screen voltage was stable, the VR tubes were well past their rated currents. Therefore the VR tubes were abandoned and an electronically regulated supply installed. A triode-connected 5881 is used as a pass tube, and a 6SJ7 is used as the control tube, with a VR-75 supplying the reference voltage. Bleeder current is passed through the VR-75 so that changes in 6SJ7 current have no effect.

Power Supply Circuits

Referring to the schematics for the power supplies—*Figs.* 4 and 5—it will be noted that the screen supply circuits are similar. During the development program, two types of power supplies were constructed. The first type used two receiver-type power transformers, with the high-voltage windings series-connected. The second employed a standard type of plate transformer which delivers 900 volts each side of center tap. This latter unit has a streamlined appearance, and results in an attractive power supply, but a number of extra filament transformers must be employed. *Figure* 4 shows the schematic of the two-transformer supply, with a number of filament windings being available on the existing transformers. *Figure* 5 shows the unit employing the single plate transformer with a multiplicity of filament transformers. There are advantages to both arrangements, but aside from the differences in transformer conections, the remainder of the power-supply circuit is essentially identical in both types of construction.

Referring to the regulator circuit, it is seen that the potentiometer P_1 is used to set the output voltage to exactly 200 volts—although it may be set anywhere in the range from 150 to 250 volts. Changes in input voltage have no effect. It will also be noted that the 6146's are operated with fixed bias. To provide this, a separate circuit is employed, using the 1-to-1 isolation transformer and a 75-ma selenium rectifier. Another VR-75 tube is used to stabilize this voltage, and enough current is drawn to make it steady. Two potentiometers, P_3 and P_4, are used in the amplifier to balance plate currents as well as to set the bias. Note that the positive side of the bias supply is grounded; therefore, the anode of the VR-75 should be grounded, and the cathode connected to the negative side of the bias supply.

A 100-watt, 10,000-ohm bleeder resistor is used to supply the 400-volt requirements of the regulated screen supply and the 375-volt requirements for the voltage amplifier. Details of the circuit are seen in the schematic, with the parts listed at the end of the article. 10-contact Jones plugs are used to interconnect the amplifier and the power supply. No trouble has been encountered in cabling the 750-volt plate supply with the other wiring, but care should be taken to place all live connections on female connectors.

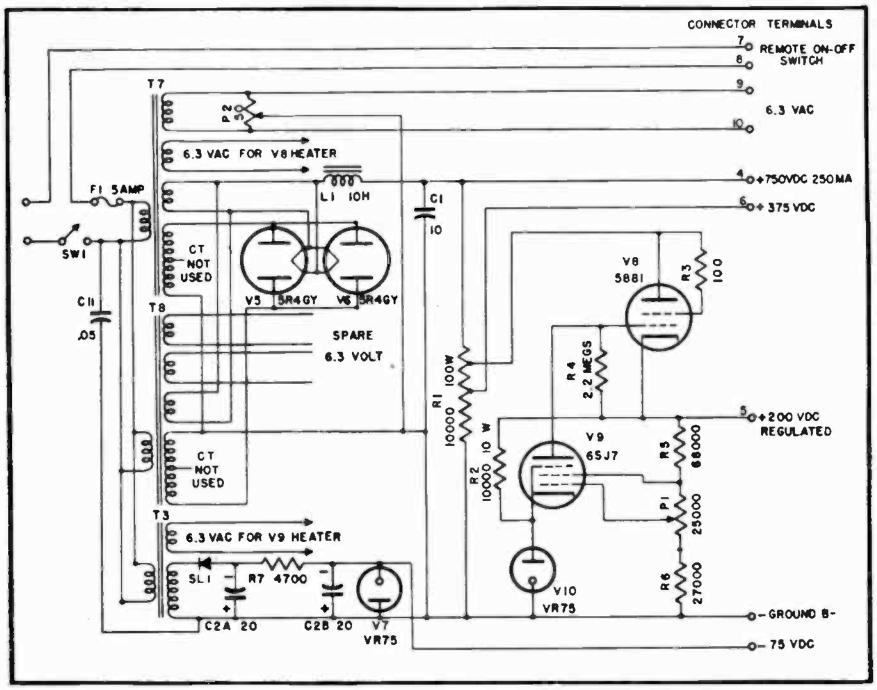

Fig. 4. Schematic of the No. 1 power supply, which employs two conventional receiver power transformers and the bias-supply transformer.

Performance

The performance of the Maestro amplifier fully justifies the name. The general requirements for frequency response, power output, distortion, and noise have been stated, and the results will be considered in that order.

The frequency response was measured with a 1000-ohm source resistance as this is typical of the source impedance of cathode followers used in the better "front ends." Under these conditions, the response is flat with no perceptible variation from 10 to 70,000 cps. There

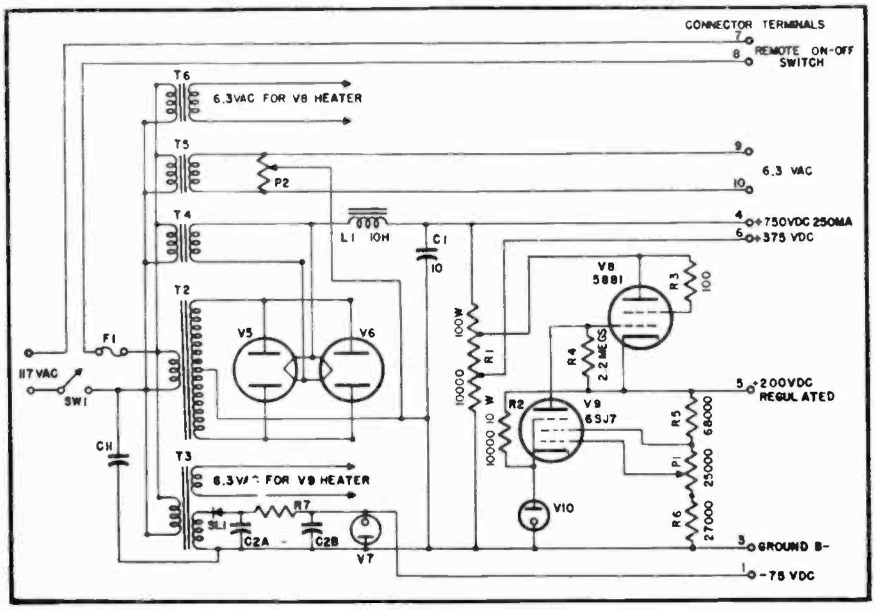

Fig. 5. Schematic of the No. 2 power supply, using a plate transformer and several filament transformers, in addition to the bias-supply unit.

is a 1.5-db rise at 5 cps, and there is a droop of 0.6 db at 100,000 cps. These frequencies represent the limits of our present measuring equipment. From the smoothness and steepness of the square-wave transmission, it appears that the response is better than the measured value. The completed amplifier passes square waves even better than the Musician's Amplifier, up to a 10,000-cps fundamental. At 30,000 cps the rise time is still vertical while preserving a flat top.

The single-frequency power output at 1000 cps is 90.2 watts, as shown in *Fig. 6*. This is just before the sine wave begins to be clipped, and when clipping does occur the clip is clean and symmetrical. There is no "fuzz" when the amplifier overloads. Full 90 watts is obtained at all frequencies from 25 to 20.000 cps with a smooth decline beginning at 30.000 cps, the 3-db-down point being at 40,000 cps. At low frequencies, the 3-db-down point is at 10 cps. The low-frequency performance of the amplifier when feeding a speaker load is superb.

The low distortion of the Maestro makes it a worthy part of a high-quality music installation. Using the power output as read on the IM set meter shows an IM distortion of 4 per cent only 1 db below 90 watts; at 2 db below 90 watts, the IM distortion is only 2 per cent, as shown graphically by the solid curve of *Fig. 7*. An important consideration in analyzing IM curves is the location of the "break" from a low-distortion flat portion of the curve to the upward bend. The ideal curve as a function of power would be horizontal up to the break point, then would rise sharply upward. This is the type of curve obtained from the Maestro. The break occurs at around 60 watts as read on the IM meter. From examination of the composite IM signal as viewed on an oscilloscope, it is evident that it is a complex wave and that meters calibrated on sine waves will not give a true measure of the IM signal output and hence the actual power output of the amplifier.

Thus it is desirable to find an equivalent sine wave which has the same peak value as the sum of the peak values of the low- and high-frequency components in the IM signal. Aston[3] points out that adding a second tone to another tone causes less than 0.5 db increase in a VU meter indication, but actually the peak amplitude of the combined signal is 1.25 times that of the low frequency. Aston also points out that IM meter measurements can be converted to equivalent sine-wave power by multiplying the IM meter power by 1.47. Strictly speaking, the term "equivalent sine wave" has no point in IM measurements since the term IM presupposes two frequencies. However, the concept is useful and does check with practice. Examination of the IM curve shows the break at around 60 watts. On equivalent sine wave, this is 88.3 watts, as

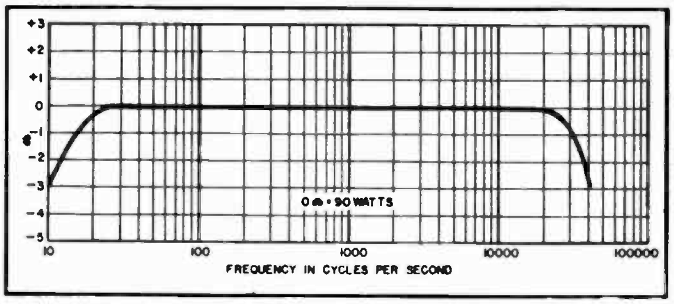

Fig. 6. Power output vs. frequency curve for the Maestro amplifier.

shown by the dotted curve in *Fig. 7*. This is close to the 90 watts—the power output at which a sine wave begins to be distorted. The amplifier can be considered to be almost distortionless below 75 watts on a sine-wave basis, since this corresponds to around 50 watts on the IM curve which is well below the knee or break-point.

The gain of the Maestro amplifier from a 1000-ohm source impedance—which is representative of cathode followers—is 50 db. This is measured in accordance with methods described by Haefner[4] and represents the power increase that is obtained from a 1000-ohm source whose open-circuit voltage is 1.9 volts. If this generator is terminated in a 1000-ohm load, the power in this load is .00091 watts. If the load be removed

damping factor of 10.6, which is entirely satisfactory on any speaker. No traces of hangover have been detected on any of the speakers used.

Preliminary Output Transformer Tests

During early development and before the arrival of the output transformer designed for this amplifier, tests were made using various other output transformers. Table 1 shows the results obtained with three transformers which were available: (A) Peerless S-265-Q, with a primary impedance of 10,000 ohms, 40 watts; (B) Western Electric KS-9496 Beachmaster transformer, primary impedance 9000 ohms, 250 watts; and (C) Partridge deluxe type CFB, primary impedance of 10,000 ohms, 60 watts.

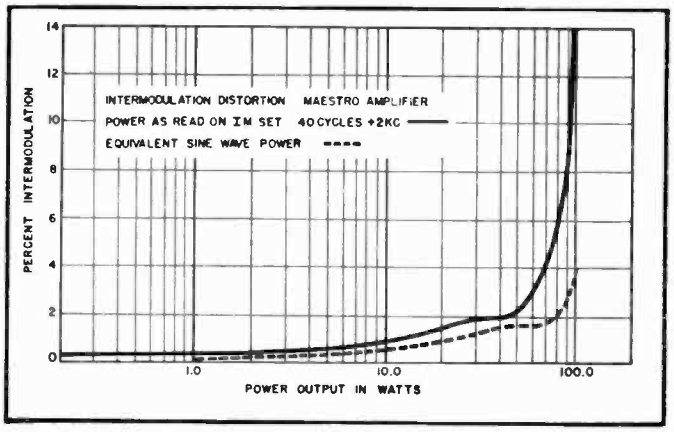

Fig. 7. Intermodulation distortion curves for the Maestro. The solid curve represents distortion for power output as read on the IM set meter. The dotted curve represents the same distortion plotted against equivalent sine-wave power.

and the amplifier connected, the power output will be 90 watts, or a 50-db increase. For those not familiar with this gain concept, let us state that the Musician's amplifier has a gain of 44 db by this method. For those not familiar with the gain requirements of their preamplifiers, it may be said that if the preamplifier will deliver 1.9 volts into a 0.5-meg load, the Maestro will put out the full 90 watts.

The noise level with the input shorted is around 5 mv across 16 ohms, which comes out to –28 dbm. This is 77.5 db below 90 watts, which is reasonably good for an amplifier of this power. In practice, it has been found that the noise is inaudible at 1 foot from efficient modern speakers.

The internal output impedance, or the source impedance which feeds the speaker, is 1.5 ohms on a 16-ohm strapping of the secondary. This gives a

Table 1

POWER OUTPUT IN WATTS

Frequency	Transformer		
cps	(A)	(B)	(C)
20	50.7	11.0	42
30	64	23.8	56
40	64	36.0	56
100	64	75.0	56
1000	64	75.0	56
5000	60	75.0	56
10000	34.4	66.6	56
20000	30.2	58.9	39

In fairness to all concerned, it should be stated that only the Beachmaster was intended for service such as imposed by the 6146, but it had very poor low-frequency power delivery, being intended for voice only. However, the other transformers are well known and were on hand so they could be tried readily.

Intermodulation measurements, using 60 and 3000 cps mixed at a 4-to-1 ratio, produced the results shown in Table 2.

[3] R. H. Aston, *Technicana*, AUDIO ENGINEERING, Sept. 1948.

[4] Sylvester J. Haefner, "Amplifier gain measurement." *Proc. I. R. E.*, July 1946, p. 500.

Table 2

INTERMODULATION DISTORTION

Power Output on IM set Watts	Transformer		
	(A)	(B)	(C)
56.2	23	10	21
42	1.5	1.4	3.5
36	1.3	1.7	1.5
25	1.2	1.3	1.0
6.2	0.7	0.5	0.5

Note that in the power output data the greatest midrange power comes from the Beachmaster which has the lowest d.c. resistance, but power at low frequencies is poor. Note also that IM measurements made at 60 cps for the low frequency do not show up the Beachmaster. Had 40 cps been used, the difference would have been apparent.

All tests were made under identical conditions using the series-connected power supply and VR tubes for screen regulation. 20 db of feedback was used. Under the same conditions, the Peerless S-268-Q developed 80 watts from 25 to 30,000 cps instead of 75 watts for the Beachmaster.

Listening Quality

We have said many times that listening quality in music is everything. In building a new amplifier, we naturally try to get the widest range, least distortion, and most power, but if the finished product does not sound better, time is wasted. Through the courtesy of Walter Toscanini, we had at our disposal some really fine tapes and a wide variety of loudspeakers to check the listening quality of this new amplifier. It turned out to be truly the finest reproduced music that we have ever heard. It has been fed into large two-way speakers where the superiority is immediately evident, and it has even been tried on speakers rated at 10 watts where it adds immeasurably to the performance. However, let us warn that this amplifier must be used with caution. In particular, great care must be taken to eliminate switching clicks in early stages as well as pops from phonograph motor switches. With the gain well advanced, any such click is almost certain to damage the loudspeaker—particularly the tweeter, if one is used—because of the high power capability of the amplifier. Remember that you can't crank up the gain on this amplifier without considering the program material any more than you can jam down the gas pedal on a super-powered car without considering the traffic conditions. A word to the wise is sufficient.

We have said that the Musician's amplifier is good, and that statement still stands for 99 44/100 per cent of music lovers, but for the 56/100 per cent who want the last word in realism as of the present state of the art, here is the amplifier. Just try it and see—but remember the precautions.

Parts List

Amplifier and Power Supplies

C_1	10 µf, 1000 v, oil-filled
C_2	20–20 µf, 150 v, electrolytic
C_3	.05 µf, 400 v, paper
C_4	20–20 µf, 450 v, electrolytic
C_5	30 µf, 500 v, electrolytic
C_6	20 µf, 450 v, electrolytic
C_7, C_8	.05 µf, 600 v, paper
C_9, C_{10}	0.25 µf, 600 v, paper
C_{11}	.05 µf, 600 v, paper, metal-cased
F_1	5-amp 3AG Littelfuse
L_1	10 H, 250 ma, filter choke; Peerless C-455-A or equivalent
P_1	25,000-ohm potentiometer, linear
P_2	50-ohm potentiometer, wire-wound
P_3, P_4	0.1-meg potentiometers, linear
R_1	10,000 ohms, 100 watt, wire-wound, adjustable, with two sliders
R_2	10,000 ohms, 10 watt, wirewound
R_3	100 ohms, 1 watt
R_4	2.2 megs, 1 watt
R_5	68,000 ohms, 2 watts
R_6	27,000 ohms, 2 watts
R_7	4700 ohms, 1 watt
R_8	1.0 meg, ½ watt
R_9	47,000 ohms, 1 watt
R_{10}	0.47 meg, 1 watt
R_{11}, R_{12}	22,000 ohms, 1 watt, matched
R_{13}	27,000 ohms, 1 watt
R_{14}	22,000 ohms, 1 watt
R_{15}, R_{16}	0.47 meg, ½ watt
R_{17}	390 ohms, 1 watt
R_{18}, R_{19}	47,000 ohms, 2 watts, matched
R_{20}, R_{21}	0.1 meg, 1 watt
R_{22}	10,000 ohms, 1 watt, for 20-db feedback
R_{23}	0.1 meg, 2 watts
T_1	Output transformer, Peerless S-268-Q; 8000-ohm primary to 16, 12, 8, and 4 ohm secondary, 90 watts power capacity
** T_2	900–0–900 v at 250 ma; Chicago P-67
T_3	125 v, 15 ma, half-wave; 63 v at 0.6 a; Stancor PS-8415
** T_4	Filament transformer, 5 v at 4 a; Chicago FO-56, Peerless F-138-E, or equivalent
** T_5	Filament transformer, 6.3 v at 3.0 a; Chicago FO-63, Peerless F-072-X, or equivalent
** T_6	Filament transformer, 6.3 v at 1.5 a; Chicago FO-615, Peerless F-036-X, or equivalent
* T_7, T_8	400–0–400 v at 300 ma; 6.3 v at 4.0 a; 6.3 v at 5.0 a; 5 v at 4.0 a; Peerless R-800-A or equivalent
V_1, V_2	6SN7
V_3, V_4	6146
V_5, V_6	5R4GY
V_7, V_{10}	VR-75
V_8	5881
V_9	6SJ7

* Use only in power supply #1
** Use only in power supply #2

A Medium-Power Tetrode Amplifier With Stabilized Screen Supply

CULLEN H. MACPHERSON*

Regulation of the final-stage screen supply permits high negative feedback and excellent performance in an easy-to-build amplifier complete with preamplifier and tone-control stages.

MANY OF THE MEMBERS of the audio fraternity view with alarm the increasing complexity of audio equipment. Although in most instances the addendum is accompanied by "higher fidelity," the price is sometimes higher yet. The need has been expressed for a fairly simple, self-contained audio amplifier of conservative design and acceptable performance which might be constructed for use in an apartment or small home. The subject of this article is such an amplifier.

Before beginning the design several features were fixed on as being desirable. (1) The amplifier should have inputs for both a tuner and a phonograph, with appropriate equalization for a reluctance-type pickup. (2) It should have a tone-control system furnishing both boost and attenuation of bass and treble. (3) Adequate power at low distortion should be available from the power output stage without requiring tubes of high dissipation capabilities or large power supplies. (4) The unit should be self contained and of small physical size.

A chassis 11½×6×2 inches with a 200-ma power transformer, output transformer, and 7 sockets mounted was available and design of the main amplifier was begun with this unit in mind. *Figure* 1 shows the result. The output stage was designed and constructed first, so that the tone compensation and preamplifier stages might be matched to it. Push-pull 6V6's were selected as power output

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tubes because of their relatively high efficiency and modest driver requirements. The 6V6's were fed from a cathodyne phase splitter which was coupled to a 6SJ7 voltage-amplifier stage. By using direct coupling between voltage amplifier and phase splitter the number of capacitors in the feed-back loop was reduced to one and the phase shift introduced by it was partially compensated for by the insertion of an R-C network between the grid of the phase splitter and ground.

It will be noted in *Fig.* 1 that the second triode of the 6SN7 is employed as a voltage regulator for the screens of the 6V6's. Deriving the screen potential from a source of low impedance such as this results in lower distortion, greater stability at peak power operation, and allows the use of more feedback around the output stage. The residual ripple present at the cathode of the voltage regulator is effectively filtered by the use of a 1-µf paper capacitor.

Since unbalance current flowing through the output transformer would cause loss of power handling capability at low frequency, steps were taken to balance the last stage. Pairs of 6V6's were selected with transconductance matched within 5 per cent. Their grids were then tied together and a 60-cps signal introduced to the grid circuit from the heater bus. Adjustment of the 150-ohm balancing potentiometer to the point producing minimum 60-cps output balances the stage dynamically.

Figures 3 and 4 show how the components were mounted on the existing chassis. This makes a compact and convenient arrangement, but other constructors will, of course, do as well with other arrangements.

The Preamplifier

The method of bass and treble control employed in the preamplifier, *Fig.* 2, was selected because it required only two shielded leads from the amplifier proper to the treble, bass, and volume controls. Such a configuration allows simplicity of both electrical and physical arrangement. Placement of a variable

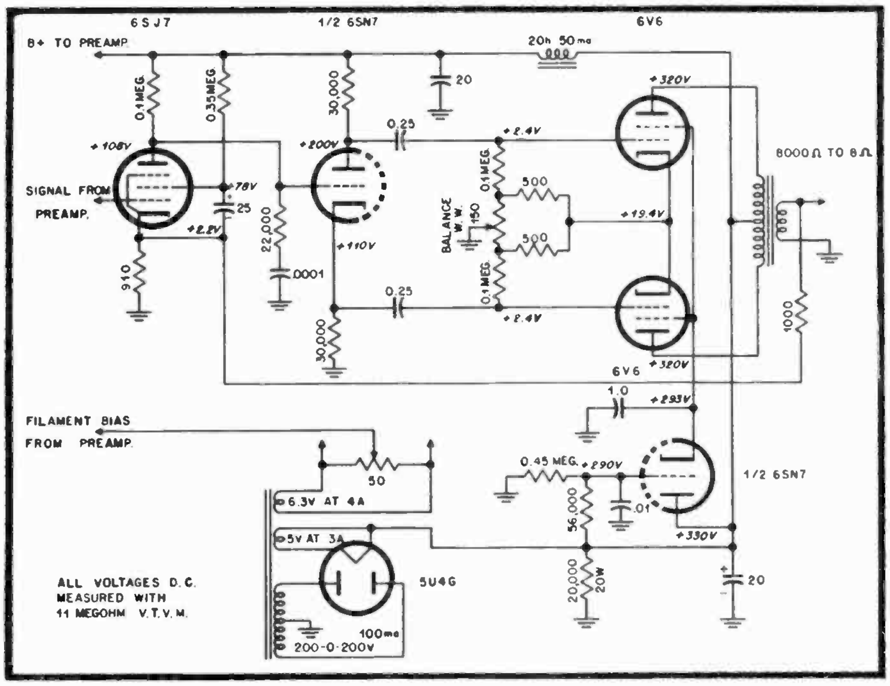


Fig. 1. The main amplifier. Half of a 6SN7-GT is used as a screen-voltage regulator. The 6SJ7 voltage amplifier and the phase splitter are direct-coupled.

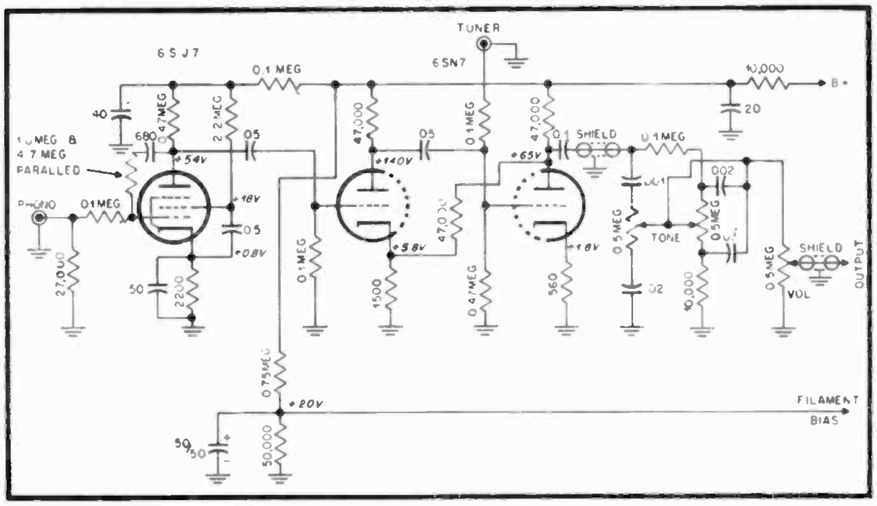
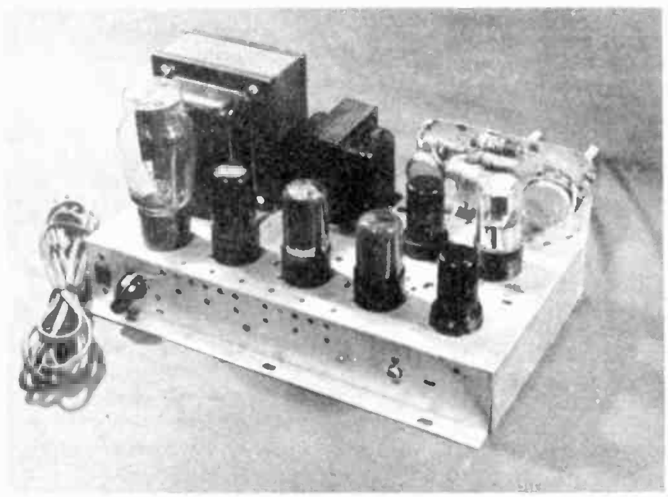


Fig. 2. The preamplifier provides for both magnetic phonograph pickup and tuner. The 6SJ7 is a phonograph preamplification stage with a simple feedback loop for bass equalization. The tone control is a single-shaft affair a center "flat" position and treble and bass boosts at the two ends of rotation.



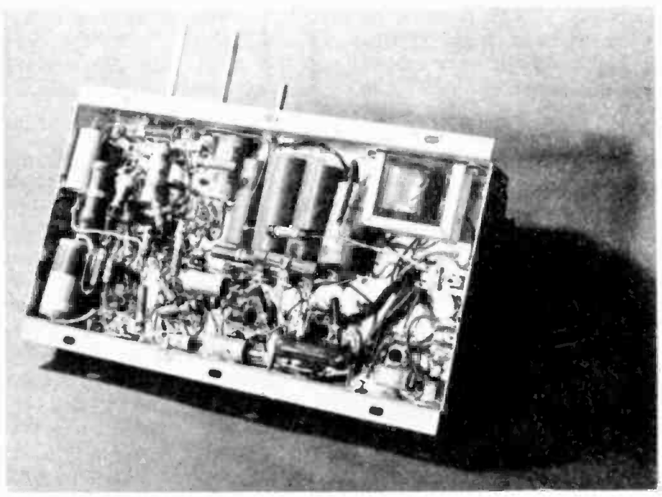


Fig. 3 (left). The entire amplifier, including preamplifier, is on a single chassis with the power supply.
Fig. 4 (right). The underside view shows that the space is well utilized.

Zero-Impedance Output Stage

RAYMOND G. ANTHES*

Excellent transient and low-frequency response and good loudspeaker damping make this amplifier suitable for high-quality, low-power applications.

THE ZERO-IMPEDANCE STAGE to be described was designed for home use, along with its driver, to give good quality performance at moderate cost. A series R-C circuit (R_1, C_1 in *Fig.* 1) shunts the primary of the output transformer so that the output tube works into almost unity power factor load. This minimizes harmonic distortion and phase shift. The feedback circuits are direct coupled and the negative voltage feedback is taken from the primary of the output transformer rather than the secondary in order to reduce undesired phase shift to a minimum in this feedback loop.

The low-frequency response is exceptionally good because the stage is effectively acting as a zero-impedance source feeding the primary of the output transformer. The output transformer used was of good quality and had 1-inch stack. A frequency response taken with the loudspeaker connected, and measuring output voltage across the secondary of the output transformer indicated the 3-db-down point was below 20 cps at the low end, and at 5000 cps at the high end, and only 9 db down at 15,000 cycles per second. At 2½ watts output into a resistance load at 400 cps, the total r.m.s. distortion was under 5 per cent. This is relatively high by most standards, but quite low for a 6V6.

A disadvantage of taking the negative voltage feedback from the primary of the output transformer is that this feedback cannot correct for the fall-off in high-frequency response in the transformer. The writer prefers to sacrifice some high-frequency response for minimum phase shift in the negative feedback circuit. This assures that the feedback works most effectively, reducing intermodulation distortion to a minimum, giving maximum reduction of harmonic distortion and maintaining a low-impedance source feeding the output transformer, over and beyond the complete audio frequency spectrum. It is possible to compensate for this loss in highs by a fixed equalizer in the preamplifier, but this was not done because the high-frequency loss was not serious. Most preamplifiers incorporate some form of tone control circuit with treble boost which

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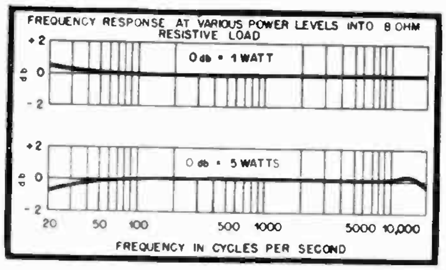
TETRODE

(from preceding page)

resistance in the input circuit of a feedback amplifier is not the best design practice, but the only way to circumvent such a situation is either to add another stage or to place the control at an earlier point. No instability has been observed in the present arrangement, however, in a year of satisfactory operation.

The 6SN7 is arranged with triodes in cascade as a feedback pair. Use of feedback in this manner reduces tube noise, lowers the output impedance to the tone-control circuit and also effectively reduces stray signals from the tuner circuit when the latter is not in use. Injection of the tuner signal at pin 1 of the second triode furnishes isolation from both the tone control and the preamplifier.

The preamplifier is quite a simple affair by present standards and the only defense offered, if one is needed, is that it performs quite satisfactorily. The circuit is similar to that of Williamson in which bass equalization is accomplished by a grid-plate feedback loop, the turnover frequency being determined by the R-C product and the amount of feedback by the ratio between R and the grid stopper resistor. The turnover point is arbitrarily fixed at 300 cps which has been demonstrated in listening as a suitable compromise. The 6SJ7 has been found to be more quiet and less microphonic than triode pairs more conventionally used in preamplifiers. Undoubtedly the metal shell is responsible for some measure of hum reduction, and biasing the heater supply above cathode potential to prevent heater-cathode emission also contributes to reducing the output hum to the point of 83 db below 10 watts.



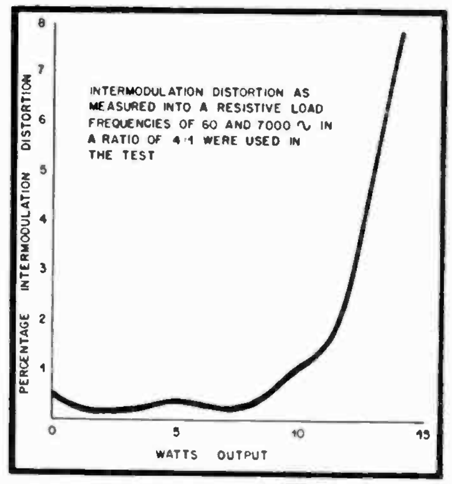


Fig. 5 (right). Distortion curve shows that intermodulation (60 and 7,000 cps, 4:1) just begins to rise above 1 per cent at 10 watts.

Fig. 6 (above). Frequency response curve made with an 8-ohm resistive load at 1- and 5-watt power levels.

Figure 5 shows that the distortion level of the amplifier is acceptable by modern standards, and may even be said to compare favorably with the hallowed Williamson. Such performance is attributed to the action of the screen voltage regulator in holding the screen potential at a fixed percentage value below that of the plate. Undoubtedly, the large reserve of the power transformer, giving excellent voltage regulation to the power supply, is also an assisting factor. *Figure* 6 shows frequency response at 1 and 5 watts output.

In conjunction with a modified Gately Superhorn transducer the amplifier has quite smooth overload characteristics, and when deliberately overdriven to a level of 22 watts of sine wave output into a speaker load at 2000 cps, rounding, rather than clipping of the waveform was observed.

It is the writer's earnest desire to steer away from the sea of superlatives. At the present time the amplifier suits him; undoubtedly this will not always be the case. It is not to be heralded as the latest answer to the audio man's conquest of the 80-meter band. Nor is it necessarily "better than any triode amplifier." It does, however, have the appealing attributes of straightforwardness, adequate performance with respect to both power and frequency, and small physical size. It is felt that the circuit may well be appealing where budgets are modest and space limited.

can be used for this equalization.

The use of the series R-C network across the primary of the output transformer to provide virtually unity power factor load to the tube is not new. The theory of this is well known. If a series *R-C* circuit and a series *R-L* circuit are connected in parallel as shown in *Fig.* 2, where the *R*'s are equal, it can be proved that the impedance of the parallel combination will be a pure resistance equal to *R* at all frequencies, if $R = \sqrt{L/C}$. At frequencies above the resonant frequency of the loudspeaker, the impedance measured across the primary of the output transformer with the loudspeaker load on the secondary may be roughly approximated by a series *R-L* circuit. Consequently, within this frequency range, which extends from approximately 125 cps to the highest audio frequencies, the composite load impedance presented to the tube is very nearly pure resistance with small variation in magnitude with frequency.

If values of R_1 and C_1 are chosen to give the optimum composite load impedance, there will be appreciable reduction in available output power at the higher frequencies where the reactance of C_1 becomes small in comparison with the magnitude of R_1. This is a serious disadvantage. A compromise between these two factors was made in this design.

Adjustment of R and C

The effect of changing R_1 and C_1 can be observed readily on an oscilloscope by the simple circuit of *Fig.* 3, and the values of R_1 and C_1 were finally selected in this way. The value of R_p used was 47,000 ohms, which approximates the plate resistance of the 6V6. The phase angle of the combination is determined from the ellipse appearing on the screen.

For the tube operating voltages used, the load impedance *Z* presented to the tube should be from 7000 to 10,000 ohms. The output transformer should match the loudspeaker to this load. A good compromise for R_1 is 2 to 4 times the magnitude of *Z* and the time constant R_1C_1 should be approximately 150 microseconds.

The value for *Z* used by the author was low, being around 5,000 ohms; R_1 was 15,000 ohms and C_1 was 0.01 μf. These values are not critical.

For R_1C_1 time constant of 150 microseconds, the reactance of C_1 is equal to R_1 at approximately 1000 cps. Consequently the power loss in the resistor is low below this frequency. In music, very few fundamental tones occur above this frequency which corresponds to two octaves above middle "C" on the piano. Even though some available power is lost at higher frequencies, it is not a serious matter.

In the usual 6V6 class "A" power amplifier, the average screen current and the average plate current both increase when a signal of sinusoidal or symmetrical waveform is applied. If the signal is keyed on and off and a d.c. milliammeter placed in the plate circuit, the meter fluctuates wildly. If instead of maintaining a constant screen voltage, a particular value of screen dropping resistor R_2 is chosen, it is possible to minimize this fluctuation in d.c. plate current provided the screen bypass capacitor is removed. The action is simple. In the presence of signal, screen current increases. The resulting increased voltage drop in R_2 causes a reduction in screen voltage just sufficient to provide the required compensation. In order to obtain instant action, the screen-bypass capacitor must be removed. The removal of this capacitor results in about 10 per cent reduction in voltage gain, and a loss in screen filtering.

The use of direct-coupled voltage feedback eliminates the need for a large blocking capacitor and ensures proper operation of the feedback circuit at the lowest audio frequencies. The current through the feedback resistor R_{f1} must come through the output transformer, which is a disadvantage. Resistors R_{f1} and R_{f2} should preferably be wire wound.

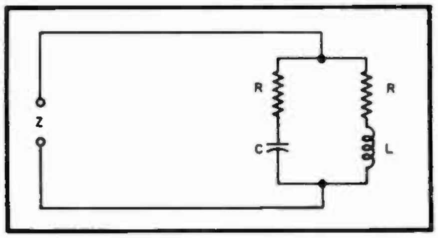


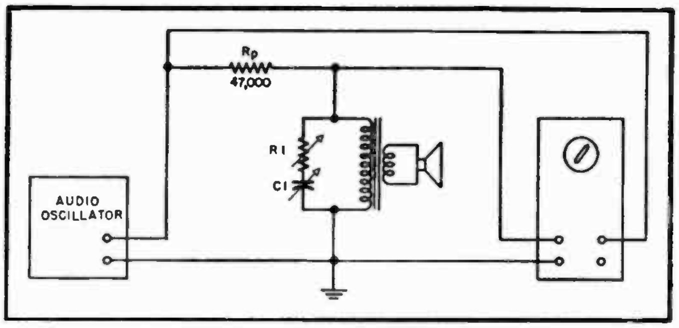
Fig. 2. Series R-C circuit and series R-L circuit in parallel.

Positive current feedback is obtained from a portion of the cathode bias resistor R_{k1} of the 6V6, as shown in *Fig.* 1. R_{k1} is a wire wound potentiometer, and serves as a control to adjust the output impedance. This potentiometer may be replaced by a 200-ohm resistor and the current feedback taken across the full 200 ohms.

The positive feedback control is adjusted in the following manner so that the tube presents zero impedance to its plate load. With the loudspeaker connected, an a.c. voltmeter is connected across the primary of the output transformer with moderate signal applied to the amplifier. A resistor of 5000 to 10,000 ohms (not critical as to value) is then shunted across the transformer primary. The feedback control is adjusted to a point where there is no change in output voltage as this resistor is connected or removed. With full 200 ohms in the feedback control, the output voltage actually increases when the resistor is connected across the transformer, indicating a negative-impedance source. It was found by measurement that the source impedance in the author's amplifier remains zero from 20 to 20,000 cps. It was not checked above this frequency. The drop off below 20 cps is due to the coupling capacitor C_2.

It should be noted that a blocking capacitor cannot be used in series with the negative feedback resistance R_{f1}. If one were used, the negative feedback would become ineffective at some low frequency, yet the positive feedback would still be effective, and low-frequency oscillation or motorboating is likely to occur.

Fig. 3. Circuit used to determine optimum values of R_1 and C_1 of Fig. 1.



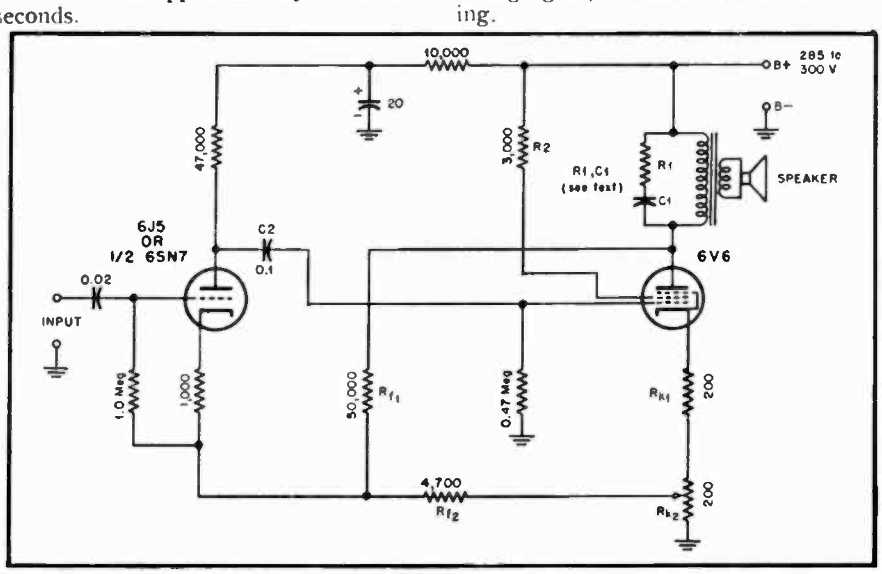


Fig. 1. Schematic of simple two-stage amplifier employing negative voltage feedback an positive current feedback to obtain an output impedance approximating zero.

A Dual–Channel Control Amplifier For Stereophonic Music Systems

WAYNE B. DENNY*

This control center affords facilities for either genuine two-channel (binaural or stereophonic) use or for what the author terms pseudo-stereophony—meaning distribution of a single-channel source to several speakers through two channels.

Readers of Audio Engineering are probably well acquainted with the principles and advantages of the stereophonic reproduction of music. Those who had the opportunity to hear demonstrations of stereophonic reproduction at the Audio Fair will testify to the enhanced realism of stereophony over the usual single-channel reproduction. Recently a few stereophonic discs have appeared on the market and it appears likely that more will follow. A few radio broadcasters have employed their AM and FM outlets to provide 2-channel pick-ups of studio programs, and those listeners who have separate AM and FM receivers have been able to enjoy two-channel stereophonic reproduction. Response to these experimental programs has been excellent.

Pseudo-Stereophony

Long before the advent of practical stereophonic reproduction there existed a group of listeners who preferred to hear their music reproduced *via* a multiplicity of loud-speakers. This "multiple source" school is to be contrasted with the "point source" school. The writer belonged to the former group and, to the extent that he must be content with single channel audio, he still does. When high-quality program sources are available he uses as many as six loudspeakers situated at various points in the listening room. Sometimes a single power amplifier was used for all speakers. At other times two and even three power amplifiers have been used to drive the various speakers When two or three amplifiers were used it was possible to tailor the signals to the individual speakers by changing the volume and frequency response to provide what may be termed "pseudo-stereophonic" reproduction. Such reproduction takes on some of the characteristics of genuine stereophonic reproduction but the two must not be confused. Pseudo-stereophonic systems are essentially single-channel systems despite their use of separate amplifiers and speakers. Genuine stereophony employs a multiplicity of channels which are completely separate from microphone to speaker. However, many who have employed a pseudo-stereophonic system will testify that it *seems* to be far superior to the usual single-channel, single-speaker system. For them, the concept of the "hole in the wall" point source does not provide the necessary realism. The pages of this publication are probably not the place to indulge in the metaphysics of space perception. Yet, in a matter of this kind the reader is entitled to know the particular (and, perhaps peculiar) prejudices of the writer, so that due allowance can be made for them. Like true stereophonic sound, pseudo-stereophonic sound has to be heard to be appreciated. It is *not* the equal of stereophonic reproduction and it is manifestly inferior to true binaural reproduction. But in the opinion of many who have heard it, pseudo-sterophony is definitely superior to single-channel point source reproduction.

Early experiments with two or more speakers operating from a single channel showed rather conclusively that it is not sufficient to connect the voice coils in series or in parallel and hook them to a single amplifier. The results obtained from such an arrangement are better than from a single speaker but are not all they might be. What is needed is separate control of volume and frequency response for each speaker. Resistance networks could, perhaps, accomplish the former but not the latter. What is required can best be provided by two complete amplifying channels, each with its own volume control and equalizer circuits. This is unfortunate because such a system is, admittedly, a complex one. In fact, it is essentially the same as a true stereophonic system as far as speakers and amplifying equipment are concerned.

Thus, one of the major difficulties with either true or pseudo-stereophonic reproduction is the multiplicity of equipment required. Two sources—phonograph pickups, radio receivers, or tape reproducers—are required for the former. In addition two complete amplifying systems and two complete loudspeaker systems are required. More than two channels can be used, provided only that the required signals are available. Such a system is expensive and it is complicated to adjust. The multiplicity of controls serves to scare off all but the most hardened audiophile. Such a system, though technically excellent, is hardly suitable for use by Aunt Minnie!

These are some of the difficulties which confronted the writer. Was it practical to install a system which would be suitable for either single- or double-channel operation? Would it be too expensive? Could it be engineered to the point where it was suitable for use by the nontechnical listener?

There seem to be at least three solutions. The simplest, though not the best nor the least expensive, is the purchase of two complete music systems. Each system would require the usual components—a phonograph preamplifier, a control amplifier with selector switches. volume, and equalizer controls, a power amplifier, and suitable speaker with enclosure. To this outlay must be added a stereophonic arm with suitable cartridges, a good turntable, AM and FM

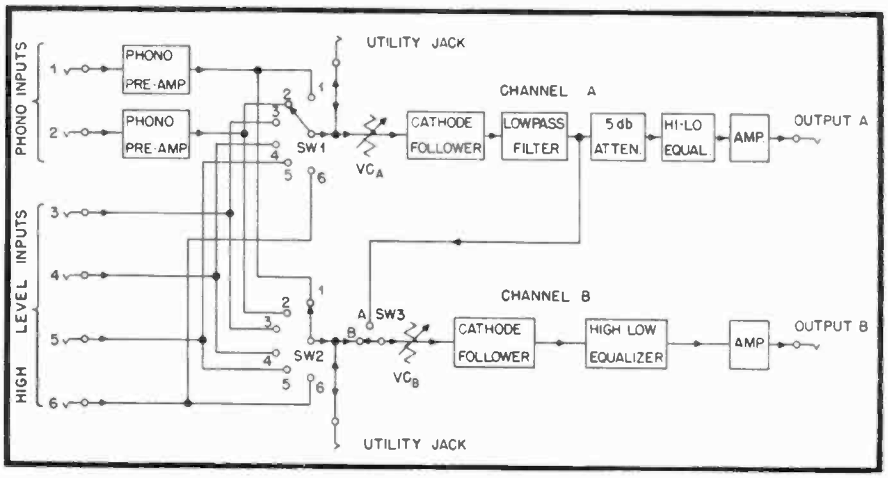


Fig. 1. Block diagram shows what the dual-channel control amplifier contains. The switches are shown in the positions required for dual-channel stereophony from phonograph records.

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tuners. The worst feature of this first alternative is the complexity of connections and controls. It is likely to lack flexibility unless the user will tolerate the use of temporary connecting cables. Aunt Minnie would hardly appreciate this kind of installation. It looks too much like the pilot compartment of a jet bomber. Its wires and knobs, completely unintelligible to the uninitiated, do not favor relaxed listening.

Another alternative would employ a switching panel wherein each signal source could be routed to one or more output channels. Each signal source might terminate in a key switch by means of which connection to channel A or channel B is made. Satisfactory adjustment of such a system requires separate control of the volume of each input source. But those readers familiar with the type of audio equipment used in broadcasting will notice the similarity of this possible system to the standard audio mixers which grace the control rooms of radio stations. Clearly such a mixer, while perhaps adequate for the purpose at hand, is hardly a practical solution to the problem as stated.

The Practical System

A much better and thoroughly practical solution is shown in *Fig.* 1. A glance at this functional diagram shows that this system will handle a total of six input sources. Two are low-level inputs provided with equalized preamplifiers suitable for use with variable-reluctance phonograph cartridges. The four high-level inputs are suitable for AM and FM tuners, television audio, crystal cartridges, and tape recorders.

Two output channels, A and B, are provided. Each output channel has its own selector switch to connect it to any one of the six possible inputs. Each channel has its own volume control (VC_A and VC_B in the figure), its own equalizer, and its own voltage amplifier for raising the level to a value suitable for driving a power amplifier. Channel A contains a low-pass filter for suppression of needle scratch and intercarrier radio noise and it also contains a resistance network providing about 5 db loss inserted between the filter and equalizer. channel B contains neither filter nor fixed attenuator for reasons to be discussed later. In each channel the volume control is succeeded by a cathode follower. The cathode follower in channel A isolates the volume control from the filter and it also provides a low-impedance source for the filter. In channel B the cathode follower isolates the volume control from the equalizer and provides a low-impedance source for the equalizer.

In addition to the main features of the system there are two gimmicks which require explanation. The first of these is the inclusion of two utility jacks, one for each channel. Each utility jack is connected directly to the arm of its selector switch and *ahead* of the volume control. Any signal fed to either channel is thus available for operation of a recorder. Further, if the switch is turned to an unused position, the utility jack can be used for temporary connection to an additional input source.

Although only six input sources are normally provided for, the selector switches are actually 12-position units. Only taps 1, 3, 5, 7, 9, and 11 are used but in these positions are consecutively numbered from 1 to 6. The increased spacing between live contacts materially reduces crosstalk. Crosstalk may be further reduced by grounding the unused terminals.

Another item of interest is the inclusion of switch 3. When this switch is thrown to the B position each channel functions independently. However, when it is in the A position channel B is bridged across channel A. This is a convenience when the system is used for pseudo-stereophonic reproduction. Under this condition the volume control VC_A regulates the gain of both channels. Further, switch 1 selects the single signal source. During this type of operation switch 2 serves to connect its utility jack with any one of the six inputs. This permits recording from one source while listening to another, a useful feature.

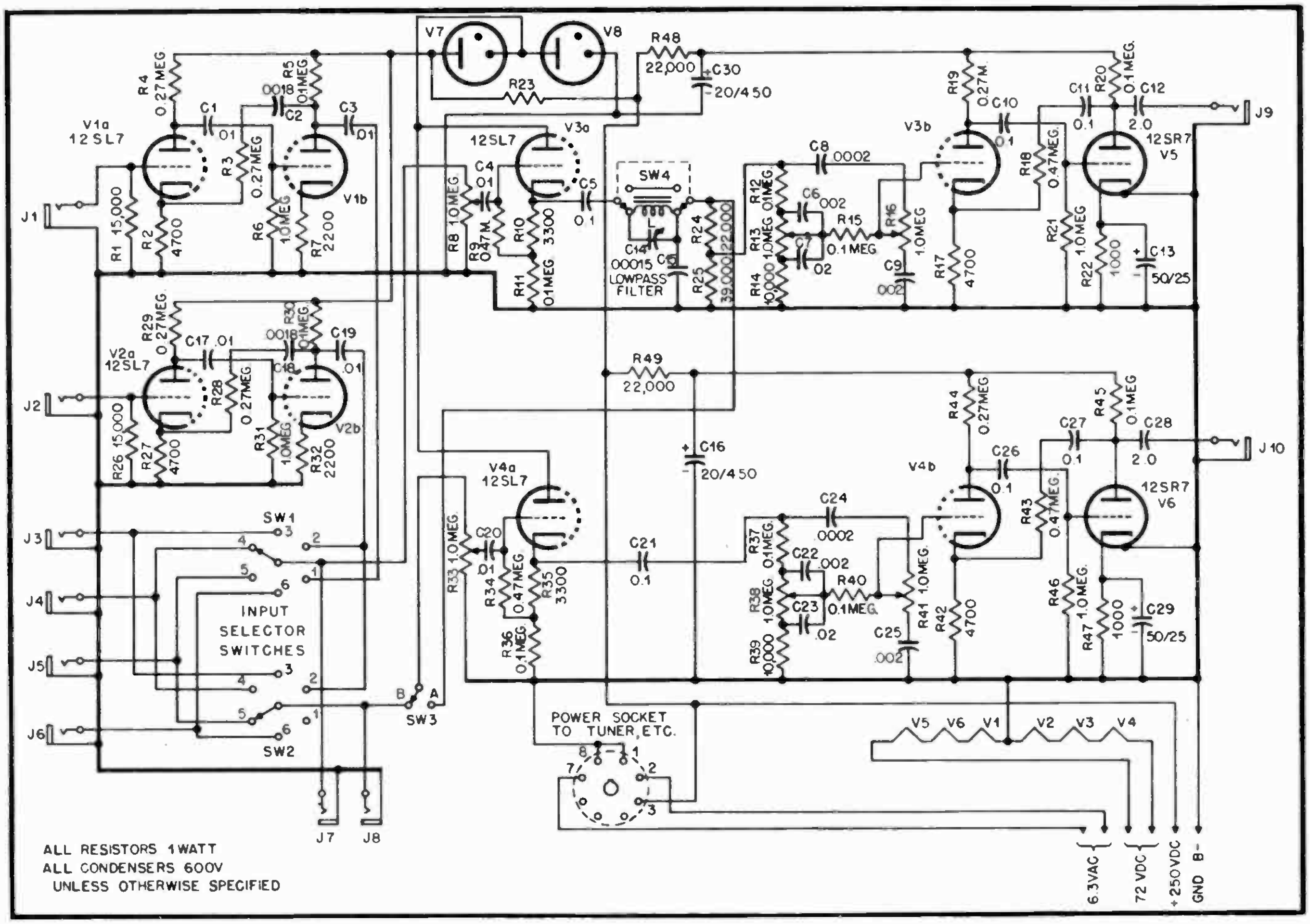

Fig. 2. This is the complete schematic diagram. The separate elements of the systems are conventional, with the switching the unique feature.

Now comes the other gimmick. Examination of *Fig.* 1 shows that the lower volume control VC_B is included in channel B at all times, even when switch 3 is thrown to the A position. It will be seen that under this condition the gain of the upper channel is about five db lower than the gain of the lower amplifier channel as measured from switch 1, provided VC_B is turned full on. Consequently, VC_B acts as a balance control; it serves to adjust the relative signal levels of the two outputs and has a margin of gain adjustment from zero to + 5 db. As stated earlier, some means of balance adjustment is a requirement for proper operation of a pseudo-stereophonic system. Note that the filter, if used, operates on both channels in this type of operation.

Pseudo-stereophonic reproduction is also possible with switch 3 thrown to the B position. Here, both selector switches are connected to the same signal source. Again, the maximum gain of channel B will be about 5 db higher than the gain of channel A. However, operation of the equipment is simplified when adjusted according to the preceding paragraph because selection of the input signal can then be accomplished entirely through adjustment of switch 1. In either case, channels A and B are separately equalized. In this respect the system is different from a commercially available control amplifier designed for single- or 2-channel use. But experience has shown that separate equalization, together with balance control, is desirable for best results.

It should be emphasized that when switch 3 is thrown to the A position, the operation of the system is rather similar to that of any other high-quality home music system. Even if Aunt Minnie doesn't balance the two outputs correctly the results can be no worse than those of an ordinary single-channel system. When it is intelligently used, the results can be very much superior.

This is a good place to take a tip from one prominent manufacturer of communication receivers. This manufacturer was well aware that many of his communication receivers—complete with selectivity controls, crystal filters, beat-frequency oscillators, bandspread dials, etc.—were often used by Mrs. Amateur for regular broadcast listening. To simplify the operation of the receiver in the hands of the lady of the house, this manufacturer indicated the proper settings of all controls by little colored dots. When each control was set so that it pointed to the appropriate dot the receiver was correctly adjusted for broadcast reception. There is no good reason why this same technique cannot be used for relatively complicated home music systems. Set each knob to point to the colored dot and the system is correctly adjusted for ordinary single-channel operation. In this way, Mrs. Audiophile can enjoy good music during the hours when her husband is away at work. The system may not be in optimum adjustment but it won't be too far off. Then, when Mr. Audiophile returns from work

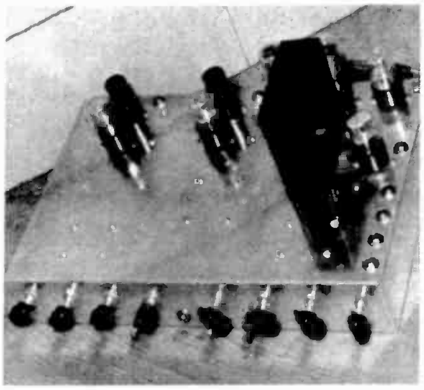

Fig. 3. The control amplifier is mounted on a chassis 11 × 17 × 3 inches. Probably a smaller one could be used.

he can twiddle the knobs to his heart's content while trying to squeeze the last decibel of "fidelity" out of the speakers.

The Circuit

The system whose functional diagram appears in *Fig.* 1 has been engineered into practical form. The schematic diagram appears in *Fig.* 2. The individual elements of the system are entirely conventional. Only the inclusion of two channels, the switching arrangements, and the gimmicks mentioned earlier make this system different from other more conventional systems, in principle at least.

Two power supplies (not shown) are required. One furnishes about 72 volts at 150 ma for six heaters connected in series. This power supply is *not* grounded. The heater string is grounded between V_1 and V_2 which are the two tubes operating at the lowest signal level. The other power supply furnishes about 250 volts d.c. for the plate supply. It is entirely conventional but should include plenty of filtering. The writer found a 2-section choke-input filter to be entirely satisfactory.

More than ordinary attention to lead dress is necessary to guard against crosstalk between channels. Inspection of *Fig.* 3 will indicate that no attempt was made to crowd apparatus. The chassis is large; it measures 11 × 17 × 3 inches and these dimensions are somewhat greater than those of the usual control amplifier.

It was found that the plate-supply filtering shown is adequate to reduce crosstalk between channels provided the input signal levels are all comparable. The writer's AM tuner produced signals higher in level than any other signal source (when measured at the selector switch) and there was some riding through of AM signals under certain conditions. This was corrected by inclusion of an attenuator in the tuner itself. Another alternative is to provide level controls (screwdriver adjusted) in all six input circuits as is done in some commercial units.

Two voltage-regulator tubes in series are used instead of electrolytic capacitors in one part of the plate-supply filter. The available plate-supply voltage *under load* should be about 250 volts. V_7 and V_8 should be OB2's (108 volts) or OC3's (105 volts). This arrangement permits about 210 volts for the preamplifiers. Resistor R_{83} should be rated at 10 watts and its resistance should be such that the current through it is at least 10 ma. A value of 4,000 ohms works out well for two OC3's and a 250-volt supply. Other combinations can be worked out for different supply voltages. However, make certain that the power-supply voltage is at least 30 volts higher than the sum of the nominal voltage ratings of the regulator tubes. Filters using voltage-regulator tubes are superior to filters using capacitors, particularly at low frequencies. (*Caution:* Do not shunt regulator tubes with a filter capacitor. This causes oscillation.)

The plate supply for the cathode followers is obtained from the junction of the two regulator tubes. This is permissible because the cathode followers normally operate at very low signal voltages in this unit. Further, the inverse feedback inherent in the cathode follower reduces distortion to extremely low values. Incidentally, it will be seen that every amplifying triode is included within a feedback loop. This feature contributes to the very low distortion of the amplifier which is a necessity if the full benefits of available high-quality power amplifiers are to be realized. Comparison of this unit with its predecessors, which did not employ as much feedback, gave immediate evidence of the lowered distortion. Intermodulation measurements using 60 and 7,000 cps, 4 : 1, indicate that the distortion is never more than a small fraction of 1 per cent with normal signal levels.

Design Details

The equalizers are conventional in every respect; the values of the components are similar to those used in commercial equipment of comparable quality. It will be noticed that only a single transition frequency is used in the preamplifiers. Previous preamplifiers built by the writer provided for a choice of three or four transition frequencies but experience shows that only the one corresponding to the AES curve was ever used to any extent. Anyhow, in a unit like the one described here the number of controls should be reduced to a minimum. Of course, the purist may, if he so desires, introduce more flexibility if he is willing to tolerate the added complexity of operation.

Ordinary volume controls are used in preference to loudness controls. The latter could be used but the bass boost available from the equalizers is sufficient to provide a close approach to the Fletcher-Munson curves at the volume levels normally used.

Examination of the photographs shows that there are six variable resistors and two selector switches, all operated by pointer knobs. Each channel requires a selector switch, a volume control, a bass control, and a treble control. Two toggle switches and two jacks also appear on the panel. One s.p.d.t. switch provides for single-chan-

Improved Phonograph Compensation Circuits

R. H. BROWN*

An up-to-date revision of a circuit presented over a year ago provides for the presently used recording characteristics, including the RIAA curve without which no modern amplifier is complete.

Since the preparation of the article entitled "Hi-Fidelity Phonograph Preamplifier Design"[1] the new RIAA characteristic has been announced and there has been much clarification regarding the recording characteristics actually used by different disc manufacturers. Because of these developments it has become desirable to revise the compensation circuits in the triode preamplifier design given in that article.

The over-all revised circuit is shown in *Fig.* 1. The 480-μμf capacitor shown in the diagram represents the total value of circuit capacitance at that point. Since the stray capacitance present will depend upon the construction of the preamplifier, one should begin with a lumped capacitance of around 400–420 μμf and add additional lumped capacitance as may be required to obtain the correct rolloffs. For convenience in making this adjustment, the legend gives in parentheses the proper relative output level of the amplifier in decibels at 10,000 cps for the various treble settings.

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[1] See page 36.

With the exception of the 12-db-per-octave rolloff on position 1, this compensator may be used in the grid circuit between two amplifier tubes which are not connected by a feedback loop. When so used the 43,000-ohm resistor between positions 6 and 7 of S_{1A} should be reduced by an amount equal to the internal resistance of the source of the signal supplied to the compensation network. For a simple plate-loaded voltage-amplifier stage without feedback this source resistance is simply the parallel combination of the plate resistance of the amplifier tube, the plate load resistance, and the following grid return resistance. The grid return resistance should be connected between ground and position 7 of S_{1A} so that it will not interfere with the performance of the compensation circuit.

Figure 1 also gives the revised bass compensation circuit. The bass rolloff resistors on positions 3 and 4 of S_{1B} may be selected by neglecting the unity in the denominator of Eq. (5)[2] in the article referred to above and selecting R_3 to provide a maximum low-frequency gain which gives the boost required by the characteristic as zero frequency is approached. For the LP and RIAA characteristics this zero-frequency boost is 15.4 and 20.7 db, respectively. Another method for selecting these resistors is to make the time constant of the C_4 and R_3 combination in *Fig.* 2 (Fig. 1 of the previous article), equal to the bass pre-emphasis time constant of the recording characteristic. For the LP and Orthophonic characteristics this is 1590 and 3180 microseconds, respectively. The method for selecting these resistors suggested in the previous article makes the compensation correct for the extreme low frequencies at which the gain of the basic amplifier begins to become a limiting factor, but it does not make the compessation correct in the range covered by the majority of the bass-frequency components.

The method of connecting the switching-transient-eliminating resistors as shown in *Fig.* 1 is preferable to that

[2]
$$A_{LF} = \left[\frac{A'}{1 + \dfrac{R_1 A'}{R_1 + R_2 + R_3}} \right], \qquad (5)$$

DUAL-CHANNEL AMPLIFIER

(Continued from previous page)

nel or stereophonic operation. The other switch is a d.p.d.t. unit for placing the low-pass filter in or out of the circuit. The two jacks are the utility jacks mentioned earlier. They are placed on the panel to permit ready access to connections to one or two tape recording channels.

A word about the low-pass filter is in order. You can calculate the constants for the filter from the equations for a single *m*-derived section if you want to. However, for the purpose at hand this is neither the easiest nor the best method for designing the filter. Instead, the writer installed a 150-μμf *variable* capacitor for C_{14} across the inductor, L. If L has an inductance on the order of 1 henry this capacitance can be adjusted to give "infinite" attenuation for interchannel whistles of 10 k.c. An audio oscillator and a v.t.v.m. (or oscilloscope) are helpful here but the adjustment can be made by ear, if necessary, while the signal source is an AM tuner. The value of C_{15} can be determined so as to provide attenuation starting around 7,000 or 8,000 cps. The actual capacitance required will depend on the Q of the coil and the load into which the filter works. The values of capacitance and load resistance given in the diagram worked out well for the particular coil used by the writer. Some changes may be expected when using other coils. Unfortunately, the inductor shown in the photographs is not commercially available.

The circuits used in the amplifiers were chosen on the basis of low distortion. Some preamplifier circuits—including some designed earlier by the writer—amplify the signal up to 10 volts or so and then the equalizer attenuates the signal. Most triodes show high intermodulation distortion at such high levels. For this reason the volume controls used in this unit are placed immediately following the selector switches. The equalizers *precede* the two-stage feedback pairs; the signal is reduced before amplification, not after it has reached the level of high distortion. Sufficient feedback is incorporated to reduce the generator impedance to a value of 2,000 or 3,000 ohms. More feedback could be used if desired because the gain is higher than required for any normal input signal. This same feedback also reduces distortion, as mentioned earlier. Distortion is already low because the signal never exceeds 1 volt anywhere in the entire unit.

Another feature of interest in connection with lowered output impedance is the use of large (2-μf) oil capacitors for C_{12} and C_{22}. There is little point in reducing the generator impedance of the amplifier to a low value if this advantage is lost by high series reactance in the output coupling circuit.

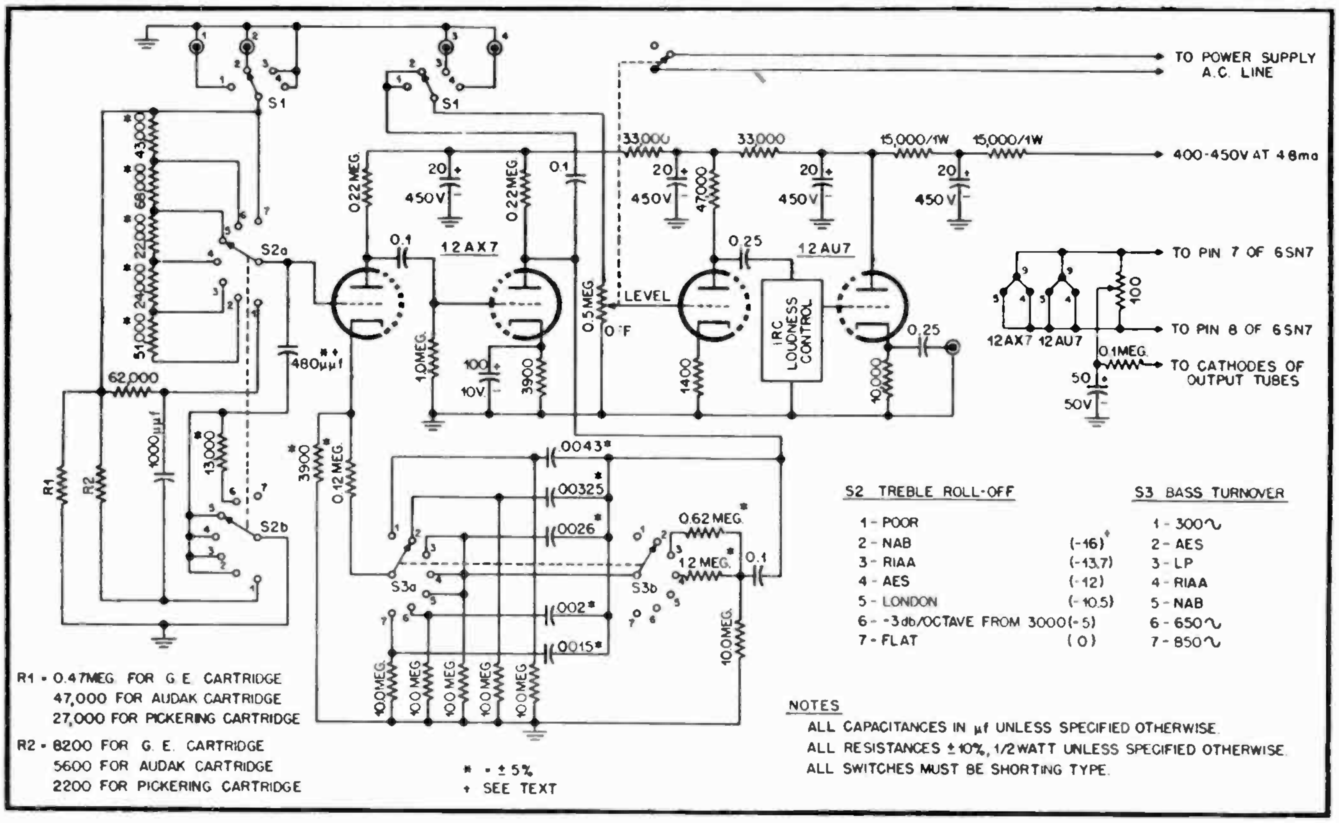

Fig. 1. Revised preamplifier circuit now includes the new and almost universally accepted RIAA curve.

given in the orginal article in that it places only one of these resistors in the feedback circuit at a time.

Ideally, the 0.12-meg resistor between S_{3a} and the first 12AX7 cathode should be shunted with a capacitance equal to 3.9/120 times the total capacitance to ground shunting the 3900-ohm cathode resistor. This point is simply a technical purism for a preamplifier using a tube like the 12AX7, but there might be some value in giving attention to it in a circuit which used a larger feedback ratio; e.g., a feedback bass-boost circuit in which the basic amplifier used low-mu tubes.

The author has received a number of inquiries regarding the possibility of adapting the triode preamplifier in the earlier article to include tone controls. With a bypass on the first 12AU7 cathode a tone control circuit with about 10-db insertion loss could be substituted for the loudness control. With the compensation adjustments provided by the revised circuits presented herewith, plus or minus 10 db of tone control should be more than adequate for all but the very unusual situation. The output section of the preamplifier should be redesigned if a greater range of tone control is required.

The variety of recording characteristics in use is now sufficiently great that one may dare to hope no new characteristic will appear which would make further revision necessary.

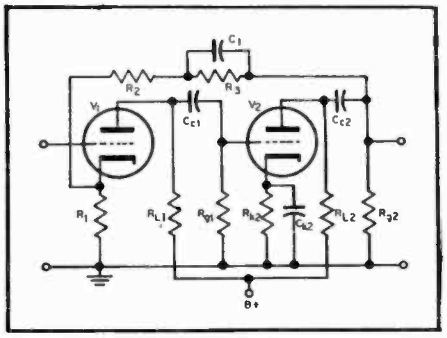

Fig. 2. Basic degenerative-bass-boost amplifier circuit. V_1 and V_2 and associated components, excluding feedback elements R_2, R_3, and C_1, comprise the basic amplifier. Triodes are shown for simplicity.

A Note on Volume Controls

CHARLES P. BOEGLI*

The author shows that it is not always satisfactory to place a high-resistance potentiometer just anywhere in an amplifier circuit if specified performance is to be obtained.

An audio amplifier is generally made up of several stages arranged in cascade so that the input signal is fed through one after the other, being amplified on the way, finally appearing as an output voltage at the loudspeaker terminals. Somewhere in this unit, it is usually necessary to provide a control over the gain of the entire amplifier, so that input signals of the given voltages can be made to yield output voltages of the desired magnitudes. This is most often accomplished in high-impedance locations, by means of a simple potentiometer; the output from the plate of one stage is connected to the top of the potentiometer, the other end of which is grounded, and the grid of the following stage is fed from the slider. The purpose of this note is to mention some considerations that appear to be generally overlooked in providing a "volume" control of this type.

Since hum and noise are most apt to assume noticeable proportions in the input stages of the amplifier where signal voltages are quite low, the volume control should generally be located after

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the first few stages. In this manner, the signal is maintained at the maximum value compared to the noise through these critical stages and becomes attenuated only when it has reached a value quite large in comparison to noise voltages. On the other hand, should the volume control be located too near the output tubes, the possibility is increased that the signal will attain sufficient magnitude before reaching the control to cause unnecessary distortion in previous stages. Placement of the control at a point where the maximum signal level is from *2* to 4 volts will usually lead to a fairly satisfactory design, suffering neither from distortion nor excessive noise. It goes without saying that the control must not be placed inside the feedback loop, if any are employed in the circuit.

The only disadvantage to locating the control otherwise than at the input to

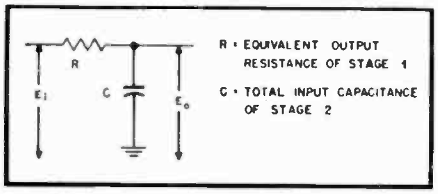

Fig. 1. Low-pass filter existing between two stages of an audio amplifier.

the amplifier is that the voltages of the signal sources used with the amplifier must—to a degree, at least—be controlled. Thus, with the volume control at the input we need concern ourselves only with the minimum input required for full output, but if the control is situated later, the *maximum* input also becomes important. An input stage handling a minimum of 0.5 volt may well be designed to carry a maximum of the order of 1.5 volts; obviously, if a tuner supplying 6 volts is connected to such an input, overloading will result. The best arrangement would therefore seem to be one in which the volume control is placed near enough to the output stages to maximize the signal-to-noise ratio and close enough to the input stages to prevent serious distortion, and in which individual pads are used on the various signal sources to bring their outputs to approximately a uniform level, preferably slightly over the minimum required to drive the amplifier fully.

Effect on High-Frequency Response

The high-frequency response of an amplifier (utilizing a good output transformer) is largely determined by the cutoff frequencies of the series of low-pass filters incorporated into it and made up of the output resistance of one stage and the input capacitance of the next, as indicated in *Fig.* 1. When one stage feeds directly into the next the problem of attaining good high-frequency response is fairly well defined, and the solution consists simply in keeping the output resistance of each stage as low as possible. Low-mu triodes with plate resistances of the order of 10,000 ohms are excellent in this respect and may be used with plate-load resistances of almost any size without adversely affecting high-frequency response. With high-mu triodes and pentodes the plate-load resistors, as well as the grid resistors of the following stages, must be kept small in order to realize good high-frequency response. But what occurs upon the introduction of a volume control?

Obviously, when the slider is at the top of the control, the situation is the same as if no control were present. When the slider is near the bottom, also, the following stage is fed from a very low impedance so that the high-frequency response is even better than when the slider is at the top. In less extreme positions, however, if the volume control resistance is quite high, there may be a substantial loss of high frequencies—and it is precisely these positions that are most important.

Reference to *Fig.* 2 will make this clear. R_2 is the output resistance of the preceding stage and R_1 is the volume-control resistance. The symbol a represents the position of the slider and measures the fraction of the voltage appearing at the top of the control that is applied to the following grid. Now R_o, the resistance into which the following grid looks is aR_1 in parallel with $R_2 + (1 - a)R_1$, which is found to be

$$R_o = \frac{aR_1R_2 + (a - a^2)R_1^2}{R_2 + R_1} \qquad (1)$$

We are interested in the manner in which R_o varies with a. Hence, differentiation of the above expression with respect to a yields

$$\frac{dR_o}{da} = \frac{R_1R_2 + R_1^2 - 2aR_1^2}{R_2 + R_1} \qquad (2)$$

The resistance R_o is a maximum at the value of a found by setting the right side of Eq. (*2*) equal to zero and solving for a:

$$R_1R_2 + R_1^2 - 2aR_1^2 = 0$$

$$a = \frac{R_2 + R_1}{2R_1} \qquad (3)$$

When a assumes this value, the maximum output resistance

$$R_{om} = \frac{R_1 + R_2}{4} \qquad (4)$$

For example, if we had a low-mu triode stage with an output resistance of 10,000 ohms connected to the top of a 1-meg. volume control, there would be some point at which the next stage would look back into as much as

$$\frac{10{,}000 + 1{,}000{,}000}{4} = 252{,}500 \text{ ohms}$$

and if the input capacitance of the following stage were sufficient to cause a

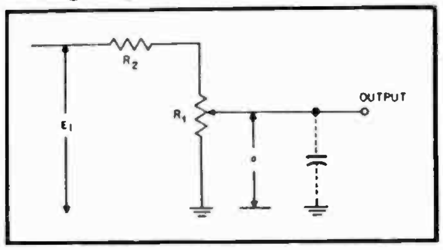

Fig. 2. Equivalent circuit of a stage followed by a volume control.

3-db drop at 1000 kc without the volume control, there would actually be a drop of 3 db at about 4000 cps at this position of the volume control.

We find, consequently, that the resistance of the volume control is rather stringently limited by the capacitance into which it must work. As an example, suppose we had a 12AY7 stage with an output resistance of 20,000 ohms working into a 6SL7GT with an input capacitance of 100 μμf. At best, there will be a drop of 3 db at 80 kc. Suppose further that we cannot tolerate a drop of more than 3 db at 50 kc in this particular stage. Then the maximum output resistance at any position of the control must be 32,000 ohms. To find the maximum permissible volume control resistance we use Eq. (*4*) with 20,000 ohms for R_2, 32,000 for R_{om}, and solving for R_1 we obtain $R_1 = 108{,}000$ ohms, and a 100,000-ohm control would be used.

Suppose we are more exacting and state that the high-frequency response with the volume control must never be any poorer than without it. This is the same as saying that R_{om} in Eq. (*4*) must be the same as R_2:

$$\frac{R_1 + R_2}{4} = R_2$$

$$R_1 = 3R_2 \qquad (5)$$

To realize this performance in the case just described we should have to utilize a 60,000-ohm control. That these values are much lower than those commonly in use will readily be recognized.

The installation of a volume control thus appears to be somewhat more involved than simply inserting a 1-meg. potentiometer at the amplifier input. In particular, the use of relatively low values is indicated in cases where the control is working into appreciable capacitance.

The Selection of Tone-Control Parameters

EDGAR M. VILLCHUR*

Proper design of tone controls requires a study of the conditions which must be corrected. The author delineates these conditions, and explains the requirements for each.

Various tone-control circuits with given transition frequencies[1] and rates of boost or cut have been extensively discussed in technical literature. Little has been written, however, about considerations involved in the selection of the response-curve parameters, which sometimes seem to be chosen more or less arbitrarily. While "flat" power amplifier stages are carefully designed for an audio-frequency response which is kept to a maximum random deviation of a fraction of a decibel, improper tone compensation (either manual or automatic) in the same amplifier may introduce, or leave uncorrected, very large inaccuracies of frequency distribution relative to the perceived original. These inaccuracies have on occasion completely overshadowed the benefits of the above-mentioned careful design, and have become a primary factor in determining over-all quality.

Fixed tone equalization is used when the frequency characteristics for which compensation is being made are constant. Modern home reproducing systems contain several fixed or automatically variable audio equalization circuits associated with FM pre-emphasis, recording characteristics, pickup frequency distortion, or changes of volume level. The inclusion of such circuits does not make variable tone control superfluous, since there are many unpredictable conditions for which it may be desirable to adjust the frequency response of the reproducing equipment.

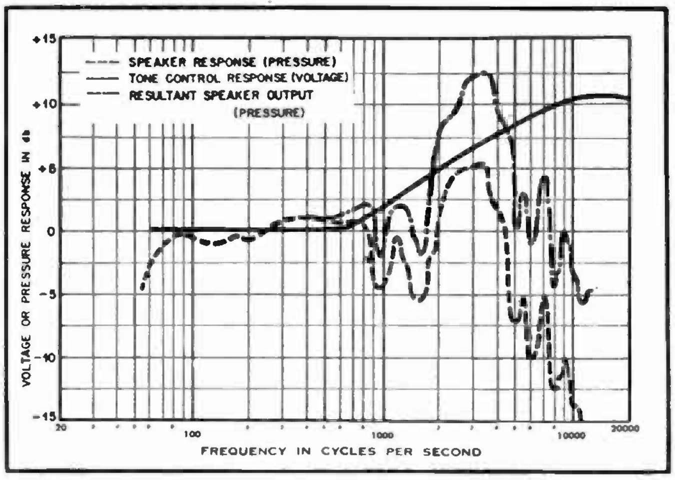

Fig. 3. Resultant frequency response of an audio system using a commercial speaker in the $50 class and the treble boost of (A), Fig. 1.

The designer of fixed equalizers does not ordinarily have to worry about what parameters to use; with certain exceptions his transition frequencies, and the required rate of boost or cut, have been exactly determined by his problem. In contrast a tone-control designer must furnish the means for correction, by the same circuit, of all sorts of signal aberrations. He can either provide a complex control system which allows for equalization of almost any frequency conditions, or he can reduce the flexibility of control, choosing compromise response curve parameters which are capable of producing approximate compensation for those conditions most likely to be encountered. The degree of tone-control complexity accepted in non-professional equipment has increased quite a bit since the day of the single treble-cut switch, and the most common arrangement in current audio amplifiers is a two-control, continuously variable system. This allows progressive cut or boost of either bass or treble, but no independent control of the reference frequencies at which the response curves begin their slopes.

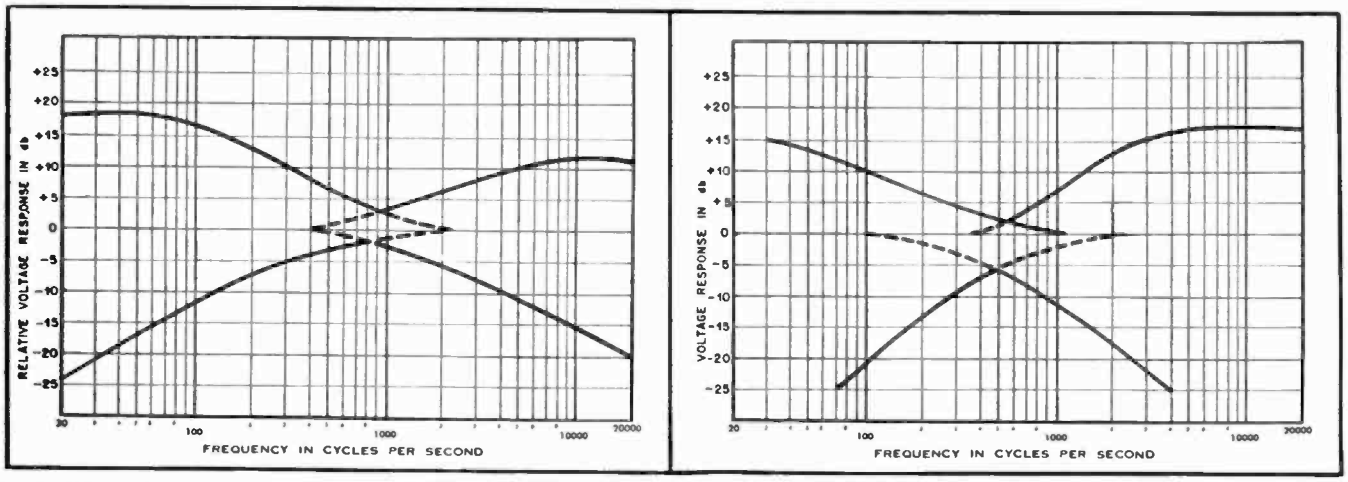

Fig. 1. (A), Voltage response curves of a typical commercial tone control circuit of the R-C type, maximum positions. (B), Voltage response curve of a commercial tone control circuit of the L-R-C type, maximum positions.

* Woodstock, New York.

[1] The transition frequency is the intersection of the theoretical linear slope of the response curve (a slope which the actual curve only approaches) with the frequency axis—see *Fig. 5*.

A circuit which introduces a progressive response slope rather than uniform elevation or depression of a whole band of frequencies, besides being of far simpler design, provides the desired form of compensation in most applications. It is common although not universal practice for dual tone controls to use approximately the same transition frequency region, usually at or below 1,000 cps, for both bass and treble variation. *Figure* 1 shows the response curves of two such tone control circuits employed commercially.

The justification for selecting this

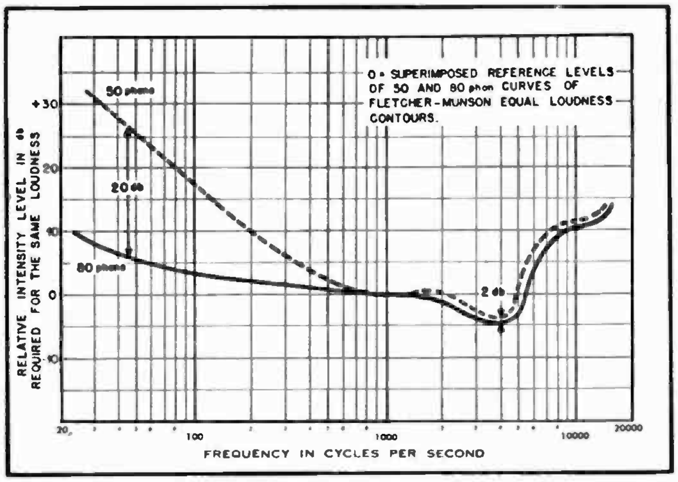

Fig. 4. The 50- and 80-phon equal loudness contours (from Fletcher and Munson) superimposed.

reference frequency region is that it is considered psychologically "neutral" in pitch. It has been pointed out that 800 cps is the geometric mean between 40 and 16,000 cps, frequencies which may be taken as nominal limits of hearing under average conditions. Since the perception of frequency, like that of amplitude, closely follows the Weber-Fechner law (in that the degree of sensation varies logarithmically with the stimulus) the geometric and not the arithmetic mean of the audible frequency spectrum is its psychological mid-point. Thus there are about four and one-half octaves of useful audio frequencies on each side of 800 cps.

If we assume, however, that the operator's adjustment of tone controls is in the nature of a search for maximum fidelity to the perceived original, the psychological mid-point ceases to have much significance. Tone control becomes tone compensation, and the problem of response-curve parameters revolves about the question of what the controller must be equipped to compensate for. In the following discussion it will be assumed that the amplifier is being designed as an independent unit, and that the brands of components with which it is to be used are not known.

Treble Boost

The most common purposes for which treble boost will be needed are:

1. Compensation for treble deficiencies in associated reproducing equipment.
2. Compensation for treble deficiencies in program material.
3. Compensation for discriminatory acoustic absorption with a frequency characteristic different from that of the hall or studio in which the sound originated.

An additional purpose for which treble boost has on occasion been considered necessary, but is not, is compensation for the variation in hearing frequency characteristics associated with changes of sound intensity level. This factor, referred to as the Fletcher-Munson effect, will be discussed in the paragraphs devoted to bass boost.

Equipment treble deficiencies are usually most severe in electro-acoustic devices such as loudspeakers, pickups, and recording heads, but also occur in coupling circuits in audio or intermediate-frequency amplifiers.

The worst offender in a given system is ordinarily the loudspeaker. In *Fig.* 2 manufacturers' published on-axis response curves for six speakers in different price ranges are plotted. Although the performance represented by these curves will vary greatly under different acoustical conditions the graphs may be taken as indicators of a general trend. There is one feature which all may be seen to have in common, and which is characteristic of the great majority of cone loudspeakers with high-frequency droop; the frequency region of the first two octaves above 1,000 cps, far from being attenuated, is accentuated (because of the new resonances introduced by cone break-up), and treble droop does not begin before 4,000 cps or higher. This fact has a significance beyond the obvious implication concerning compensation for speaker deficiencies. Any losses in the first two treble octaves which are likely to be met with from other causes will probably be compensated or even over-compensated by speaker characteristics.

Crystal phonograph pickups have a typical velocity response above a few hundred cps which decreases with frequency quite regularly up to a rather sharp cut-off somewhere between 4,000 and 10,000 cps. When this regular droop does not conform to the desired recording characteristic further compensation is properly provided by a fixed R-C network rather than by the tone control. The comparatively smooth and accurately predictable slope lends itself to fixed equalization, and the recommended circuits or necessary data for their design are usually readily available from the manufacturer. But whether or not such a fixed network is provided, the treble boost control cannot be designed

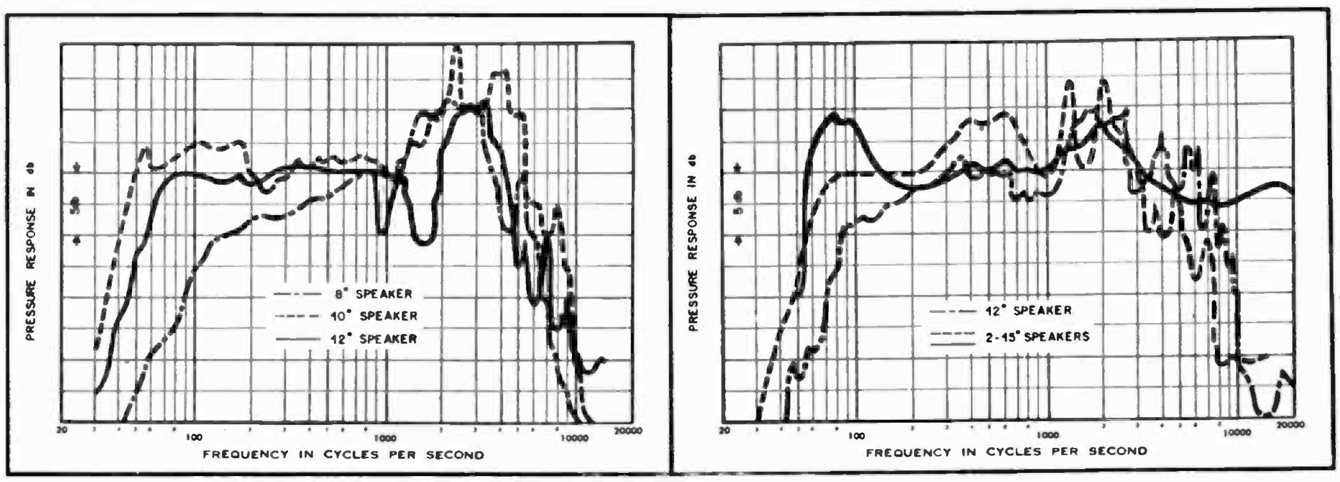

Fig. 2, (A) and (B). Pressure response curves of three commercial speakers in a price range of $20 to $150.

for correct crystal pickup equalization except for the second droop, because the required transition frequency would be well into the bass region.

The contribution of high-frequency losses by other components cannot, of course, be predicted, except as to one factor; it may be expected that losses will be confined to frequencies above the second treble octave. A survey of circuit components, recording heads, etc., will indicate that it is rare for treble droop to set in before 4,000 cps or so. Even AM broadcast band i.f. transformers provide, at worst, relatively even coupling to 3,000 cps.

Treble deficiencies in program material may result from low-grade studio equipment, from transmission circuits, and from old records. Such losses are almost always associated exclusively with the third and/or fourth treble octaves.

The writer has not found a quantitative study which compares the frequency transmission of typical living rooms with that of halls or studios, although methods for room transmission measurements have been outlined.[2] Therefore no comment will be made on the subject other than to mention the fact that room acoustics—notably the reverberation time *versus* frequency relationship—may be a factor requiring tone compensation. Reference is made to this factor under the heading of treble boost merely on the basis of subjective experience, but undoubtedly other compensations are also involved.

When treble boost is needed, then, a high transition frequency is usually desirable. The use of a lower transition frequency for the sake of increasing the amount of boost available at the upper end might prove satisfactory if it were not for the marked tendency of loudspeakers to emphasize the frequency region of the first few thousand cps. A transition frequency of 1,000 cps or lower may cause treble boost to accentuate a shrillness towards which the speaker performance is already inclined, and needed boost at frequencies above 4,000 cps may carry such a penalty that the compensation will not be used. This unhappy effect is illustrated by *Fig.* 3, in which the tone-control and speaker-response curves of two high-grade commercial units are combined.

Determination of an optimum transition frequency that will best suit the various requirements of treble boost cannot be made with precision, but it would seem that present commercial practice makes use of a frequency which is at least two octaves too low. The operator of the set may be impressed with the dramatic power of his treble boost but may still be loath to use it. (Some British manufacturers lean towards higher transition frequencies for treble emphasis—one manufacturer uses 3,500 cps.)

A single R-C network can approach 6 db per octave in rate of boost. The simplicity of the single network is one good reason for accepting this slope as the maximum, and it will prove ample for most purposes, particularly in view of the fact that treble emphasis brings to the fore harmonic distortion in the higher ranges. If the transition frequency is chosen as 3,200 cps—two octaves above 800 cps—an insertion loss of 20 db will produce something less than 12 db of maximum boost at 13,000 cps, including about 3 db of boost at the reference frequency itself. These characteristics seem to represent a reasonable compromise between the requirements of all factors. The use of an even higher transition frequency might be desirable, but would place a greater limitation on the maximum boost of useful frequencies that could be achieved with a single R-C network.

Treble Attenuation

Treble attenuation is required for:

1. Compensation for varying treble pre-emphasis in recording.
2. Reduction of record suface noise and high-frequency distortion.
3. Compensation for rising treble response of associated reproducing or studio equipment.
4. Tonal balance against a thin bass.

Treble pre-emphasis in recording varies considerably, and there is no standard, either of transition frequency or rate of boost, to which recording

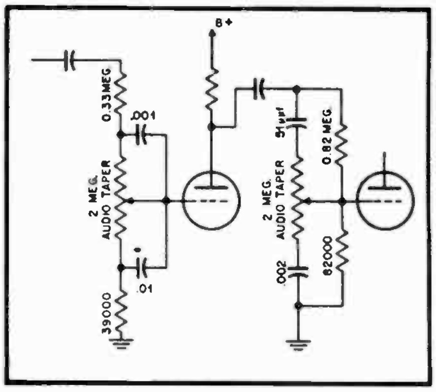

Fig. 6. Circuit values to approximate the curves of Fig. 5. As many of the values of the original Æ circuit as possible have been retained.

companies subscribe. Table 1 lists some of the different ways in which the treble spectrum is or has been treated by record manufacturers.

Although treble recording characteristics are ideally equalized at the pickup, variable pickup equalizers are still the exception rather than the rule, and the tone-control designer must assume that at least some of the burden will fall on the treble-cut control. If the setting for minimum treble furnishes close to 6 db of attenuation per octave from a transition frequency of 2,000 cps, approximate compensation for any of the recording pre-emphasis curves can be achieved.

TABLE 1

Typical Transition Frequencies

Records	Transition Frequency (cps)	Rate of Boost (db/octave)
AES Standard Curve	2500	6
Columbia 78 & 33 1/3[3]	1590	6
RCA 78[3]	1000	2.5
London ffrr 78[3]	3000	3
London 33 1/3[3]	3000	6
EMI 78[4]	—	none
Older records	—	none

Settings of the treble control which yield less than maximum attenuation will also, in most circuits, automatically shift the transition frequency higher, correcting equalization for some of the records with more gradual pre-emphasis slopes. The desirability of equalization for treble recording characteristics prior to the tone control is emphasized by the differences between the transition frequencies of some of these recording curves and the frequencies at which the equipment deficiencies previously discussed introduce treble losses. For example, an improperly equalized disc which has been recorded with a given treble characteristic can present an unsatisfying choice between shrill and muffled reproduction, because the tone

[2] E. C. Wente, "The characteristics of sound transmission in rooms," *J. Acous. Soc. Am.*, 7, p. 134, Oct., 1935.

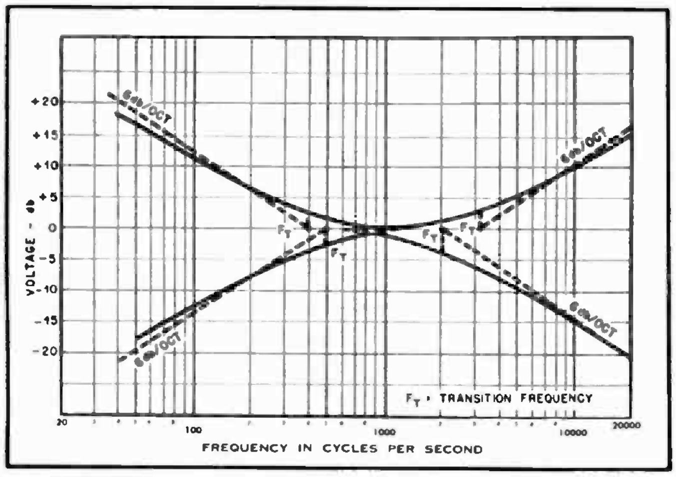

Fig. 5. Suggested tone control frequency characteristics, controls at maximum positions.

[3] Paul W. St. George and Benjamin B. Drisko, "Versatile phonograph preamplifier," Audio Engineering, 33, p. 14, March, 1949.

[4] D. T. N. Williamson, "High-quality amplifier modifications," *Wireless World*, 58, p. 173, May, 1952.

control may not be able to compensate for the characteristics of both the record and the reproducing equipment.

It has been established by several investigators that record-surface noise is fairly evenly distributed over the frequency spectrum on the basis of energy content per cycle. Since each arithmetic frequency interval has a similar amount of noise energy, the noise content will increase with each successively higher octave.[5] It is therefore correct to think of perceived surface noise as increasing with frequency, and compensation for treble pre-emphasis in recording is very effective in reducing scratch, as the system intends it to be.

There is however, an advantage in being able to introduce sharp cut-offs at given points of the frequency band, since records which have little or no frequency content above the cut-off point can then have their surface noise reduced without the signal being severely penalized. Some types of distoration are also most pronounced in the range above 5,000 cps, and sharp cut-off helps reduce such effects with least change of the signal. Sharp cut-off at high frequencies, however, is incompatible with the other duties of the treble tone control, and should be accomplished by separate networks. The general tone control must be limited in its scratch reducing role to a more gradual attenuation of the treble band.

Certain microphones, if incorrectly equalized at the studio, have a rising treble response above one or two thousand cps. Attention is occasionally called to this characteristic when broadcast studio use of an incorrectly compensated microphone produces over-crisp and "hissy" speech. The compensation required for this type of emphasis is compatible with that already planned for records.

Tonal balance of bass and treble for the most satisfactory over-all result is an inherently subjective problem. Like compensation for the Fletcher-Munson effect, it involves a decrease of objective fidelity for the sake of an increase in apparent realism. The fact that the problem is one of perception rather than of reality indicates that investigation requires a statistical technique.[6]

Discussions of aural balance[7] have indicated that approximately equal frequency distortion, geometrically relative to the frequency mid-point, is more desirable than unequal frequency distortion, but that there is considerable latitude in design for balanced response. The parameters of treble attenuation for achieving careful balance would be determined by a study of bass deficiencies likely to be met with. The treble attenuation discussed in previous paragraphs is fairly symmetrical to most of these low-frequency deficiencies.

It will be seen that the transition frequency suggested for treble attenuation is different from that for treble boost, and conditions are likely to exist in which both are required simultaneously. Such simultaneous compensation, however is inherently barred when a single control is used for both boost and cut.

Bass Boost

The conditions requiring bass boost are:

1. Decreased hearing sensitivity to bass frequencies at low sound intensities (Fletcher-Munson effect).
2. Recording characteristics whose bass turnover frequency is higher than the one for which the repoducing equipment is designed.
3. Bass deficiencies in records.
4. Bass deficiencies in reproducing or studio equipment.

The well known family of equal loudness contours published by Fletcher and Munson makes it evident that the apparent frequency distribution of energy in given program material will vary greatly at different intensity levels. If the amplifier has a correctly designed compensating network associated with its volume control this effect will be counteracted, but the volume control setting required for a desired intensity level is not necessarily an accurate index of the intensity level of the original program, and further adjustment may be necessary. The electrical level of an input signal does not have a constant relationship, in different program material, to the sound level which it represents.

The purpose of volume compensation is not, of course, to straighten out the curve of frequency perception, (that's the way the music sounds in the concert hall) but to shape this curve at the reproduced intensity level so that the perceived frequency distribution is similar to the perceived distribution at the original intensity. We may take 80 db[8] as the average level of a 75-piece orchestra heard from a good seat, and 50 db[9] as the lowest level at which this music is likely to be reproduced, with any concern for quality, in the living room. Superimposing the two appropriate curves on the same horizontal axis (*Fig.* 4), it will be seen that in order to achieve the original perceived frequency distribution in reproduction at the 50-db level, bass boost at slightly more than 6 db per octave, with a transition frequency of 400 cps, is required. This is, of course, for the extreme case of a 30-db difference between the original and the reproduced level. At this 30-db difference the bass boost required during orchestral peaks, when both levels may be increased by 20 db, is much less; during very soft passages the boost required will be more. We must work on the basis of the average levels.

The 50- and 80-phon curves, and the 60- and 70-phon curves in between, are practically identical in shape and slope from 500 cps up, meaning that a reduction of intensity level from 80 db to 50 db will produce no significant change in the apparent frequency distribution in the treble region. Even if it were desired to compensate for the 2-db maximum loss in treble, the ordinary R-C network could not produce the necessary shape of response curve, which may be read from *Fig.* 5 as a uniformly elevated plateau over most of the treble spectrum. Correct volume compensation should therefore involve no adjustment of treble frequencies.

If the record reproducing equipment has only a single turnover frequency a compromise value of 500 to 600 cps is usually chosen. Some records have been made with higher turnover frequencies as high as 800 cps. Thus a moderate amount of boost to add to that of the fixed equalization will on occasion be called for. Such equalization, however, is required by a relatively small number of records.

Acoustically recorded discs, and some electrically recorded ones, have a thin bass because of weaknesses in recording equipment and techniques. These ordinarily require boost below two or three hundred cps, although the drop in bass response is in general too sharp for adequate compensation.

Bass deficiencies in reproducing equipment, like treble deficiencies, tend to occur towards the extreme of the frequency scale. Most moderate quality loudspeakers (see *Fig.* 2), pickups, etc., do not show a significant drop in bass response until frequencies below 100 cps, and then with sharply dropping curves. Here too compensation cannot be adequate.

Adjustment for the Fletcher-Munson effect is probably the most frequent function of bass boost. The transition frequency required for this equalization lies between that for a high turnover frequency and for record and equipment deficiencies, and the compromise parameters that appear most reasonable to the writer include a transition frequency one octave down from the spectrum mid-point, or 400 cps, and the standard maximum boost rate of 6 db per octave.

[5] B. B. Bauer, "Crystal pickup compensation circuits," *Electronics*, 17, p. 128, Nov., 1945.

[6] It is suggested that a useful approach would be to compile judgment data in which the jury compares various conditions of tonal balance, both symmetrical and asymmetrical to 800 cps, with the full range of sound, indicating which condition seems more like the undistorted one. (Asking the jury to indicate preference between balanced and unbalanced conditions, without an ever-present standard, involves the danger of measuring factors other than apparent frequency distortion.)

[7] Hugh S. Knowles, "Loudspeakers and room acoustics," *The Radio Engineering Handbook*, Keith Henney, editor, p. 881, 3rd edition, 1941.

[8] Ibid.

[9] *"Frequency Range and Power Considerations in Music Reproduction,"* Jensen Technical Monograph No. 3, Jensen Manufacturing Company. Note that a level of 50 db is less than 10 db above the average random noise level in a city apartment.

Flexible Tone Control Circuit

BASIL T. BARBER*

The advantages of variable inflection points in tone controls are well recognized, but the circuit complications and additional expense does not always justify their use. The author Americanizes a previously published circuit which is relatively simple.

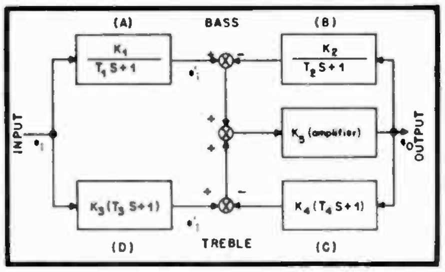

Fig. 1. Simplified block diagram of the tone control which employs negative feedback to provide variable inflection points. Idealized lead and lag networks are shown.

RECENT INVESTIGATIONS on the subject of tone controls indicate the desirability of having the transition frequencies variable, with the amount and direction of variation determined by the degree of boost or attenuation required.

The analysis and determination of the varying parameters and their range are ably covered in Mr. Villchur's article.[1] Most tone-control circuits available at present have a single transition frequency between 800 and 1000 cps, which remains essentially constant at any position of the controls and their flexibility and effectiveness is therefore limited.

A method of tone compensation is presented in this article offering a continuously variable transition frequency and a number of other advantages. The original article was written by P. J. Baxandall[2] and for a detailed explanation of the circuit the reader is referred to his article.

One way of looking at this circuit is to analyze it in terms of Laplace Transforms. Oversimplifying the original schematic, the circuit can be broken down into two lead and two lag networks, as shown in *Fig.* 1. The lead network (C), being inside a negative feedback loop, becomes a lag network and similarly the lag network (B) becomes a lead network.[3] The net curve will therefore be either a boost or an attenuation, as shown in *Fig.* 2, depending on the relative position of the time constants T_a, T_b, T_c and T_d. When $T_a = T_b$ and $T_c = T_d$, the curves cancel each other producing a flat frequency response.

Figure 3 presents the tone-control circuit in detail. It is similar to the original circuit,[4] but with parameters slightly modified to make the circuit adaptable to American tubes and components.

Figure 4 shows the actual frequency response of the schematic of *Fig.* 3. The low-frequency turnover is made variable from 800 to about 100 cps, depending on the degree of bass boost or attenuation required. The high-frequency turnover varies similarly from 1000 to 8000 cps, depending on the degree of treble boost or attenuation required. These curves

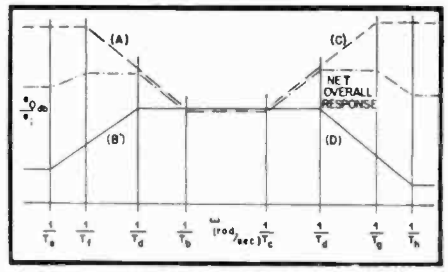

Fig. 2. Frequency response of the actual lead and lag networks of Fig. 1. B' and C' are curves B and C after feedback.

should therefore prove to be effective in compensating certain deficiencies of an audio system explained in Villchur's article. Specifically, "partial" speaker compensation can be accomplished without any boominess (especially on male voices), and without high-frequency distortion.

The circuit presented has, in addition, a number of other advantages which are equally important. In most tone-control circuits at present, the apparent boost is attained at the expense of an equal attenuation at the center frequency. An additional gain equal to the boost attainable must therefore be provided. In a "flat" position, there is unity gain *plus* all the distortion generated by the stage of amplification. In the proposed circuit, the loss is in the form of negative feedback with its associated advantages of reduction of distortion, impedance, and noise, and the increase of the linearity and frequency response of the circuit. For instance, if we assume that a boost

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[1] "The selection of tone-control parameters," AUDIO ENGINEERING, March, 1953.

[2] *Wireless World*, October, 1952.

[3] See Appendix.

[4] Ref. 2, p. 404, *Fig.* 6.

SELECTION

(*from preceding page*)

Bass Attenuation

Bass attenuation may be required for the following conditions:

1. Recording characteristics with a bass turnover frequency lower than that of the reproducing equipment.
2. Accentuation and distortion of bass frequencies by reproducing equipment.
3. A weak treble which creates tonal imbalance.

The required compensation for equipment designed with a 500-cps turnover, and playing a record with a turnover frequency of 250, 300 or 400 cps (all of which values have been used) may be approximately achieved by a gentle attenuation of about 2 db per octave from 750 cps down, or by a sharper downward slope from a lower transition frequency of 500 cps. The second method is more consistent with the other requirements of bass cut.

Bass accentuation in reproducing equipment is most often associated with acoustical resonance of an open speaker enclosure at some frequency below 200 cps and with mechanical-acoustical resonance of the speaker system below 100 cps. The effects of tone-arm resonance and turntable rumble usually occur below 50 cps. As in the case of low-frequency boost, we cannot furnish accurate compensation for equipment frailties at the extreme low end, but we can alleviate the condition in some measure.

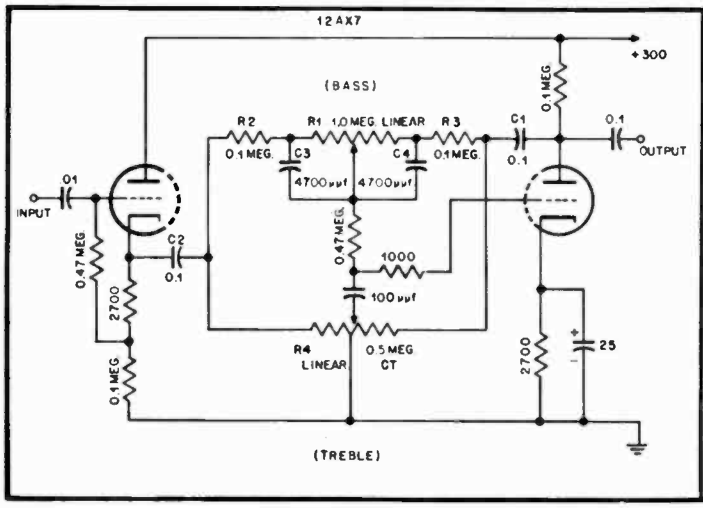
Aural balance of a weak treble appears to demand a low transition frequency. Probable treble droop occurs, as we have seen, at least two octaves above 800 cps, and geometrically symmetrical bass losses would begin at about 200 cps.

A reference frequency and slope to compromise between the various requirements for which bass attenuation will be needed are: transition frequency 500 cps, maximum slope, 6 db per octave.

Conclusion

Figure 5 is a graph of tone-control frequency characteristics chosen on the basis of the above discussion. The parameters vary from many of those in common commercial use in that the reference frequencies are shifted away from the spectrum mid-point, about one octave down for the bass, and two octaves up for treble boost. As a consequence the total amount of control is somewhat less than is often provided.

Figure 6 is a tone control circuit with values assigned to approximate these curves. Some allowance has been made for the fact that the transition frequencies are affected by the degree of boost or cut used by the operator.



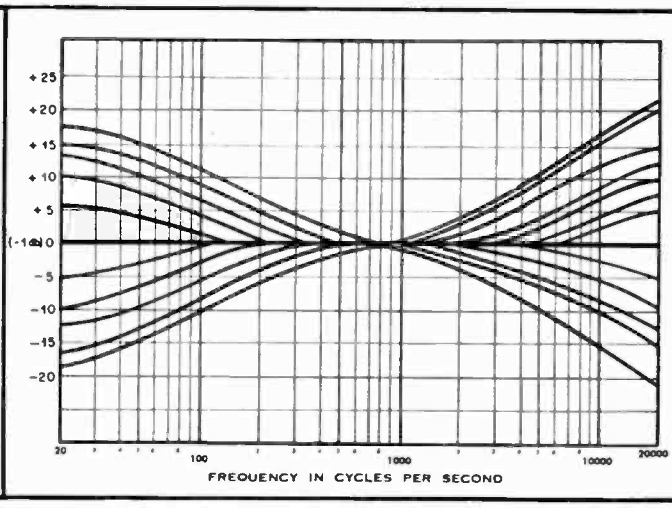


Fig. 3 (left). Over-all schematic of the circuit as modified for American tube types. Fig. 4 (right). Response curves obtainable with the feedback type tone control.

of 10 db is required at both extremes of the audio spectrum, then 7 db of negative feedback will still remain at both extremes, and 17 db at the mid-frequencies. At "flat" position, the circuit has a loss of 1 db[5] which is negligible. The circuit can therefore be inserted anywhere with no complications.

The only disadvantage is that the input to the tone control must come from a low-impedance source but not necessarily from a cathode follower as shown in *Fig. 3*. A source impedance of 10,000 to 15,000 ohms will have no adverse effect, and this impedance can be readily attained from a low-mu tube such as a 6SN7 or 12AU7.

Because to the large amount of negative feedback employed, the output source impedance is low, giving the circuit all the advantages of a cathode-follower output stage.

Use and Abuse of the Tone Controls

While the writer is not naïve to the point of believing in "ideal curves," flat speakers, and acoustical heavens in general, it is believed that in a reasonably high-quality preamplifier-equalizer the tone controls should not be called upon to perform functions which they were not originally designed to perform.

Following are the areas where abuse is most likely to occur:

Preamplifier

There are scores of well designed circuits which offer adequate flexibility and adaptability to meet any record characteristics and their deviations without resorting to tone-control compensation. Since the low-frequency turnovers and the high-frequency pre-emphasis fall well below and above the "center" frequency of the tone controls, no adequate compensation can be attained.

As for reducing any record noise, hiss, or rumble, the tone controls are not especially effective since, at best, most of them have an attenuation rate of 6 db/octave, hardly adequate for any effective suppression without sacrificing the musical content of the program.

Loudness Control

Compensation for the Fletcher-Munson curves with the tone controls would seem to make them perform a function actually belonging to a loudness control. In addition, once the tone controls are set on the reciprocal of the Fletcher-Munson curves, their usefulness for any other function will be restricted, if not nullified.

Speaker

The most serious misunderstanding seems to be on the question of speaker compensation. It is well known by now that phase as well as amplitude response of a speaker plays an important role on the quality of music reproduction. A speaker, being essentially a non-linear electromechanical transducer, has a phase response which includes several reversals, rendering any effects for effective compensation rather futile, especially with simple networks. Circuits have been designed which effectively compensate for some speaker deficiencies, using variable slope as well as variable turnover frequency, but their cost is usually several times the price of the average speaker, making any impending "Fire Sale" on high-quality loudspeakers highly improbable.

Why use tone controls at all then? They fill a definite need and most of their functions are covered in Mr. Villchur's article and can be summarized as providing:

1. Over-all tonal balance of the complete system.
2. Compensation for any acoustical such as resonance, for example.
3. Compensation for any deficiencies of studio equipment, means of transmission, or for faulty records.
4. Bass reduction to eliminate boominess of male voices, especially on low settings of the loudness control.

APPENDIX

Negative feedback can be employed to transform a network into its reciprocal, as shown below. Although the following derivations are for ideal lead and lag networks, they apply equally on actual networks having two break frequencies, both being important in determining the over-all frequency response of the circuit.

1. *Transformation of a Lag into a Lead Network (Bass).*

Referring to the upper section of *Fig. 1* we have,

$$\left[e_i - \frac{K_t e_o}{T_t S + 1}\right] K_s = e_o$$

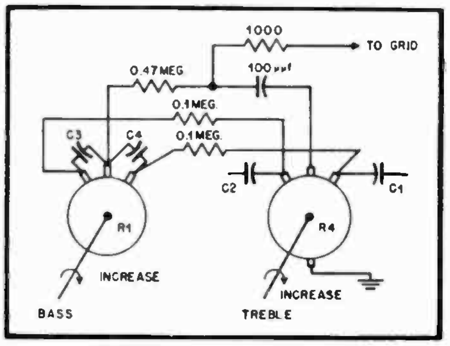


Fig. 5. Suggested wiring arrangement for the tone control. The two potentiometers R_1 and R_4 could easily be combined as a concentric control.

from which

$$\frac{e_o}{e_i} = \frac{K_s(TS+1)}{T_tS + 1 + K_tK_s} = \frac{\dfrac{K_s}{1+K_tK_s}(T_t+1)}{\left(\dfrac{1}{1+K_tK_s}\right)T_tS + 1}$$

The original lag (B), has been therefore transformed into a lead network (B′) having two break frequencies, $1 + K_tK_s$ apart.

2. *Transformation of a Lead into a Lag Network (Treble)*

Referring to the lower section of *Fig. 1* we have,

$$[e_i' - [K_t(T_tS+1)]e_o]K_s = e_o$$

from which

$$\frac{e_o}{e_i} = \frac{K_s}{K_tK_sT_tS + 1 + K_tK_s}$$

$$= \frac{\dfrac{K_s}{1+K_tK_s}}{\left(\dfrac{K_tK_s}{1+K_tK_s}\right)T_tS + 1}$$

The original lead network (C) has been transformed into a lag network (C′).

The method of combining the two original networks (A) and (D) and the two modified networks (B′) and (C′) to obtain an overall response (boost in this case) is shown in *Fig. 2*.

When $T_b > T_a$ and $T_d > T_c$, then an attenuation of both extremes is attained, while at $T_a = T_b$ and $T_c = T_d$, we have a completely flat frequency response.

[5] Since the total gain in a negative-feedback circuit is equal to the forward gain divided by one plus the loop gain. In our case, this ratio is less than one. In addition, there is a slight reduction in gain due to the cathode-follower input, but the circuit parameters have been selected to keep the total loss less than 1 db.

Hi-Fidelity Phonograph Preamplifier Design

R. H. BROWN*

A comprehensive discussion of the principles of determining component values to provide the compensation required, combined with practical pointers on construction.

Several excellent phonograph preamplifier circuits have appeared in the literature but it is difficult for the individual who is not a full-fledged audio engineer to find the information he may need for designing a high-quality preamplifier to meet his individual preferences, or to make desirable changes intelligently in equipment already on hand. This article aims to set forth in a summary fashion basic material which a technically minded audio hobbyist or an amateur engineer would need in the design or revision of quality preamplifying and compensating equipment. Rather than attempt a comprehensive survey of the problems and theory of preamplifier and compensation circuit design, attention will be confined to straightforward basic circuits.

Basic Amplifier With Degenerative Bass Boost

Figure 1 shows the basic circuit around which most high-quality phonograph preamplifiers are designed. Resistor R_2 provides degenerative feedback to set the amplifier gain to the desired value for high frequencies, reduce amplifier distortion in the high-frequency range, and extend the high-frequency range beyond that which could be obtained from the basic amplifier without feedback. R_3 provides degenerative feedback to set the maximum value of bass boost, reduce amplifier distortion at the extreme low frequencies, and extend the low-frequency range below that which could be obtained from the basic amplifier without feedback. R_3 may be omitted (made infinite) when compensation is desired down to a frequency requiring the full gain of the basic amplifier. C_f provides 6-db-per-octave bass boost as figured from the turnover frequency for which C_f is selected.

The first step in the design of a preamplifier is to design a basic amplifier which has sufficient gain to provide down to the lowest frequency of interest compensation for the highest turnover frequency to be used. Information on electrically correct turnover frequencies may be found in the literature, and it will be observed that there are combinations of listening conditions, listener preference, and recording characteristics which may require turnover frequencies as high as 900 cps. As a 900-cps turnover requires approximately 29 db boost at 30 cps, the preamplifier gain at 30 cps must be the antilog of 29/20 or 28.2 times as great as in the high-frequency range if bass compensation is to be provided down to 30 cps for a 900-cps turnover. Since due to the action of C_f the gain only approaches the full basic amplifier gain as the frequency is reduced, in order to obtain good compensation down to a given frequency it is necessary to design for a frequency one octave lower. Thus for good compensation down to 30 cps for a 900-cps turnover one must have a basic amplifier with a gain at least 56 times greater than that required in the high-frequency range.

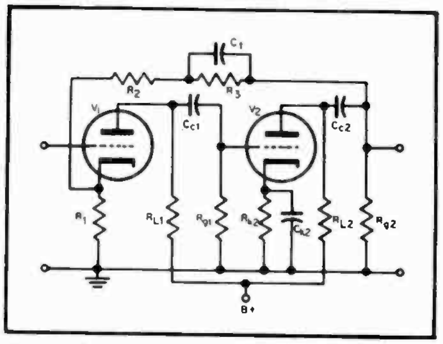


Fig. 1. Basic degenerative-bass-boost amplifier circuit. V_1 and V_2 and associated components, excluding feedback elements R_2, R_3, and C_f comprise the basic amplifier. Triodes are shown for simplicity.

A preamplifier designed for use with a "Williamson" type power amplifier and a G.E. magnetic cartridge should provide between one and two volts output for a 10-millivolt input at 1000 cps. If this preamplifier is to provide good compensation down to 30 cps for turnovers up to 900 cps it must have a basic amplifier gain of between 5600 and 11,200, with R_2 selected to obtain a gain of between 100 and 200 in the high-frequency range. If no turnovers above 500 cps are to be used, good compensation to 30 cps may be secured with a basic amplifier gain of between 3100 and 6300. High-level cartridges, such as Clarkstan and Pickering, require minimum preamplifier gains approximately only one-sixth as great.

The selection of V_1, V_2, and associated components is best made from reference to the resistance-coupled amplifier data supplied by tube manufacturers. Because of the unbypassed cathode resistor the gain provided by V_1 will be

$$A_1 = \left[\frac{A_1'}{1 + \left(\frac{R_1}{R_{L_1}}\right) A_1'}\right], \qquad (1)$$

where A_1' is the gain provided by V_1 with a completely bypassed cathode. C_{c_1}, C_{c_2}, C_{k_2}, and screen bypass capacitors if pentodes are used, must be selected to provide essentially uniform gain down to the lowest frequency f_L for which full compensation is desired. To accomplish this it is desirable to choose the value of C_c (in farads) so that the product $f_L C_c R_g$ is between one and two, and to choose C_{k_2} so that the product $f_L C_{k_2}$ is between one and two times the mutual conductance of V_2. In many cases adequate gain may be obtained without use of C_{k_2}. Data required for the selection of a pentode screen bypass capacitor is ordinarily not available, but one can usually make a satisfactory choice by requiring that the product of the bypass capacitance (in farads), the lowest frequency of interest, and the plate resistance of the tube when triode connected have a value between one and two. If it is not convenient to use capacitances as large as those required by the foregoing conditions, the situation may be relieved by providing around 10 db of feedback through R_2.

Once the basic amplifier has been designed to provide adequate gain for the maximum bass boost desired, the next step is to select R_2 so that the amplifier will give the right amount of gain for the high frequencies. At high frequencies (5000 to 10,000 cps) the gain is given by

$$A_{HF} = \left[\frac{A'_{HF}}{1 + \frac{R_1 A'_{HF}}{R_1 + R_2}}\right], \qquad (2)$$

where A'_{HF} is the gain of the basic amplifier when a resistance equal to $R_1 + R_2$ is placed in parallel with R_{L_2}. The gain provided by V_2 under these circumstances is equal to the mutual conductance of V_2 multiplied by a resistance equal to the parallel combination of the plate resistance of V_2, $R_1 + R_2$, R_{L_2}, and R_{g_2}. To obtain accurate 6-db-per-octave bass boost it is necessary for A'_{HF} to be large enough to make negligible the unity in the denominator of Eq. (2)—i.e., for A'_{HF} to be greater than $10R_1/(R_1 + R_2)$. When this

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condition is satisfied Eq. (2) becomes

$$A_{HF} = \frac{R_1 + R_2}{R_1}. \tag{3}$$

Since R_1 will usually be chosen from resistance-coupled amplifier data for proper biasing of V_1, one may write the following equation for R_2,

$$R_2 = R_1(A_{HF} - 1), \tag{4}$$

where A_{HF} is now the required high-frequency gain—i.e., the gain without bass boost. If the loading effect of $R_1 + R_2$ keeps A'_{HF} from being large enough to render insignificant the unity in the denominator of Eq. (2), the situation may be remedied by connecting R_3 and C_t to an unbypassed cathode of a stage following V_2 (possibly a cathode follower output for the preamplifier).

The maximum gain available from the amplifier may be designated A_{LF} and is given by

$$A_{LF} = \left| \frac{A'}{1 + \dfrac{R_1 A'}{R_1 + R_2 + R_3}} \right|, \tag{5}$$

where A' refers to the normal gain of the basic amplifier without feedback. In practice the coupling and bypass capacitors often prevent realization of the full value of A_{LF}.

From the fact that a simple resistance-capacitance arrangement which will provide 6-db-per-octave boost computed from a turnover frequency f_t must give a 3 db boost at f_t,[1] one obtains the following relation for the selection of C_t.

$$C_t = \frac{1}{2\pi f_t (R_1 + R_2)}. \tag{6}$$

[1] The boost will be very close to 1 db at $2 f_t$, 3 db at f_t, 7 db at $\frac{1}{2} f_t$, and 12 db at $\frac{1}{4} f_t$.

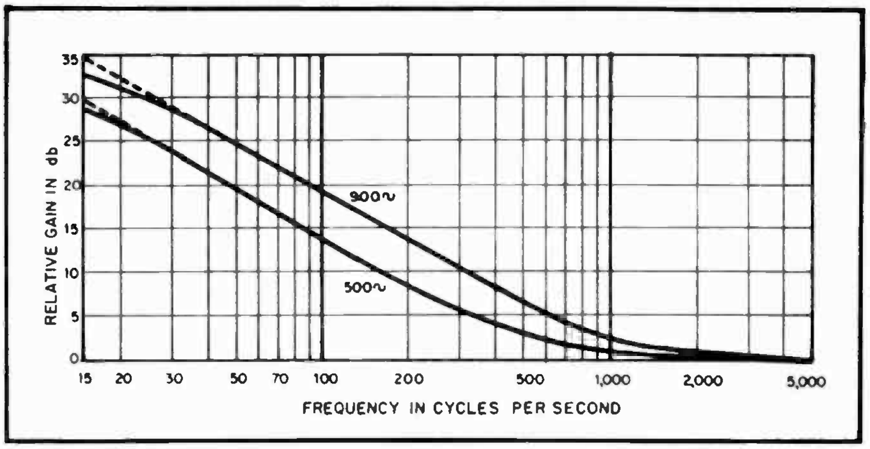


Fig. 2. Bass performance of pentode amplifier example discussed in text. Dotted lines indicate idealized 6-db-per-octave slope.

It will now be instructive to see how these principles are applied in selecting the components for a practical high-fidelity preamplifier for use with a GE cartridge and a "Williamson" type power amplifier. On consulting the resistance-coupled amplifier charts in a receiving tube manual one finds that a basic amplifier gain A' of approximately 18,000 may be obtained by using for V_1 a 6SJ7 with 180-v. E_{bb}, 0.5-meg. R_{L_1}, 2.4-meg. screen resistor, 2.0-meg. R_{g_1}, and 2410-ohm R_1 to obtain a gain of 95; and for V_2 a 6SJ7 with 180-v. E_{bb}, 0.5-meg. R_{L_2}, 2.2-meg. screen resistor, 1.0-meg. R_{g_2}, and 2180-ohm R_{k_2} to obtain a gain of 192.

For 10-db minimum feedback $A_{LF} =$ 1800/(antilog 10/20) which is 5700. From Eq. (5) one finds that this requires 20 megs. for R_3, $R_1 + R_2$ being negligible in comparison with R_3. For $A_{HF} = 100$, Eq. (4) requires a value of 0.241 meg. for R_2. This provides at high frequencies a 45 db feedback in addition to the 6.8 db of current feedback on V_1 due to its unbypassed cathode resistor. For 900 and 500 cps turnovers Eq. (6) requires values for C_t of 740 and 1310 μμf respectively.

The actual performance of this preamplifier is shown in *Fig. 2*. The value chosen for A_{LF} was 35.1 db above that chosen for A_{HF}, an amount just equal to the boost required at 15 cps by a 900 cps turnover. The necessity for designing to a frequency one octave below the lowest frequency to which full compensation is desired is illustrated by the 900 cps turnover curve in *Fig. 2*.

In a preamplifier designed for use with a high-level magnetic cartridge, a value of but 15 for A_{HF} would be adequate. One could use the basic 6SJ7 amplifier discussed above with 33,700

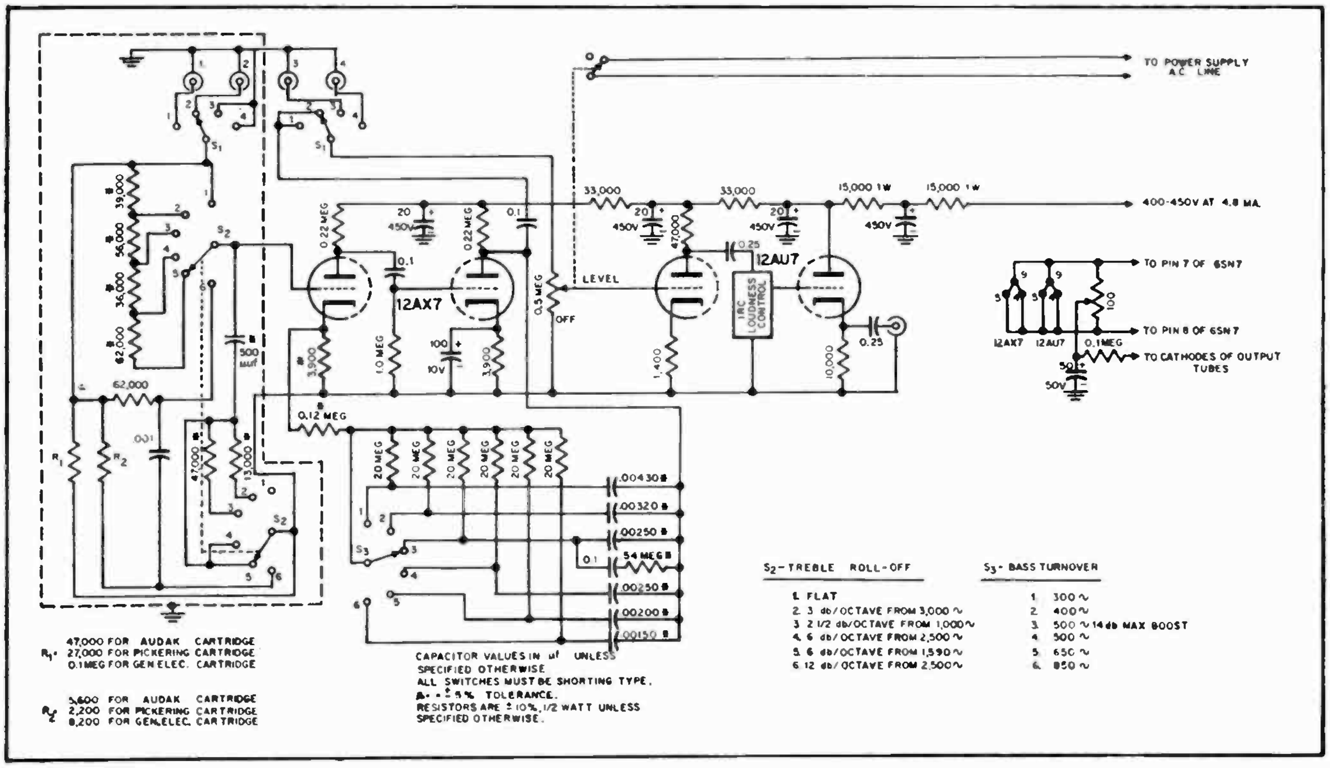


Fig. 3. Complete schematic of author's preamplifier with six bass turnover conditions and six roll-off positions.

ohms for R_2 and 4 megs. for R_3 to obtain good compensation down to 20 cps for a 900-cps turnover. Actually the 6SJ7 is a poor choice for V_1, for unless d.c. is used on the heaters one is likely to experience hum difficulties. The 6J7 would be more suitable in this respect, even though it will not provide quite as much gain. The high-level cartridge makes possible good bass compensation with a simple preamplifier using a twin triode such as the 12AX7, 6SL7, or 7F7. Triodes may be used in a preamplifier designed for use with a low level cartridge by following the basic amplifier with an additional stage of amplification as shown in *Fig.* 3. With a high-level cartridge the amplifier noise and hum level will be about 16 db lower with respect to the signal level than would be the case with the same amplifier and a low-level cartridge.

If the preamplifier is to provide for switching between various turnovers,

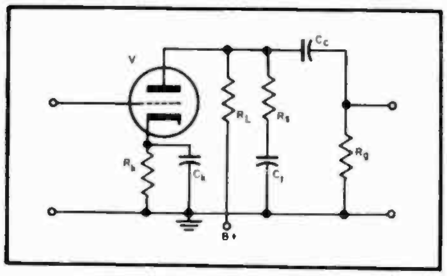


Fig. 4. Basic variable-load-impedance bass-boost amplifier circuit. Triode shown for simplicity.

the selector switch must be of the shorting type to avoid serious switching transients high-quality capacitors should be used for C_t, and a 10- to 20-meg. resistor must be placed in parallel with the selector switch for each switch position. These precautions are required because of the d.c. potential difference across C_t. The value of the switch-shunting resistors is not critical, but the combined resistance of all of them in parallel should be no less than about 40 times R_s.

The Columbia long-playing recording characteristic requires a 500-cps turnover with a maximum bass boost of about 14 db. For this compensation the capacitor used for C_t should have shunting it a resistor of such value as to provide in parallel with the resistor used for R_s an effective resistance for the R_s term in Eq. (5) that will give a value for A_{LF} which is five times the value of A_{HF} for the amplifier. To avoid serious switching transients a blocking capacitor should be placed in series with this resistor. This blocking capacitor should be chosen so that the product of its capacitance (in farads) and the lowest frequency of interest is at least equal to the reciprocal of three times the resistance of the shunting resistor in series with it.

Variable-Load-Impedance Bass Boost

A simple, commonly-used type of bass boost circuit is shown in *Fig.* 4. R_k, R_L, and R_g may be selected from resistance-coupled amplifier data to provide the maximum gain desired at the lowest frequency of interest. The requirements on C_c and C_k are the same as discussed for the basic amplifier in the degenerative-bass-boost circuit. The high-frequency gain is given by the product of the mutual conductance of the tube and a resistance equal to the parallel combination of the plate resistance of the tube, R_L, R_s, and R_g. The required value for C_t may be computed from Eq. (6) by substituting R_s for $R_1 + R_2$. The maximum gain should be at least 6 db above, or two times greater than, that required for the greatest bass boost desired, if a 6 db per octave slope is to be followed to the lowest frequency of interest. A 6SJ7 operated to obtain a maximum gain of 200—46 db—will provide for a high-frequency gain of about three and

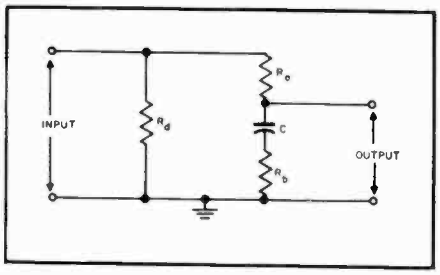


Fig. 5. Treble roll-off circuit. R_d selected for cartridge damping and/or to obtain roll-off rates between 6 and 12 db per octave. R_b selected to obtain roll-off rates less than 6 db per octave.

full bass compensation down to 30 cps for a 900-cps turnover when used in this simple circuit. There will be, of course, no degenerative feedback to provide for low distortion obtainable with the previously discussed system. For a 500-cps turnover compensation may be carried about one octave lower, or about 6 db more gain may be provided for the high frequencies.

By following a degenerative-bass-boost amplifier with a stage of amplification containing variable load impedance bass boost with a very low turnover frequency—50 cps, for example—extra boost for the low bass may be obtained if desired.

Treble Compensation

For a 6 db per octave roll-off, the simplest treble compensation is obtained by selection of the resistance shunting the magnetic reproducer cartridge.[2] Neglecting the effects of cartridge internal capacitance (usually negligible in the audio range), 6-db-per-octave attenuation will be obtained as figured from a frequency equal to 6.28 times the cartridge inductance in henries divided into the sum of the internal resistance of the cartridge and the effective external resistance shunting it. One can select the roll-off frequency by adjustment of the preamplifier input resistance with either a fixed position switching arrangement or a continuously adjustable potentiometer control.

A more flexible treble roll-off circuit

[2] Norman Pickering, "Effect of load impedance on magnetic pickup response." Audio Engineering, Mar. 1953.

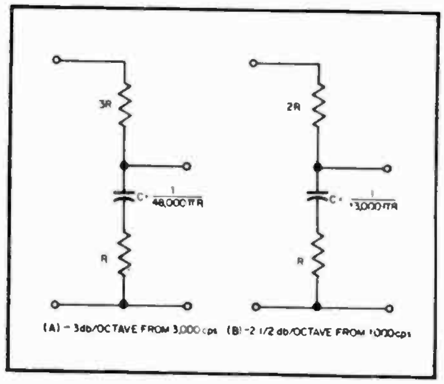


Fig. 6. Circuits for commonly used roll-off less than 6 db per octave. (A) is accurate within ± ¼ db to 15,000 cps. (B) is accurate within ± ¼ db to 10,000 cps and gives 1 db less than ideal attenuation at 15,000 cps.

is shown in *Fig.* 5. R_d is usually selected for cartridge damping in accordance with the recommendations of the cartridge manufacturer. If R_b is zero, an attenuation at 6 db per octave will be obtained as figured from a frequency equal to the reciprocal of the quantity 6.28 times the product of R_a in ohms and C in farads. R_a should be large enough so that its parallel combination with R_d acting in connection with the cartridge inductance will not cause appreciable additional high-frequency attenuation, unless a roll-off rate greater than 6 db per octave is desired. To accomplish this the resistance of R_a and R_d in parallel should be at least 125,000 times the cartridge inductance in henries. *Figure* 6 gives data for selection of components for treble roll-off at less than 6 db per octave as required by some RCA Victor and some British 78 rpm recordings.

Noise and Hum Reduction

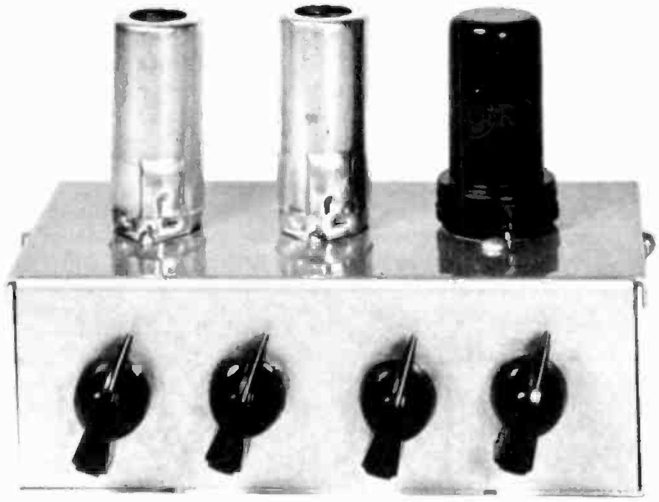
If the preamplifier is laid out and wired carefully, hum and noise may be reduced to a negligible level by simple means. Resistor noise will ordinarily be unnoticeable; the ultra-perfectionist may practically eliminate it by using wire wound resistors in the plate circuit and any unbypassed portion of the cathode circuit of the first stage. Most of the hiss noise from an amplifier usually comes from the first tube. Tube noise varies greatly between tube types, and between individual specimens of a given type. Unless a selected low-noise tube is used, tube noise will usually be less in a preamplifier employing a triode in the first stage than in one using a pentode. If the first stage is to use a pentode it is well worth while to use the 1620 or the 5879 for their low microphonism, hiss, and hum. Where a low-noise triode is desired one can use the 12AY7 or a triode-connected 5879.

The most obvious and straightforward way of eliminating hum arising in the cathode circuits is to use direct current heater power. D.c. for heaters may be obtained from a full-wave dry disc bridge rectifier with a simple capacitance filter of between 1000 and 5000 microfarads. The rectifier will require an r.m.s. input voltage about 50 per

A Three-Element Bass Control

GLEN SOUTHWORTH*

One solution to the problem of providing realism in music reproduction is to introduce a controlled amount of "hangover" by means of the circuit described which permits boosting bass at three separate frequencies independently.



The three-element bass tone-control unit designed to obtain power supply from main amplifier.

In THE DESIGN of conventional tone controls a cut and dried procedure is very often followed. A sine-wave test signal is used and the circuitry adjusted to provide the desired number of decibels boost or cut in reference to 1000 cps, typical steady state response curves being shown in *Fig.* 1. While very useful in many cases, this type of tone control is not very likely to provide adequate compensation for some program sources as well as the listener's speaker system, room acoustics, and individual musical tastes.

The bass-boost circuitry described in this article is an attempt to provide a musical form of bass emphasis similar to the kind generated naturally by good acoustics. To do this, particular attention has been paid to the transient character of most bass sounds as well as some of the significant factors of human hearing. The results obtainable should be highly valued by the listener who desires good, full, audible and "feelable" bass together with a minimum of interaction or apparent distortion of high frequencies.

Figure 2 shows three of the most commonly used methods of bass boost. These are the R-C network, the R-C-L or resonant circuit, and the inverse-feedback method of obtaining bass emphasis. A fourth type of frequency emphasis is obtained by altering the feedback path of a triode or pentode in such a manner that the circuit simulates a resonant inductance-capacitance com-

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PREAMPLIFIER DESIGN

(*from preceding page*)

cent greater than the required filament voltage. One should provide a variable series resistance between the rectifier and the capacitor so that the filament voltage may be adjusted to the proper value. With this arrangement a standard 12.5-volt transformer may be conveniently used for a 6.3-volt d.c. heater supply.

If one is not so unfortunate as to get a poor tube in the input stage, hum due to a.c. heater operation may usually be eliminated by the simple arrangement of *Fig.* 7. The value of the bypass capacitance is not critical (values less than 1.0 µf have been used successfully). Satisfactory hum reduction may often be accomplished by eliminating the balancing potentiometer and connecting the positive bias to the center tap of the filament transformer. When the balancing potentiometer is used it is well to locate it and the associated bypass capacitor near the first stage of the preamplifier as it takes the place of the usual grounding of one side of the heater line.

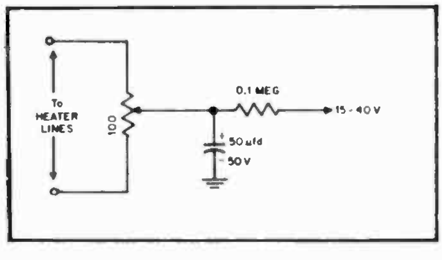

Fig. 7. Hum reduction circuit. Neither of the heater lines may be grounded, and a center tap on the transformer should not be used. Potentiometer is adjusted for minimum hum.

Illustrative Preamplifier Circuit

As an illustration of the principles discussed in this article, *Fig.* 3 gives the schematic of an easily constructed preamplifier designed for use with the author's Ultra-Linear Williamson power amplifier. This preamplifier has ample gain for use with a low-level GE reluctance cartridge and carries compensation fully down to 30 cps for the highest turnover frequency. At maximum gain settings approximately one-third volt input on the high-level inputs 3 and 4 will drive the power amplifier to full output. The author's unit is completely contained within a 10 x 4 x 2½ inch aluminum box. In constructing the preamplifier it is important to keep to a minimum the stray capacitances in the treble compensation circuits and in the input connection to the 12AX7. If the preamplifier is to be used only with a Pickering cartridge, or any other make of similarly low inductance, it would be well to double the compensation capacitor connected to the first 12AX7 grid to 1000 µµf and reduce to one-half the values given for all the compensation resistors connected to the treble switch points 2, 3, 4, and 5, thus reducing stray and input capacitance problems.

In the author's opinion, any adjustment of the system frequency response beyond the simplest that will compensate for the recording characteristics of the disc being played will deteriorate the transient response. As a concession to situations in which simulated live program loudness is undesirable or unpermissable, the circuit of *Fig.* 3 has been designed to incorporate an IRC loudness control. For this control to function properly the level control must be set to give simulated live program loudness with the loudness control at its maximum setting. Much of the criticism of loudness controls has come from failure to do this.

bination. Though more complicated than the LC circuit of (B) in *Fig.* 2, This circuit has a number of distinct advantages. It requires no expensive high-Q inductances, and is therefore not

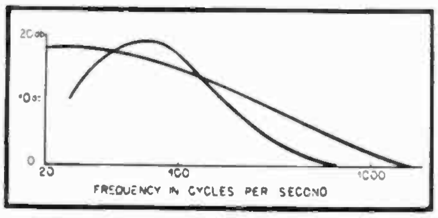

Fig. 1. Comparative steady-rate response curves of conventional bass-boost circuit (A) and resonant bass boost (B).

susceptible to hum pickup from stray magnetic fields Likewise, it is a high-impedance device with essentially zero insertion loss and as such is easily adapted to, or combined with, other audio circuitry. In addition, control of the circuit is very flexible and apparent Q's from zero to near infinity may be obtained by varying the setting of a single potentiometer.

The steady-state response curves of a simulated resonant circuit are shown in *Fig.* 1 in comparison with those of a conventional tone control. Two factors are worth noting: First, that the curve obtained is such that added accentuation is obtained in the low-bass region where it is frequently needed to compensate for poor speaker and baffle efficiency, inferior acoustics, and possible defects in original program material. Secondly, the response can be made to drop off quite sharply below a chosen frequency in order to reduce the possibility of amplifier or speaker overload by very-low-frequency components, or in some cases to attenuate serious hum in the program material.

Steady-state measurements tell only part of the story, and in many cases the transient response characteristics will be more important in determining what the reproduced bass sounds like. *Figure* 3 shows a series of scope photos illustrating the effects of various types of bass circuits on transient pulses. In the case of the synthetic resonant circuit, nearly pure bass fundamentals are generated from the transient pulses not only at the main resonance frequency, but for an appreciable range on either side.

The synthetic construction of fundamental tones from transient pulses may be considered to give the following benefits. It tends to duplicate effects produced naturally by good acoustics, it tends to provide a signal more easily reproduced by present day speakers, and—rather paradoxically—it can result in better apparent highs. This last characteristic is a psycho-acoustic effect resulting from the fact that the masking characteristics of the low-frequency reproduction are radically changed. As a result, more bass boost can be used without drowning out the highs with strongly masking pulses or noise components. At the same time it may be found that certain transients become semi-audible that were not reproduced at all previously and these give a softening and more musical effect to what might otherwise be strident highs.

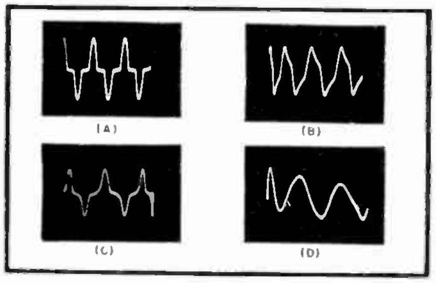

Fig. 3. Oscilloscope photos showing pulse series with repetition rate of 100 cps. (A) shows initial pulse shape; (B) shows output of R-C bass-boost network such as that of (A) in Fig. 2; (C) illustrates output from feedback type of boost circuit; and (D) is 'scope trace of output of the three-element bass-boost circuit.

In addition to its pulse-forming characteristics, the resonant circuit has another aspect that has suffered considerable misunderstanding. This is the fact that a high-Q circuit will generate "hangover" when shock excited. Unfortunately, hangover has come to be associated with a host of undesirable distortions in bass reproduction, and as a result, resonant circuits have been little used in recent years. Actually, a thumpy or boomy reproducing system is apt to be suffering from too little hangover rather than too much. The reason for this is that rapidly damped wave trains produce strong "side-bands," or adjacent frequencies, and the ear tends to hear and be irritated by the spurious high-frequency components of a too-rapidly damped bass note. Similarly, adjacent resonances in the speaker, baffle, or acoustics tend to be strongly stimulated and consequently may produce disagreeable beats.

The introduction of artificial hangover before the loudspeaker means several things. Short duration wave trains are lengthened and non-symmetrical components are largely eliminated, with the result that an objectionably pitched speaker or cabinet resonance is much less apt to be stimulated. Similarly, fewer spurious sidebands are produced, with the result that the listener hears deeper and clearer bass. This last may be considered of definite importance when listening at low levels due to nonlinearity of the ear which gives the effect of heavily damped wave trains. However, certain precautions should be taken to secure optimum results with resonant bass boost circuits. In the average phonograph, there are apt to be three major sources of mechanical resonance in the bass region. These are the loudspeaker, the speaker cabinet, and the phonograph pickup arm, and sometimes the proper combination of these elements can result in a system with warm, vibrant, bass without electrical boost. However, if the electronic resonance is peaked at or near the frequency of one of the mechanical resonances very disproportionate response may occur.

Practical Circuit

Figure 5 shows the schematic of a bass-boost circuit which employs three different resonant elements. By using two or more resonant elements in the bass region the over-all response may be more closely compensated for and tendencies toward "one-note" reproduction are greatly reduced. In the circuit shown, each control serves a dual purpose, acting as a means of controlling the "Q" of the resonant element as well as an attenuator of that particular channel. Thus, when the three bass controls are turned all of the way down, no bass boost occurs. When the controls are turned part way up, a low "Q" resonance is simulated and pulse forming occurs. When the controls are turned all the way up, a high "Q" resonance is simulated and artificial hangover occurs. The fourth control shown is the input level to what is called a "side amplifier," a channel which introduces no frequency discrimination, and which

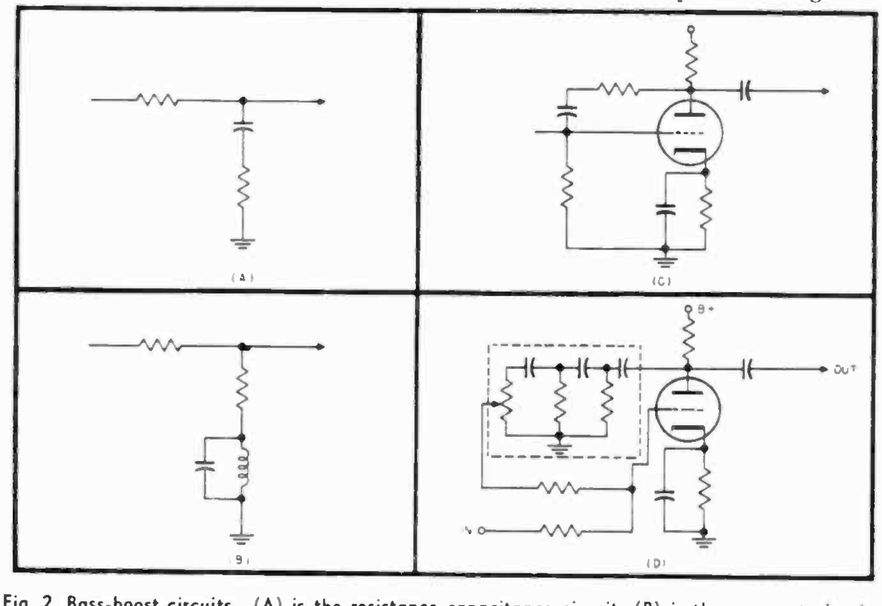

Fig. 2. Bass-boost circuits—(A) is the resistance-capacitance circuit; (B) is the resonant circuit, usually low impedance; (C) is the feedback type of bass equalization; and (D) is the simulated resonant circuit, with frequency determining elements shown in dotted box.

is used to set the level of the high-frequency response in comparison to the bass. In some respects, this part of the circuit is worthy of special attention in itself, inasmuch as the slope of the high-frequency response curve is not changed, but merely the relative amplitude of all of the highs. This means that it is possible to achieve emphasis of clarity giving middle highs without adding excessive high-frequency noise to the reproduction.

While the unit shown in *Fig.* 5 obtains power from the main amplifier, the system shown in *Fig.* 4 is completely self contained, and is designed for easy insertion between a crystal pickup or a preamplifier and the main power amplifier. The self-contained power supply, shown in *Fig.* 6, furnishes a B+ voltage of about 150, which is adequate for most purposes. This supply can be incorporated into a single chassis, as in *Fig.* 4. and will mount readily in a 3½×6×2 in. chassis. In a

Fig. 4. Three-element bass-boost circuit with self-contained power supply.

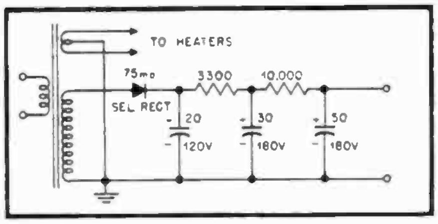

Fig. 6. Power supply incorporated in unit shown

compact, self contained, unit, care should be taken that the two dual triodes are located in the minimum field from the power transformer. This position can usually be located by means of a small pickup coil—such as a magnetic phono cartridge—and a high-gain amplifier, the best procedure being to determine the point of minimum field intensity from the power transformer before any of the components are mounted. This may be done by connecting the primary leads of the transformer and moving the exploring coil about it until the region of minimum field is established.

Performance

The frequency of each resonant circuitis determined by the values of the three resistors and capacitors in the feedback loop. Increasing the value of any of these components will lower the resonant frequency, while decreasing the values will raise the point of resonance. Suggested values for specific frequencies are shown on the diagram, but superior performance with a given audio installation may likely be obtained with a different group of frequencies selected to match the resonances of the system.

Bass tone is likewise strongly influenced by the amount of low-frequency distortion present in the signal source or reproducing equipment. In the construction of the three-way bass circuit, the use of different components tend to alter the character of the resonant decay curve due to changed linearity characteristics. For example, the substitution of pentodes for the triodes shown in the schematic may produce significant, though not necessarily undesirable, changes in tone color. Likewise, when using larger values of capacitance in the feedback loops, such as .01 µf or greater, the effect of dielectric hysteresis may alter the resultant sound, but in some cases this may be beneficial.

Aside from the benefits of deeper and more musical bass, the use of the three-element bass control seems almost to add a new dimension to radio broadcasts and the various elements of a station's technical personality seem to stand out more clearly due to the superior reproduction of low-frequency transients. For example a control room microphone that is poorly shock mounted may exhibit an interesting serious of thumps and bumps while the operator is reading news or changing records in the middle of an announcement. Similarly, matters like hum in different audio channels, microphonics, turntable or record rumble, air conditioner noises, and so on, tend to show up more clearly. Of course, if these sounds become too distracting they can easily be eliminated by proper setting of the bass controls, a procedure somewhat easier than getting out the hammer and saw to modify the acoustic resonances of the room or cabinet.

In conclusion, the use of multiple resonance bass circuits should appeal to many listeners who desire fullness in the lower registers. To date, the only detrimental effects that have been noted when using circuits of this nature are found when the electronic resonance is set too closely to one of the mechanical resonances of the system, or when excessive distortion elsewhere has tended to counteract the benefits derived. Particular attention might be paid to the elimination of cabinet or room rattles which may be generated by the fundamental bass tones. Similarly, amplifiers such as the "Ultra-Linear" are recommended.

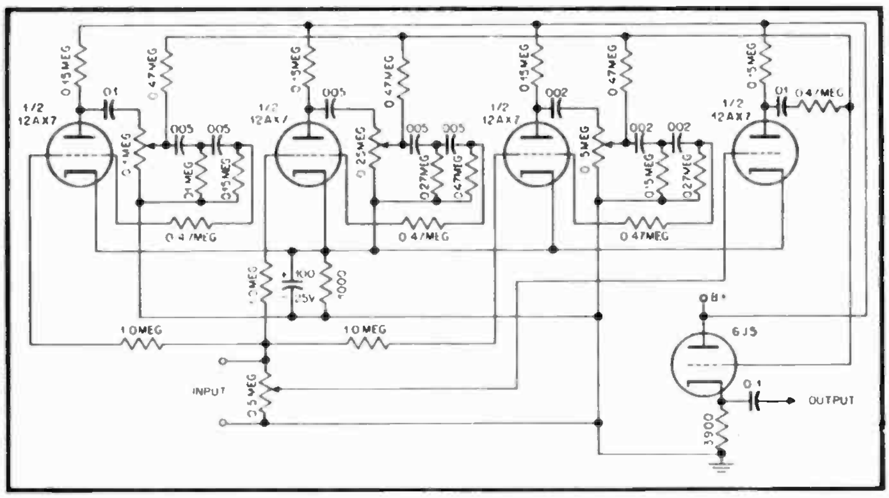

Fig. 5. Schematic of tone control system using three different simulated resonant elements, a side amplier, and a cathode-follower output. Note that no over-all volume control is included. Resonant frequencies are 45, 80, and 120 cps.

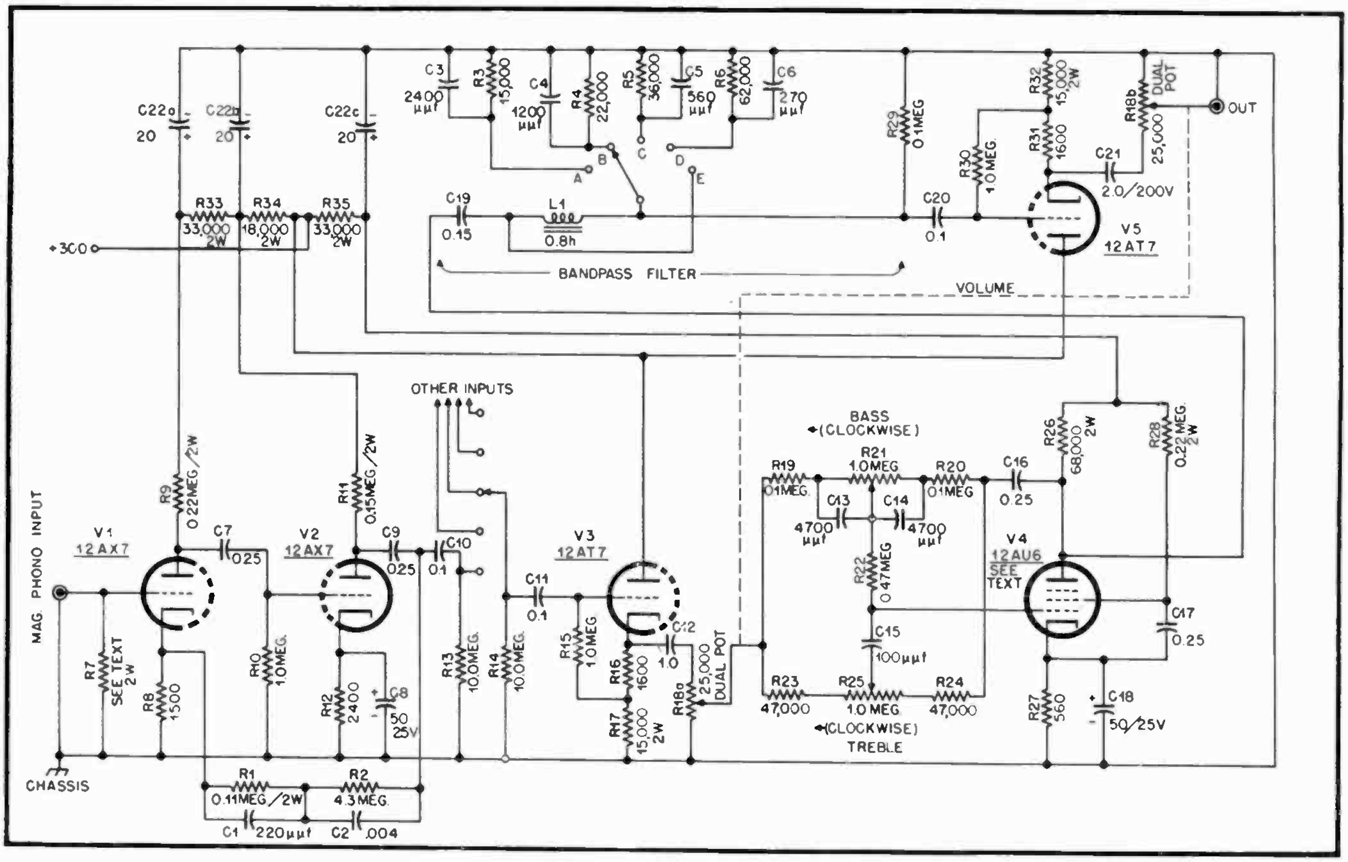

Fig. 2. Complete schematic diagram except for d.c. heater supply.

Versatile Control Unit For The Williamson

CHARLES R. MILLER*

A combination preamplifier, tone controls, and bandpass filter giving minimum noise and distortion and maximum convenience.

IN THE FIELD OF HOME MUSIC reproduction, the Williamson amplifier and its modifications have come to enjoy preeminence due to outstanding performance, but there has been no similar standardization in tone-control and preamplifier units intended to complement the Williamson. The unit described in this article is an attempt to meet that need.

Functionally, the unit is divided into five sections as shown in *Fig.* 1. First, it was decided that to preserve a high signal-to-noise ratio, all tubes were to be operated with direct current on the heaters, and secondly that all the sections must be capable of low-distortion performance matching the Williamson. The first specification can be met most easily by using all the heaters in series as the cathode-bias resistor of the power tubes, using a method to be described later. It was felt, further, that the second could be met only by using negative feedback on each stage. While it may be argued that the signal in low-level stages is so small that negative feedback is not needed, it has been shown that this is not the case.[1]

The complete circuit appears in *Fig.* 2. The design of the preamplifier V_1-V_2 is fairly conventional, with the equalizer values R_1, R_2, C_1, and C_2 being obtained by appropriate transformation of the networks given by Boegli[2,3] for record

[1] W. B. Bernard, "Distortion in voltage amplifiers," AUDIO ENGINEERING, Feb. 1953.

[2] C. P. Boegli, "A preamplifier for magnetic and crystal pickups," *Radio and Television News*, July 1950.

[3] C. P. Boegli, "An improved equalizer-preamp," *Radio and Television News*, April 1951.

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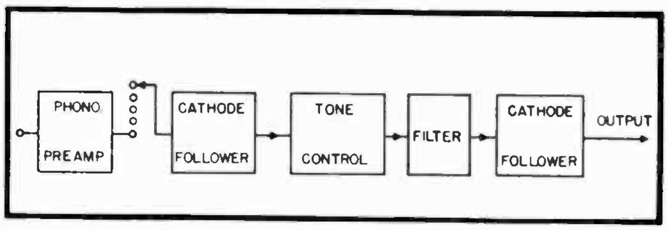
Fig. 1. Block diagram of the control unit.



compensation. The gain-frequency characteristics of the preamplifier for various equalizer values is as shown in *Fig.* 3. If simplicity is desired, the network shown as the compromise network can be used and any additional correction needed obtained through the tone controls. It should be noted that in no event are half-watt resistors used, these generating a much higher noise level than would be predicted on the basis of thermal noise. Furthermore, cathode bias is used in the second stage to reduce distortion[3]

There may be some question as to the use of the 12AX7 as the preamplifier tube for low-noise service instead of the 12AY7. Fairly extensive tests were made on the noise and hum level to be expected from various tubes. In particular, about two dozen 12AX7's and 12AY7's were checked for noise equivalent input at the grid. The equivalent hum and noise input were compared, both for a.c. and d.c. heater operation, with the only point of superiority of the 12AY7 being in the matter of hum with a.c. heater operation. With d.c. heater operation and the usual gain *vs.* frequency characteristics for phonograph equalization, the most important source of noise is flicker noise, presumably caused by cathode poisoning.[4]

Flicker noise varies inversely as the square of the frequency and hence the output due to flicker noise varies inversely as the cube of the frequency. For the tests mentioned, there was more variation within a given tube type than from one type to another. In any event, the average 12AX7 was better than the average 12AY7 for flicker noise. Representative 12AX7's in the preamplifier gave an unweighted signal-to-noise ratio of 68 db for a nominal 1 volt output (10 mv input at 1 kc). As predicted, the output noise spectrum had the sharply rising low-frequency characteristic typical of flicker noise. Hum level was so far below this noise as to be unmeasurable. In the complete schematic (*Fig.* 2) C_{10}, R_{11}, and R_{11} are for the purpose of eliminating switching transients due to changes in d.c. level.

The input cathode follower V_3 and the output cathode follower V_4 are perfectly conventional, serving the purpose of isolation and impedance transformation. A dual-potentiometer gain control gives best compromise between minimization of distortion and noise. If only an input gain control were used, noise generated in the following stages would reduce signal-to-noise ratio and if only an output gain control were used, distortion might be caused in the preceding stages due to overloading. It may be asked why electrolytic capacitors were not used in the cathode followers. First, it is the writer's experience that such capacitors are a source of noise and distortion and should be avoided whenever possible.[5] Secondly, a definite low-frequency cutoff was wanted to insure stability. Wow in a turntable can easily overload a poorly designed system in which the amplifier is very nearly at the motorboating point. It is idle to talk about flat response down to subaudible frequency unless the listener is prepared to "listen" with air-tight seals between his ear drums and the loudspeaker cone.

Filter and Equalizer

The tone control stage V_4 perhaps needs explanation. This is simply an anode follower[6] with feedback voltage determined by the impedances between input, control grid, and plate. If the internal gain of the stage is infinite and the input impedance at the grid infinite, the over-all gain can be shown to be in the ratio Z_2/Z_1, where Z_2 is the impedance from plate to grid and Z_1 the impedance from input to grid. More simply, the actual input signal is determined by the setting of the two controls, R_{11} and R_{18}. Bass signals go through the upper section and treble through the lower. With either control arm nearer the input, the corresponding output signal is boosted, and with the arm nearer the output, the output is attenuated. The resulting frequency response is as shown in *Fig.* 4. One advantage of this system is that at boost positions, there is still enough negative feedback to insure very low distortion. One disadvantage if the circuit were to be used with a.c. heater operation is that the actual grid signal is very low and consequently hum trouble might be expected. Note one important precaution—the potentiometers used are linear-taper units and are not the usual logarithmic controls used in conventional tone circuits.

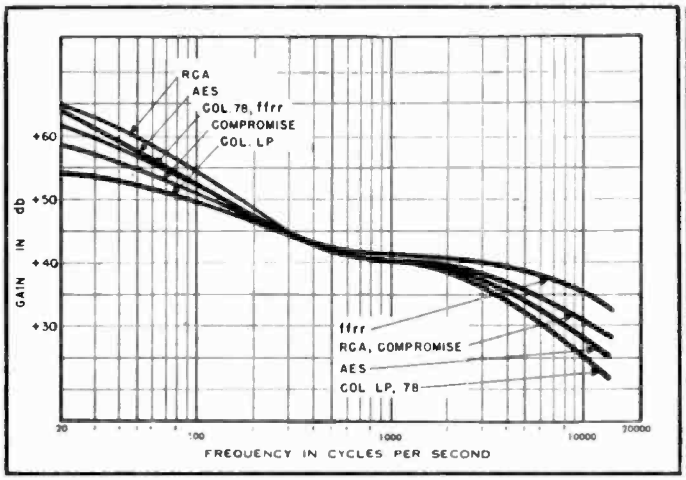
The bandpass filter is of conventional design with a 12-db-per-octave slope at the high-frequency end. To preserve balanced response C_{11} was selected to give low-frequency cutoff matching that at the high-frequency end, as it was felt that an additional 12-db-per-octave high-pass filter at the low end was not justified. The Chicago Transformer NSI-1 unit was specified since tests showed this choke to have the highest self-resonant frequency of any of the commercially available units. Note that if the self-capacitance of the choke is too high, the filter is converted from constant-*k* to *m*-derived operation, with consequently less satisfactory transient response. The values of resistance and capacitance were chosen for critical damping and not for maximally flat response. The resulting behavior is shown in *Fig.* 5.

The power supply for the control unit is somewhat unconventional. The heater current is taken from the main amplifier by using the cathode currents of the output tubes, about 15 ma drain from B+ as shown in *Fig.* 6. Previous tests on the Williamson have shown an almost constant current taken by the output tubes for any power level up to the overload point. To prevent a burnout of one of the heaters from pulling the cathode voltage up to B+, a neon lamp is inserted as shown. In normal operation the lamp is below 50 volts and thus will not fire or affect operation. Burnout of a heater is signified by the neon lamp glowing, at

[4] Valley and Wallman, "*Vacuum Tube Amplifiers*," Radiation Lab Series, Vol. 18, McGraw-Hill, New York, 1948.

[5] J. E. Lilienfeld and C. R. Miller, "Distribution of conductivity within dielectric films on aluminum," *Jour. Electrochem. Soc.*, May 1953.

[6] P. J. Baxandall, "Negative feedback tone control," *Wireless World*, October 1952.



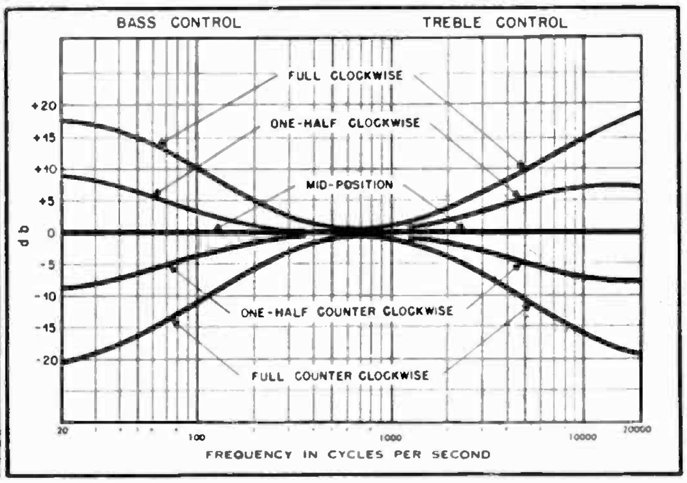
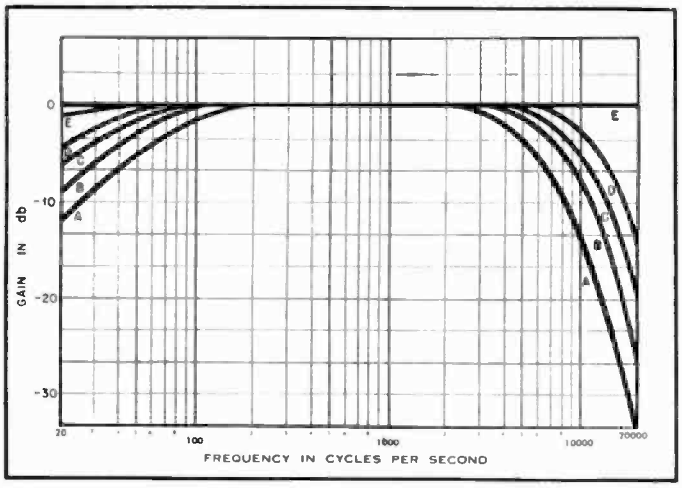


Fig. 3. (left). These curves show the equalization afforded by the various combinations of C_1-C_2-R_1-R_2. Fig. 4 (right). The range of tone control available from the circuit.



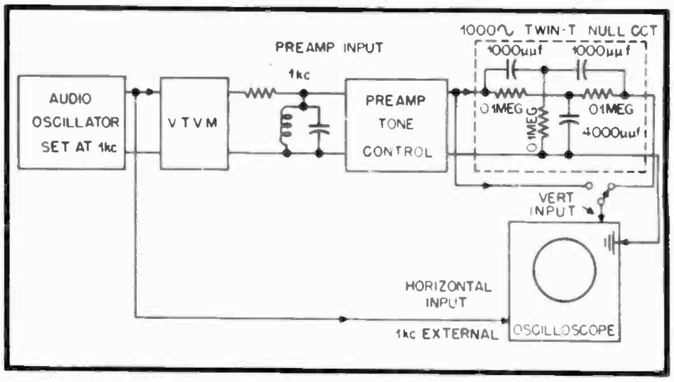


Fig. 5. (left). The five positions of the bandpass filter switch give these results.

Fig. 7. (above). Test setup for distortion measurements.

which time the system should of course be turned off and the trouble fixed. This scheme has the further advantage of preventing B+ from rising to a value destructive to the electrolytics if one of the heaters should burn out. The second feature of the power supply is that B+ for the control unit is taken from voltage-regulator tubes. This simple precaution will save a good deal of grief in trying to design for decoupling adequate to prevent motorboating. It is also usually more economical than the electrolytic capacitors needed to give the same low-frequency stability.

Final Results

To test the total distortion generated by the unit, an intermodulation analyzer would be best. Without access to such a unit, the following test procedure was adopted. Equipment layout was as shown in *Fig. 7*. With the tone controls set at flat position and the gain control at maximum, the signal from the GR Microvolter was raised until the output was at the desired level. At the design output level of 1 volt, the distortion was unmeasurable for any setting of the tone controls or the filter. For 10 volts output and maximum bandpass, the distortion was about 0.1 per cent. With 10 volts output and minimum bandpass, the distortion increased to 0.3 per cent. However, since 10 volts represents a value 20 db above overload for the main amplifier, it is felt that for normal operation, both harmonic and intermodulation distortion will be negligible. Since most of the distortion is caused by the filter loading the tone control output, this could be made negligible by insertion of a third cathode follower to drive the filter. For the 1-volt operating level, this was not deemed necessary.

The mechanical layout is quite normal. The entire control unit is built within an 8 × 12 × 3-inch chassis (see *Fig. 8*), with the tubes being arranged on a terminal board, which itself is shock mounted by means of grommets. To keep electrostatic coupling from getting hum into the signal circuits, the power switch and pilot light are external to the main chassis as shown in the photograph. There is thus no 60-cps source within the chassis. Magnetic coupling is prevented by the usual wiring practice of a ground bus connected to the chassis only at the input. The shorting switches used for the signal circuits are the new Centralab PA2000 series, which both are smaller and give better performance than the usual wafer type. The additional tube socket is for a proposed microphone preamplifier. When and if this is used, the 12AU6 tone-control tube will be replaced by a 6BH6, with a second 6BH6 for the microphone preamplifier. In any event, the total heater voltage will be 36 volts for the series string. The electrolytic capacitor used for filtering B+ within the control unit is mounted on a Mallory PS-6 socket for easy replacement. Ventilation holes are drilled in the bottom plate.

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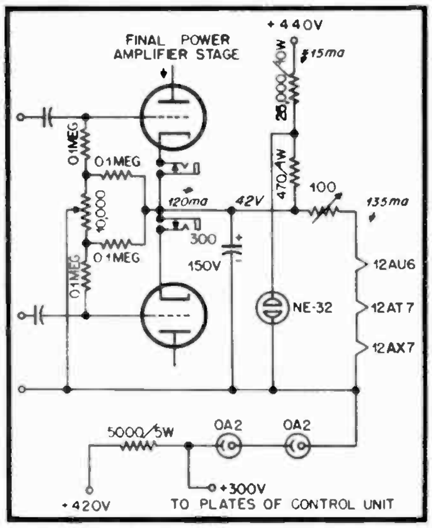


Fig. 6. D.c. for the heaters is taken in this manner from the cathode circuit of the final stage of the Williamson.

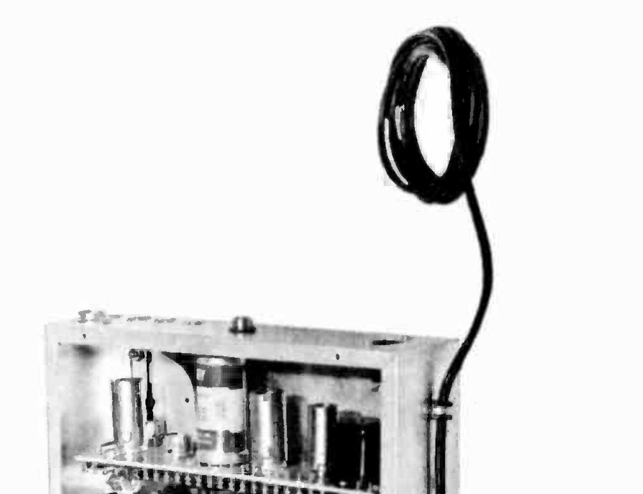


Fig. 8. Photo of the underside shows terminal-board construction.

Preamp with "Presence"

C. G. McPROUD

Introducing a circuit which departs from "flat" to provide the elusive quality called presence, and which incorporates most of the features desired by the critical listener.

WHILE THE GENERAL TREND in amplifier and equipment construction has usually been in the direction of completely "flat" response curves, it is well established that certain types of correction are useful to compensate for microphone placement, for deficiencies in equipment, and for personal preference. However, the use of a separate control for increasing "presence" has not been introduced in home equipment to date, although it has been used in professional equipment for many years—although not necessarily by that name.

In the early 30's, certain types of loudspeaker systems in theatres were in need of some form of correction in the mid-range to increase the illusion of realism. It is probable that the deficiency was due to the speakers themselves, but in any case a correction in the form of a small boost somewhere between 2000 and 3000 cps was introduced to improve presence. The boost was of the order of 4 db, and was usually located at 2700 cps.

Following some experimentation along these lines, it was found that the addition of a boost ranging from 4 to 6 db at 2700 cps gave the illusion of moving a solo violinist out in front of the loudspeaker, or of causing the singer with a popular band to step out in front. Admittedly the effect is slight, but it is sufficiently noticeable to warrant its inclusion in the amplifier to be described.

The Complete Circuit

Every so often, a confirmed experimenter feels the urge to make a change—usually having in mind the desire to add one or more operating features which appear to be desirable. Reviewing the real or fancied faults in existing circuitry, the writer planned a preamplifier and tone-control amplifier which would incorporate all of the desired features. Taking the steps one by one, they add up about as follows:

Some preamplifiers do not operate as quietly as would be desired—having a hiss or hum level too high for complete satisfaction. Furthermore, it was felt that the low-frequency boost required for magnetic pickups began to flatten off too soon, resulting in insufficient bass in the lowest range—that is, below 50 cps. Some amplifiers have insufficient compensation characteristics for the most critical listener, and while the trend for the newcomers to hi-fi seems to be to reduce the number of available curves—largely because those who are entering the hi-fi field "fresh" will generally build their libraries exclusively from the LP and 45 catalogs—there is some advantage to having complete flexibility.

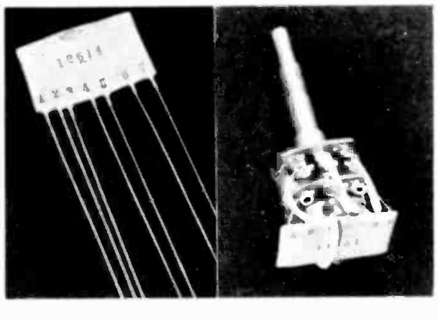


Fig. 1. (Left), Centralab Couplate for the tone-control circuit. (Right), the Couplate assembled on a dual-concentric control.

The tone-control circuit first described by Baxendall[1] and more recently for use with American tubes by Barber[2] offers some interesting possibilities. Principal among these is the variation in the inflection point as the amount of boost or cut is changed. The advantages of this show up when using a wide-range loudspeaker system with usual source material—both records and radio programs.

In addition to the presence control, it also seemed desirable to incorporate a low-pass filter system into the amplifier, and the L-C network described by Markow[3] was investigated for the purpose.

[1] *Wireless World*, October 1952.

[2] Basil T. Barber, "Flexible tone control circuit." See page 34.

[3] Elliott W. Markow, "Record improvement with h.f. cutoff filters." See page 63.

VERSATILE CONTROL UNIT

(from preceding page)

Modifications

The schematic of the amplifier—*Fig. 1*—shows fixed values for the frequency-determining elements, C_1 C_2, and R_2, and for the values shown, the turnover would be at approximately 375 cps, and the high-frequency response would be down approximately 6 db at 10,000 cps, which closely fits the ffrr curve. These values can be changed to suit the individual preference, or the circuit can be arranged so that various values could be switched in at will. C_2 controls the turnover frequency and R_2 controls the amount of flattening or deemphasis at the low end. The amount of rolloff is controlled by the capacitor C_1.

For equalization to fit the RIAA curve—which is becoming more and more the standard in the industry—it is advisable to decrease the value of C_2 to .0035 µf, change R_2 to 1.0 megs, and increase C_1 to 470 µµf. This should give suitable results with almost any LP record, provided some adjusting of the tone controls could be resorted to. For other equalization curves, the values shown in Table I could be substituted for those in *Fig. 1*.

TABLE I

Component Values for Different Equalization Curves

Curve	C_1	C_2	R_2
RIAA	470 µµf	.0035 µf	1.0 meg
ffrr	220	.004	4.3
Col 78	1000	.004	4.3
RCA 78	470	.003	1.8
Old AES	620	.004	1.8
Col LP	1000	.005	0.56
Compromise	470	.005	1.2

The value for R_7 is that specified by the pickup manufacturer as optimum. A 2-watt unit is recommended to ensure low-noise operation.

PARTS LIST

C_1, C_2	See Table I
C_3	2400 µµf, mica, 5%
C_4	1200 µµf, mica, 5%
C_5	560 µµf, mica, 5%
C_6	270 µµf, mica, 5%
C_7, C_9, C_{16}, C_{17},	0.25 µf, 600 v, paper
C_8, C_{18}	50 µf, 25 v, electrolytic
C_{10}, C_{11}, C_{20}	0.1 µf, 600 v, paper
C_{12}	1.0 µf, 600 v, paper
C_{13}, C_{14}	4700 µµf, 400 v, paper
C_{15}	100 µµf, mica
C_{19}	0.15 µf, 600 v, paper
C_{21}	2 µf, 200 v, paper
L_1	0.8 Hy
R_1	0.11 meg, 2 watts
R_2	See Table I
R_3	15,000 ohms, 1 watt, 5%
R_4	22,000 ohms, 1 watt, 5%
R_5	36,000 ohms, 1 watt, 5%
R_6	62,000 ohms, 1 watt, 5%
R_7	See text
R_8	1500 ohms, deposited carbon, 1%
R_9	0.22 meg, 2 watts
R_{10}	1.0 meg, 1 watt
R_{11}	0.15 meg, 1 watt
R_{12}	2400 ohms, 1 watt
R_{13}, R_{14}	10.0 megs, 1 watt
R_{15}, R_{30}	1.0 meg, 1 watt
R_{16}, R_{31}	1600 ohms, 1 watt
R_{17}, R_{32}	15,000 ohms, 2 watts
$R_{18a, b}$	Dual 25,000-ohm potentiometer, linear taper
R_{19}, R_{20}, R_{29}	0.1 meg, 1 watt
R_{21}, R_{25}	1.0-meg potentiometer, linear taper
R_{22}	0.47 meg, 1 watt
R_{23}, R_{24}	47,000 ohms, 1 watt
R_{26}	68,000 ohms, 2 watts
R_{27}	560 ohms, 1 watt
R_{28}	0.22 meg, 2 watts
R_{33}, R_{34}	33,000 ohms, 2 watts
R_{35}	18,000 ohms, 2 watts
V_1, V_2	Two halves of 12AX7
V_3, V_5	Two halves of 12AT7
V_4	12AU6

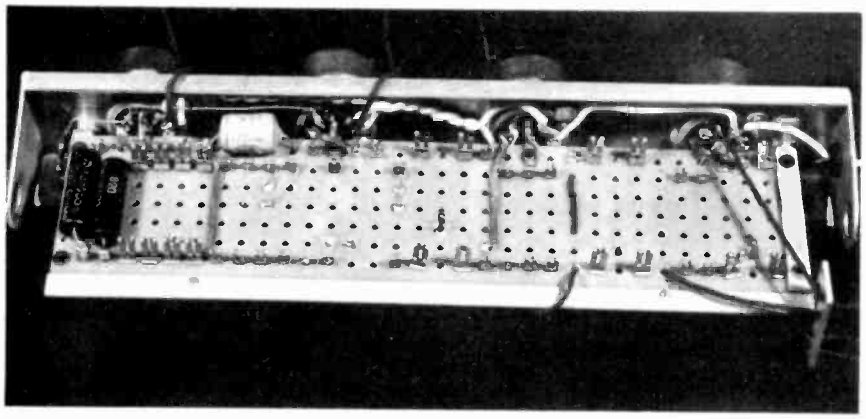


Fig. 2. Resistor mounting card assembled in small chassis with four sockets attached. Terminal lugs may be staked into any desired holes in the board, and thus fit any circuit. Grid and input circuit components are on the top side; plate and decoupling components are located on the bottom.

And as this writer has long been a proponent of the loudness control, this too had to be included—but with some form of decompensation which could easily be an uncompensated volume control.

Adding these up, one arrives at the astonishing figure of eight as the required number of controls, nearly as bad as the early TV sets before the trend toward simplification resulted in sets with only two knobs. Follownig the same procedure, the amplifier was planned to use only four "apparent" knobs—each being a dual concentric control. Thus the phonograph preamplifier becomes a dual switch each with five positions; the tone control becomes a dual-concentric assembly of two potentiometers; the presence control and the low-pass filter switch are combined into a third unit; and the fourth consists of another dual-concentric control—one section being the loudness control and the other being a volume control.

Obviously, few of these units are readily available from jobber stocks, so they were described to Centralab, and the four complete units were soon forthcoming. Centralab engineers had already designed the Senior Compentrol, which combines the loudness control with a volume control, and which employs a printed circuit "Couplate" with all the compensating elements. A study of the Baxendall circuit indicated that another Couplate could be built with three capacitors and four resistors in a single unit, as shown at the left in *Fig.* 1, and assembled onto the dual potentiometer at the right. Thus the construction of the entire tone-control circuit is considerably simplified. A standard line of Centralab units—available primarily to manufacturers—included dual concentric switches and dual units with a switch and a potentiometer. These were kept in mind during the layout of the amplifier circuit, and when the requirements were settled, the problem of making the controls was left to Centralab. All four units are available for anyone who wishes to duplicate the amplifier.

One of the features of the construction is the use of "custom-built" mounting boards for many of the components. While resistor-board construction is not uncommon in factory-built equipment, it is less often encountered in hand-crafted units because of the difficulty of obtaining these strips with terminals located ideally for the particular circuit being built. However, with the Alden terminal cards, any desired number of terminals may be installed exactly where the user wishes. *Figure* 2 shows the main terminal board with the four sockets attached to it and with most of the terminals in place. Detailed information about the terminal boards and the method of using them is included with the constructional data.

Preamplifier Details

The amount of low-frequency boost available from the conventional equalized feedback circuit depends upon the gain available from the two tubes. Consider, for example, the use of two sections of a 12AX7 with gain of around 40 for each section, or a total gain of 1600. This corresponds to approximately 64 db. For an output of 1 volt and an input of 10 mv at 1000 cps, a gain of 100 is required, which is 40 db. This gain must be available at 1000 cps. However, to provide the low-frequency boost required for proper equalization of magnetic pickups, it is necessary that an additional gain of 20 db be available at 50 cps (for a 500-cps turnover). Since the maximum gain available from the tube is around 64 db and 40 is required at 1000 cps, the remaining 24 is all that is available for boost of the low frequencies. Now the frequency-response curve does not consist of two straight lines, but is composed of the flat portion above turnover, plus a portion which increases with lowering frequency at the rate of 6 db per octave, plus a portion which is again flat, but which represents the total gain of the two sections of the tube without feedback. These three straight lines are joined with curving sections, as shown in *Fig.* 3.

From this, it is seen that there is hardly sufficient low-frequency boost available from a 12AX7 to provide full equalization for the AES curve nor for the 800-cps turnover, if the compensation is to extend fully to the lowest frequencies to be passed. This discussion does not apply to the New Orthophonic curve, since it rolls off with a maximum required boost of 18.6 db at 30 cps. The LP and NARTB curves are similar in the low-frequency range.

When tubes with less gain than the 12AX7 are employed, the difference is even more noticeable, which may account for a deficiency in low-frequency boost when a 12AY7 is used in a circuit designed for the 12AX7.

To correct fully for the necessary low-frequency boost, it is apparent that more gain is required from each of the tubes (or tube sections). The logical answer is to use two pentodes. Some pentodes are too noisy for this service, considering that the input signal is in the range of 30 mv. The circuit was first built up with 5879's, but these tubes—while of excellent low-noise characteristics—have hardly sufficient gain to give the desired low-frequency boost. The Genalex Z729 recently introduced in this country, is claimed to have even lower noise and hum than the 5879, but it has an available gain about equal to that of a 6AU6. The latter tube is likely to be somewhat noisy in low-level applications, particularly if a.c. is used on the heater. With the Z729, however, suitable performance with respect both to gain and noise has been achieved.

The circuit is relatively conventional. A feedback circuit from the plate of the second tube returns to the cathode of the first, and frequency corrective circuits are introduced in the feedback loop. Series capacitors control the turnover frequency, with the New Orthophonic rolloff being obtained by connecting a shunting resistor across one of the capacitors. The turnover frequencies available are 800, 500LP (for New Orthophonic), 500, 400 (AES), and 300.

The high-frequency rolloff is provided by connecting suitable capacitors and resistors across the feedback resistor to give rolloff of 4, 8, 12, and 16 db in addition to a flat position. All of the switching is accomplished by a Centralab Series 30 switch—the panel section consisting of a single deck with two 5-position switches and the rear section consisting of another single deck with one 5-position switch. All of the resistors and capacitors are mounted on the resistor board, with flexible leads connecting to the switch. The schematic of the preamplifier is shown in the left portion of the over-all schematic, *Fig.* 4. The response curves of the preamplifier section are shown in *Fig.* 5. It will be noted from the schematic that no provision has

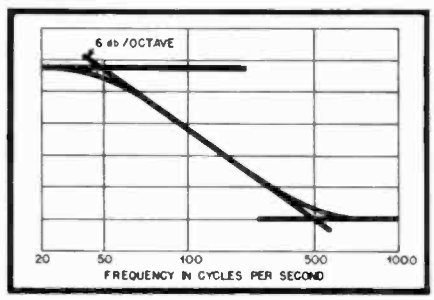


Fig. 3. Construction of the curve obtained from feedback network around two tubes. Curve is asymptotic to two straight lines.

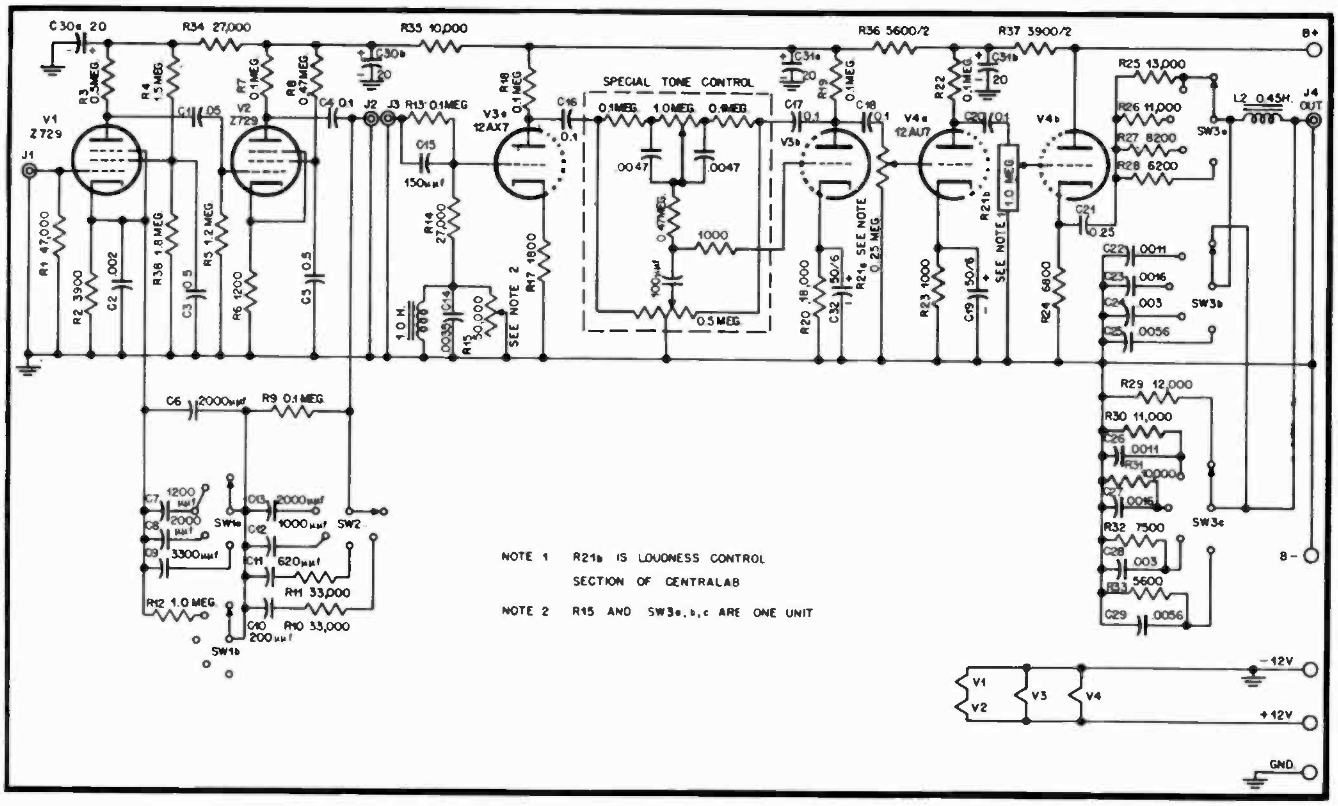


Fig. 4. Overall schematic of the preamplifier and control sections.

been made for switching inputs. The preamplifier has input and output jacks—the input coming directly from the pickup and the output being fed to a selector switch on the tuner. The output of the tuner switch if fed back to the input of the tone-control section, and its output is fed in turn to the power amplifier. There is no reason why the switching should not be incorporated in this chassis if it is desired, but for the writer's application the tuner switch was most convenient. Furthermore, this method provides complete flexibility and permits easy interconnection of other preamplifiers or tone-control amplifiers for comparative tests—a condition which is often required in this particular installation.

Tone-Control Section

The remainder of the circuit comes under this heading, although there are several other functions incorporated in this section of the amplifier. The first step consists of the "presence" circuit, which includes R_{13}, R_{14}, R_{15}, C_{14}, C_{15}, and L_1. This is a simple loss network with a tuned circuit in the shunt leg. The tuned circuit is shunted by a 50,000-ohm linear pot which varies the amount of boost from 0 to 6 db. The boost is approximately linear with respect to the amount of rotation of the pot. The L-C circuit is tuned to 2700 cps, and listening tests have shown that the L-C ratio prescribed gives about the best effect. A smaller inductance and larger capacitor could be tuned to the same frequency, but the "bump" in the response curve is narrow, and the subjective effect is not as desirable.

The Baxendall circuit provides no gain from the first tube section since it is used as a cathode follower. Furthermore, there is rather more high-frequency equalization in the tone-control circuit than is considered desirable. Since additional gain was necessary, the first section is used as an amplifier with improved performance throughout. In the final form the control section requires an input of .08 volts for an output of 1.0 volts which gives some leeway for operation of the volume and loudness controls.

With the Couplate and the dual pot, it is only necessary to connect four wires between the tube chassis and the control to have a complete low- and high-frequency tone-control circuit. The response curves obtained are plotted in *Fig.* 6 for the center or flat position and for two degrees of both boost and cut. The second section of V_3 feeds the volume control and the first section of V_4 is located between the volume control and the loudness control, giving complete isolation between the two controls and eliminating any interaction. The final tube section, V_{4b}, is a cathode follower which feeds the low-pass filter circuit.

The filter circuit is an adaptation of the network described by Markow,[3] and is connected at the output of the cathode follower. This circuit requires the use of another inductance, but the performance meets the specification laid down before the construction was begun. The control is a Centralab 30a unit, and consists of four 5-position switches mounted on two decks and operated by the outer shaft, together with the presence control—a 50,000-ohm linear pot operated by the inner shaft. Three of the switch sections are used in the filter circuit, while the terminals of the fourth serve as tie points for the matching resistors, R_{25}, R_{26}, R_{27}, and R_{28}. The eight capacitors and the four terminating resistors are mounted on another Alden terminal card, as shown in *Fig. 7*. The frequency-response curves for the filter section are shown in *Fig.* 8.

It may be argued that the presence of

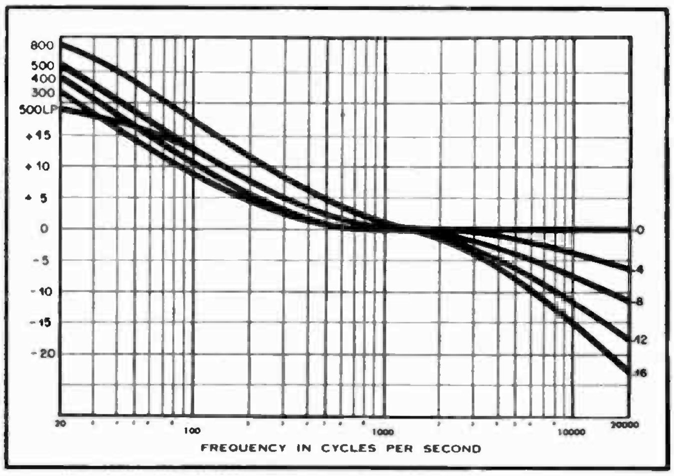


Fig. 5. Measured curves obtained from the preamplifier section.

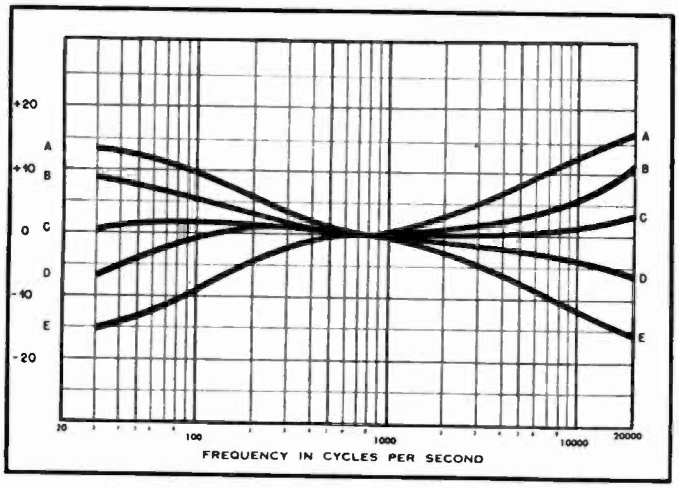


Fig. 6. Curves showing the limits of tone control action, together with an intermediate curve for both boost and cut.

the filter network after the output of the cathode follower detracts from the advantage of the follower as an output source—usually considered desirable in case the power amplifier is to be located at some distance from the control section. However, for this particular application the two units are to be located close enough that a three-foot cable will be used for making the connection, and the impedance of the output under the worst condition—that is, at "flat"—is of the order of 11.000 ohms, and a shunt capacitance of 300 μμf could be tolerated with a droop of only 3 db at 50,000 cps. Under these conditions, therefore, it was considered preferable to locate the filter at the output of the control-unit rather than at some earlier section of the circuit.

Volume and Loudness Controls

In order to provide loudness-control action in the amplifier, and yet to be able to use as much or as little compensation as might be desired, both types of controls are provided. A 0.25-meg volume control is located between V_{3b} and V_{4a}, and the compensated control is located between V_{4a} and V_{4b}. The Senior Compentrol employed here consists of two controls with concentric shafts—the volume control being operated by the outer shaft and the loudness control being operated by the inner shaft. Any degree of compensation can be obtained quite readily. The response curves of the loudness-control section, *Fig.* 9, are shown for the maximum-volume condition, and for conditions where the 1000-cps level is 10, 20, and 30 db below maximum, respectively.

Circuit values have been adjusted for minimum IM distortion at normal output, resulting in a figure of 0.7 per cent at 2 volts. There is adequate gain from any tuner with which the amplifier has been used, and the output from typical LP records is within 6 db of that from the AM and FM tuners being used.

Considered from the standpoint of the number of controls in the amplifier, this unit seems to have overstepped the boundaries of the current trend toward simplification. However, viewed from the front panel, there are only four knobs—apparently—and these are arranged in a convenient fashion. There is sufficient flexibility for the most critical listener, yet for the other members of the user's family—those who would normally be confused by eight controls—the preamplifier switch may simply be left in the 500LP and − 12 position, and the tone, presence, and filter controls can be left in the "flat" position. Either of the level-controlling knobs may be used to set volume. The choice of knobs and switch position arrangement on the preamplifier control is such that the 500LP and the − 12 position are coincident, and could easily be arranged to be "straight up" in the normal playing position for LP records. Similarly, the two indicating points on the tone-control are arranged to be "straight-up" when in the flat position, and the control tapers are designed to have a "flat" range of at least 15 deg. The non-indoctrinated user can be told to leave the filter and presence control alone—if that would solve the problem.

Construction

The construction of this unit follows fairly simple lines, with the possible exception of the resistor board which was used in the prototype. While there is no particular need for this type of construction, it does simplify the work to some extent.

Figure 2 shows the resistor mounting card used in the original model. The resistor card, together with the tube sockets which are mounted on the card, was attached to a channel designed for mounting onto the chassis with rubber grommets. With the Genalex Z729 tubes, however, the extra care in reducing microphonics has turned out to be unnecessary, and the sockets may just as well be mounted on the chassis. Adequate room is available on a 3 × 5 × 13 chassis, although the original was made on a 3 × 6 × 14 base—a size which is rather difficult to obtain. Layout of the terminal card is not difficult, but since every constructor has his own ideas of just how he wishes to do the job, specific layout is not indicated. With the Alden terminals, however, it is possible to arrange the terminals so that all of the grid-circuit components—as well as the cathode resistors and their by-pass capacitors, and the compensating resistors and capacitors—may be placed on one side of the board, and the plate and screen resistors, together with the decoupling resistors and certain other components can be placed on the other. This requires that the bottom of the chassis be cut out to provide access to the "bottom" of the resistor board, since changes may be required during the construction or—possibly—afterwards when service becomes necessary.

The mica capacitors used in the equalizer circuits may best be Silver Micas, since they may be obtained with the closest tolerance. The small amount of shift in values can readily be tolerated without apparent effect, but most constructors will wish to take advantage of the 10 per cent tolerance of this type of component. The two inductances listed may be any type of toroid available. If the entire installation is to be used at some distance from any hum-producing fields—such as a phonograph motor—

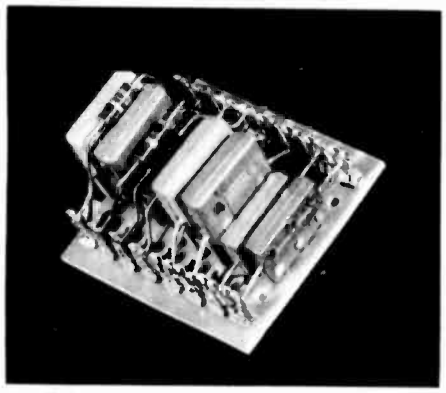


Fig. 7. Mounting of the components for the low-pass filter circuit. The input impedance-adjusting resistors R_{23} to R_{24} are mounted on the switch.

less expensive units may be used. But for complete freedom from the effect of any external fields, the toroid is the answer.

The original unit built by the author takes the place of one which has been in service for several years, and is powered by the same supply that has been in service since it was originally described in the series on "Residence Radio Systems," published in 1948. This supply was designed to furnish 12 volts d.c. at a maximum drain of 1.0 amps, and with a relatively low ripple. The unit has been tested with a 6.3-volt a.c. supply and has been found entirely satisfactory, provided the latter is suitably center-tapped to ground the circuit as near to the zero-potential point as possible.

While the circuit shown makes no provision for switching between phono. radio, TV, tape recorder, or any other possible source, there is no reason why the builder could not place another switch in the circuit and arrange to accommodate any desired number of signal sources. However, most modern tuners provide switching facilities, and this unit is designed to furnish only the basic requirements of a sound system—the preamp, the tone-control facilities, and some means for controlling the volume.

Reference to *Fig.* 7 will show the arrangement of the components—except the inductance—used in the low-pass filter circuit. The four capacitors at the left are shown with their shunting resistors, and are—from left to right—C_{22} to C_{25} respectively. The next four capacitors are C_{26} to C_{29} respectively. One end of each of the capacitors and resistors is grounded, and leads connect from the other terminals to the two arms of Sw_3. Construction is not critical, since the impedances are low at this point in the circuit.

Special Components

Special units are listed for several of the components, but all are available. The preamplifier switch—consisting of two 5-position decks, one with one arm and one with two, and both being actuated by a dual-concentric assembly—is listed as part number SPP-3002, a Centralab Type 30c switch. The Baxendall tone control unit is available completely assembled as Centralab part number C3-300, and consists of the two pots assembled as a dual-concentric unit together with the Couplate. The combination presence control and filter switch is available as Centralab part number SPB-3001. The Senior Compentrol is already available as Centralab part number C2-100. All units have standard bushings (3/8–32) 5/8 in. long, with the inner shaft 1 9/16 in. long, measured from mounting surface, and the outer shaft 1 1/16 in. long, measured from the mounting surface. This provides sufficient room for any panel up to 5/8 in. thick.

Several types of knobs are obtainable for these controls, since the dual-concentric control has become quite common on TV sets. The writer's TV set is a "home-made" Techmaster 630 mounted in a bleached oak corner speaker cabinet, and the knobs used are those normally used on the RCA 8TS30 receivers. To have matching knobs, the same type were used. Tan knobs are obtainable as stock numbers 73227 and 73231; the dark brown knobs are stock numbers 73226 and 73230. Four of each are required. Other types of knobs may be obtained from various sources, but may entail some searching. All of the Centralab units may be obtained on order from any Centralab distributor.

Performance

The performance curves of the several functions of this amplifier have been shown in *Figs*. 5, 6, 8, and 9. The presence control may appear to be another example of painting the lily, and in this era of attempting to obtain a response curve which is completely flat from zero to infinity may appear to be unnecessary, but it is only suggested that a quick trial be made of the idea before passing judgment. Like many circuit arrangements, it is not always possible to pass upon their effectiveness until one has an opportunity to listen to the results, and it is this writer's opinion that "somthing new has been added" when the presence control is turned from off to on. The effect is neither ear shattering nor eye opening, but as one approaches the optimum in performance it requires only a small change to outwit the law of diminished returns. If we can "cheat" the response curve just enough to make the change reasonably perceptible, we have accomplished something worthwhile, and the time and money have not been spent in vain.

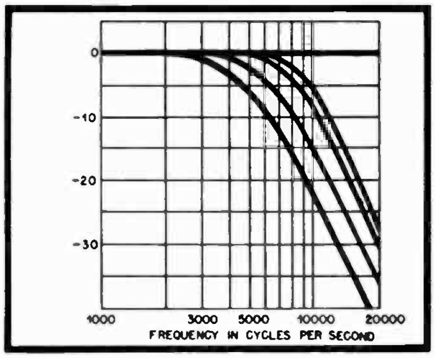


Fig. 8. Curves of low-pass filter action.

Parts List

C_1	.05 µf 600 v., Aerolite
C_2	.002 µf, molded paper
C_3, C_5	0.5 µf, 600 v., paper
C_4, C_{16}, C_{17}, C_{18}, C_{20}	0.1 µf, 600 v., Aerolite
C_6, C_8, C_{13}	2000 µµf, mica
C_7	1200 µµf, mica
C_9	3300 µµf, mica
C_{10}	200 µµf, mica
C_{11}	620 µµf, mica
C_{12}	1000 µµf, mica
C_{14}	3500 µµf, mica
C_{15}	150 µµf, mica
C_{19}, C_{33}	50 µf, 6 v., electrolytic
C_{21}	0.25 µf, 600 v., Aerolite
C_{22}, C_{26}	1100 µµf, mica
C_{23}, C_{27}	1600 µµf, mica
C_{24}, C_{28}	3000 µµf, mica
C_{25}, C_{29}	5600 µµf, mica
C_{30}, C_{31}	20–20 µf, 350 v., electrolytic (Mallory FP-227)
L_1	1.0 H, toroid inductance
L_2	0.45 H, toroid inductance
R_1	47,000 ohms, ½ watt
R_2	3900 ohms, ½ watt
R_3	0.5 meg, 1 watt, low noise
R_4	1.5 meg, 1 watt, low noise
R_5	1.2 meg, 1 watt
R_6	1200 ohms, ½ watt
R_7, R_{13}, R_{18}, R_{19}, R_{22}	0.1 meg, 1 watt, low noise
R_8	0.47 meg, 1 watt
R_9	0.1 meg, ½ watt
R_{10}, R_{11}	33,000 ohms, ½ watt
R_{12}	1.0 meg, ½ watt
R_{14}, R_{34}	27,000 ohms, 1 watt
R_{15}	50,000 ohms, linear pot, (combined with Sw_3)
R_{17}	1800 ohms, ½ watt
R_{20}	18,000 ohms, ½ watt
$R_{21a, b}$	Centralab Senior Compentrol
R_{23}	1000 ohms, 1 watt
R_{24}	6800 ohms, 1 watt
R_{25}	13,000 ohms, ½ watt
R_{26}, R_{30}	11,000 ohms, ½ watt
R_{27}	8200 ohms, ½ watt
R_{29}	12,000 ohms, ½ watt
R_{31}	10,000 ohms, ½ watt
R_{32}	7500 ohms, ½ watt
R_{33}	5600 ohms, ½ watt
R_{35}	10,000 ohms, 1 watt
R_{36}	5600 ohms, 2 watt
R_{37}	3900 ohms, 2 watt
R_{38}	1.8 meg, 1 watt
Sw_1, Sw_2	Preamplifier switch, Centralab 30c type, special (see text)
Sw_3	Centralab 30a type, combined with R_{15}, special.
J_1, J_2, J_3	Amphenol 80-PC2F receptacle
J_4	Amphenol 80-C receptacle
Tone Control	Centralab Couplate No. 12614, with 1.0-meg front section and 0.5 meg tapped rear section dual-concentric controls
V_1, V_2	Genalex Z729
V_3	12AX7
V_4	12AU7

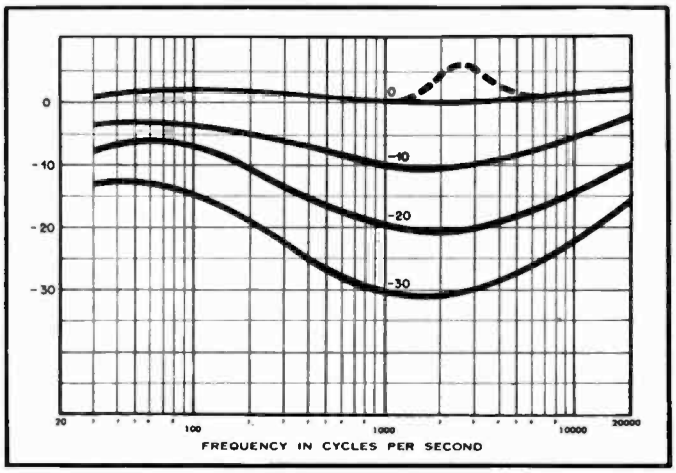
Notes on the Preamp with Presence

Numerous readers have indicated that the 0.45 H choke listed in the original data was unobtainable. Since toroids with inductances of 0.4, 0.5, and 0.6 H are regularly available, the following table indicates values for the low-pass filter section for use with inductances of these values:

	INDUCTANCE		
Component	**0.4**	**0.5**	**0.6 H**
C_{22}, C_{26}	.0012	.001	.00082
C_{23}, C_{27}	.0018	.0015	.0012
C_{24}, C_{28}	.0033	.0027	.0022
C_{25}, C_{29}	.006	.005	.004
R_{25}	11,000	15,000	18,000
R_{26}	10,000	12,000	15,000
R_{27}	7,500	9,100	11,000
R_{28}	5,600	6,800	8,200
R_{29}	10,000	13,000	16,000
R_{30}	10,000	12,000	15,000
R_{31}	9,100	11,000	13,000
R_{32}	6,800	8,200	10,000
R_{33}	5,100	6,200	7,500

The design voltage for the preamplifier was 275 volts, but satisfactory performance should be obtained with any voltage between 250 and 300.

Fig. 9. Response of the Senior Compentrol for maximum and for −10, −20, and −30 db output at 1000 cps.



Miniaturized "Preamp with Presence"

C. G. McPROUD

Simplifying the construction of this popular circuit by eliminating the low-pass filter section, reducing the phono positions from five to three, and employing a printed circuit panel for the major part of the wiring.

THE "PRESENCE" CONTROL, first introduced to the home music system in these pages over a year ago, has proved to be quite popular with those who appreciate an adequacy of control facilities—perhaps just that little shade of boost in the midfrequencies to move the principal instruments or singers a little further toward the spotlight.

Popular as the circuit was—and still is, for that matter—there is no denying that it was fairly complicated, that the two toroids used in the original model are expensive and not readily available on jobbers' shelves, and it is rather larger than usual for most control units of the present day. Coupled to that is an increasing interest in printed circuits, which are admirably adapted to production quantities but just short of impossible for the average experimenter. Anyone can design the circuit panel, but getting it built at a reasonable cost is something else again.

However, two of the problems have been eliminated in this design. In the first place, the printed circuit panels are already built, and are available.[1] Thus the construction is reduced to one of assembly and soldering, with reasonable assurance that the unit will work as described when it is completed.

[1] See last paragraph of article.

Circuit Requirements

Before designing an amplifier circuit, it is first necessary to consider what facilities are to be required. A phono preamp is a necessity in a "front end," of course, so that is included. In the light of modern LP records, however, it is not felt necessary to provide as much flexibility today as it was two years ago, for most records will play quite well on the RIAA or New AES curve. For the older 78's, a higher turnover frequency is considered desirable, together with less rolloff. In this instance a compromise was made—a 650-cps turnover and a rolloff of 8 db at 10,000 cps—which is apparently a satisfactory choice judging from listening tests. A much lower turnover frequency is required for the early foreign records, so this curve was adjusted for a turnover frequency of 300 cps, and the high end was left flat—any touching up in that region can be done by the tone control.

The presence control was considered a necessity. It does not have a large effect on the reproduction, but it does provide the midrange boost that aids in making the instruments stand out slightly. This control gives a bump on the response curve of only 5 db maximum, with some effect extending about one octave each side of the resonant frequency.

Volume and tone controls are always required, and the writer believes firmly in the loudness control as a necessary adjunct to a modern music system. Since a combined volume and loudness control is available at jobbers, this unit was selected as being simple and effective. The tone control system is the same as that used in the original "Preamp with Presence"—a circuit first described by Baxendall, and adapted to a dual concentric control and a single printed circuit unit with most of the components combined into a package not much larger than a postage stamp.

The output section should be a cathode follower to provide for those installations where the front end would be located at some distance from the power amplifier. In addition, it was desired that the selector switch should silence all inputs except the one in use to avoid crosstalk under all conditions; and an additional output was desired to permit feeding a tape recorder without any effect from volume, loudness, or tone controls.

Figure 1 shows the top and panel view of a completed unit arranged to plug into a National Horizon 20 amplifier, drawing all operating power from that source, and feeding signal to the power section through an Amphenol plug on the rear panel. *Figure* 1a shows the suggested construction, a design that permits placing the amplifier in a conventional cabinet, supported only by the control bushings. In this form the entire unit is 2 in. high by 7¾ in. long by 4¼ in. deep. While it is probable that the unit could have been made somewhat

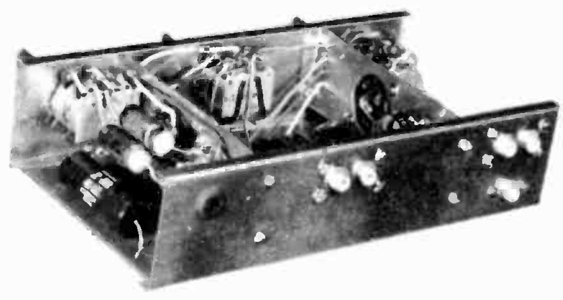


Fig. 1a (left). The completed preamp arranged for use with National Horizon 20 power amplifier. Fig. 1b (right). Finished preamp in recommended arrangement for use with any power amplifier, and suitable for mounting by means of control bushings.

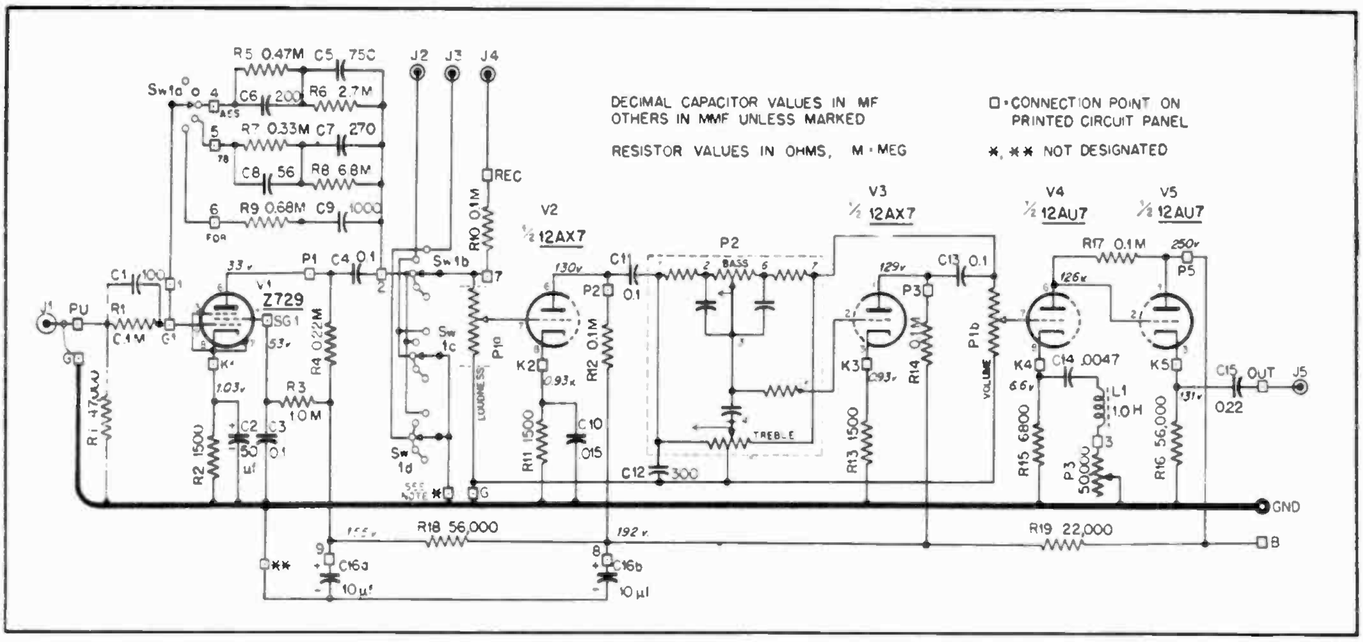


Fig. 3. Schematic of miniaturized preamp with presence.

smaller, it must be remembered that the tubes and a decoupling capacitor must be accommodated, and that the controls take up some space. In fact, it is recommended that the components be placed on the printed circuit panel before the tubes and the capacitor are put in place, and before the controls are mounted.

Figure 2 shows the unit of *Fig.* 1a in the construction stage.

Circuit Description

The circuit arrangement is fairly conventional. The phono pickup is fed into the grid of the first tube which serves only as a preamplifier. Equalization is introduced by means of feedback around the first tube, much in the same manner as used by Leak in the TL/10 amplifier control unit, which is simple, effective, and easy of adjustment. In *Fig.* 3, feedback is shown from the right side of C_4 to the grid of V_1, with R_1 and C_1 serving as the stabilizing load into which the feedback voltage is fed. This practically eliminates the impedance of the pickup as part of the feedback network, and by varying the size of C_1 the response may be kept flat up to 20,000 cps, if desired. Three feedback networks are shown—C_5, R_5, C_6, and R_6 provide the RIAA or New AES curve; C_7, R_7, C_8, and R_8 provide the 78 curve; and C_9 and R_9 provide the FOREIGN curve.

Referring only to the AES curve, C_5 and R_5 control the turnover frequency, while R_6 adjusts the low-frequency rolloff, and C_6 adjusts the high-frequency rolloff. While there is some interaction between these components, it is not difficult to arrange practically any type of curve desired. With the components shown, the New AES curve is accommodated within 0.2 db. No rolloff—either high- or low-end—is provided for the FOREIGN curve, which accounts for the use of only one pair of components.

The preamplifier tube is the low-noise, high-gain Z-729 used in the original Preamp with Presence. At normal operating settings of the volume and loudness controls the hum-and-noise level is better than 60 db below the 1-volt nominal output with the heaters operated on a.c. With d.c. operation, the improvement is less than 3 db, which speaks well for the hum-free operation of the Z-729. Certain other types have been tested in the circuit, and while the 5879 has a low hum-and-noise level, its gain is lower, so the usable range is some 6 db less. With d.c. on the heaters, selected 6AU6's perform satisfactorily, but the trouble of selection eliminates them from consideration for reliable use in the home.

The phono preamp is followed by the selector switch—a four-gang, five-position rotary switch. Two of the sections are used for shorting out the unused inputs, the third section for changing the equalization network around the first tube, and the fourth section as the actual input selector. The arm of this section connects to the top end of the loudness control section of the dual volume-loudness control unit, P_{1a}. Tape recorder feed is connected at this point through the 0.1-meg resistor R_{10} to provide isolation. The arm of the loudness control feeds the grid of the tone-control input stage, one half of a 12AX7. The cathode resistor is bypassed by C_{10} sufficiently to give flat response over the last four stages. The plate is coupled to the tone-control network, and this in turn is coupled to the tone-control output stage, the second half of the 12AX7, which feeds the volume control, P_{1b}, and its arm is connected to the grid of the output driver, one half of a 12AU7. The presence-control action derives from the partial bypass of this tube through the L-C network composed of C_{14} and L_1 and the 50,000-ohm potentiometer P_3. When the resistance is maximum, the cathode resistor is effectively unbypassed. As the control is turned, the resistor is bypassed more and more, but only at the resonant frequency of the tuned circuit, which is approximately

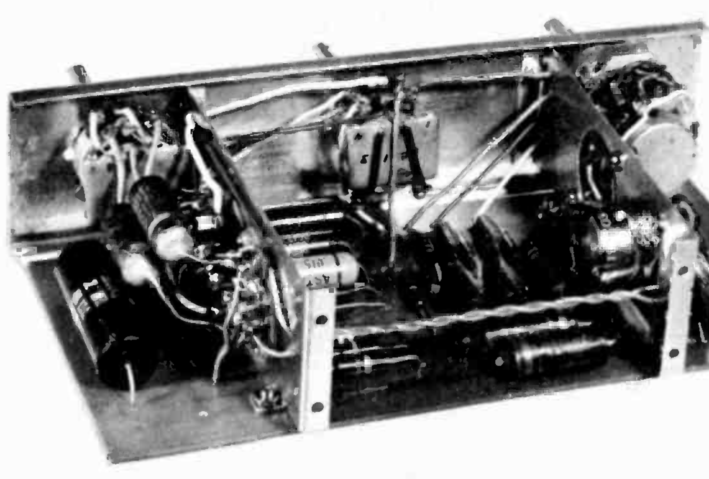


Fig. 2. Unit of Fig. 1a in construction.

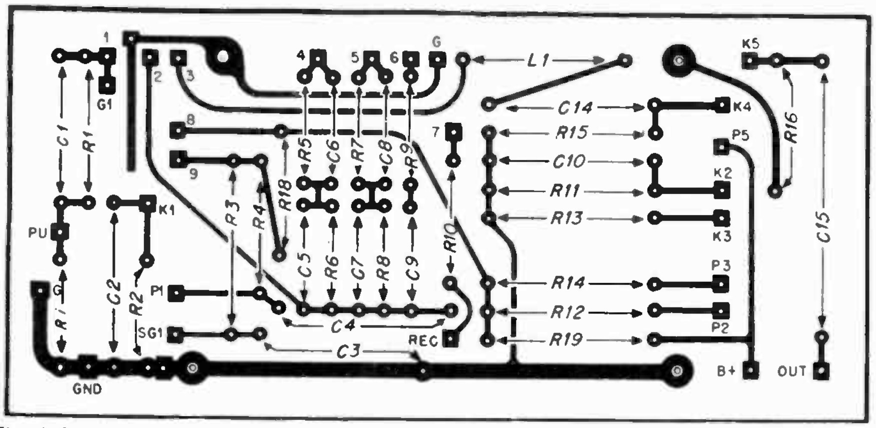


Fig. 4. Location of parts on printed wiring panel. Unmarked square between 1 and 2 connects to grounded arms of SW_1.

3000 cps. The cathode follower is directly coupled to V_4 and feeds the output line. R_{18} and R_{19}, together with C_{19a} and C_{19b} serve as decoupling networks.

It will be noted that all of the controls are the same as those used in the original Preamp with Presence, and supplied by Centralab. The selector switch is combined with the presence control, and was used originally as the low-pass filter switch. Both volume-loudness and bass-treble controls are the same as those used in the original model.

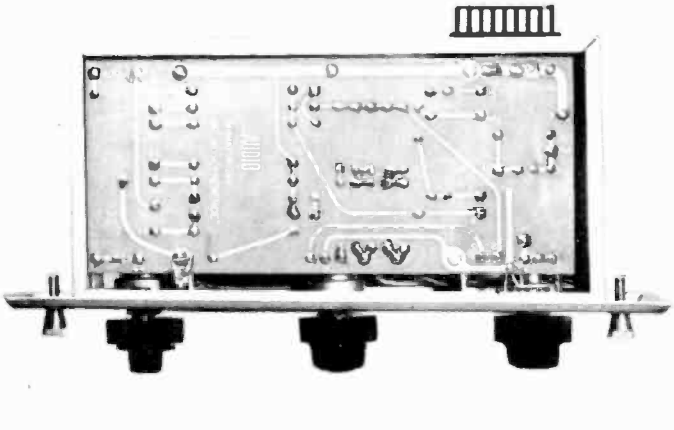
The presence control is located relatively far back in the amplifier, and is consequently at a higher-level point than previously, which reduces the susceptibility to hum pickup and thus eliminates the need for a costly toroid coil. The coil specified for this circuit is part number CS-1051, manufactured by Aladdin Radio Industries, Inc., Nashville, Tenn. The coil is wound on a powdered iron core, and the entire unit is encased in a powdered iron shell, with resulting low hum pickup and a Q of the order of 15 at 1000 cps. However, since this unit will not be available from normal outlets, it is to be furnished along with the printed circuit and the sheet metal components. Any desired arrangement may be employed, of course, and the circuit may be built up in a conventional form. However, it is much easier to build on the printed circuit panel and is likely to be more compact. Furthermore, the arrangement of parts has been tested, and the unit is almost certain to work immediately on completion.

Construction

The construction of electronic apparatus using printed circuits is very simple. In most instances the tube sockets become a part of the circuit panel itself, but for this unit it was desired that the tubes lay parallel to the panel so as to reduce the height. The process of building is simply that of inserting the component leads through holes in the panel, clipping them off 1/16 in. from the surface, and soldering them in place. This method is a complete reversal from the usual construction where it is always recommended that a strong mechanical joint be made before solder is applied. The components are inserted from the side opposite the one on which the wiring is, although leads can be connected from the wiring side with equal facility. In this unit, all components are located on the clear side, and are thus protected. Soldering presents only a slight problem—the principal caution being that a low-wattage iron be used. A person thoroughly experienced in soldering should have no trouble even with a 100-watt iron, but if too much heat is applied the copper is likely to lift from the phenolic panel or even to burn entirely off. With a 25-watt iron and a good multicore solder, the operation is comparatively slow, but it proceeds without any possibility of lifting the pattern. The copper coating on the panel furnished is tinned, and the usual procedure is to apply the tip of the iron to the lead and the wiring pattern simultaneously. The solder is then applied and the iron rotated around the lead slightly until the solder flows all around the lead, making a neat connection.

On the panel furnished, components are located as shown, the leads soldered into the small circles shown in *Fig*. 4. Connections from wires—such as those from power, input, and output connections—are indicated by small squares, and these are numbered. The schematic shows these squares. Connections to tubes, the decoupling capacitor, and other components are all indicated by the small squares.

Figure 5 shows the appearance of the underside of the panel after all the parts are soldered in place, while *Fig*. 6 shows the top of the panel with the components in place. The tube brackets are shown in *Fig*. 7, although they are supplied along with the circuit panel and L_1, and the front and back panels. The order of construction recommended is to mount



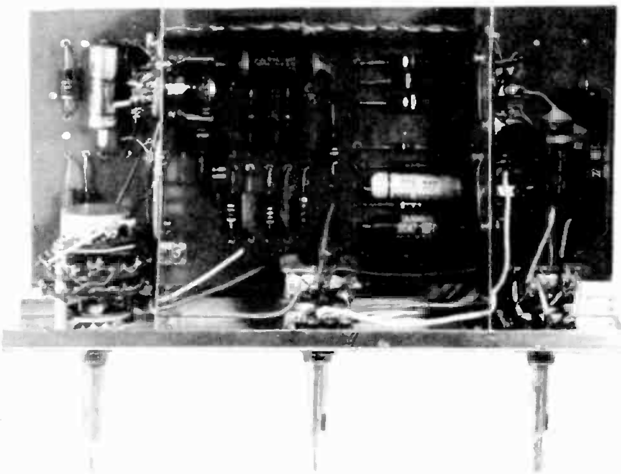


Fig. 5 (left). Underside view of completed amplifier to show soldered component leads. Fig. 6 (right). Top view of amplifier before attaching front and rear panels.

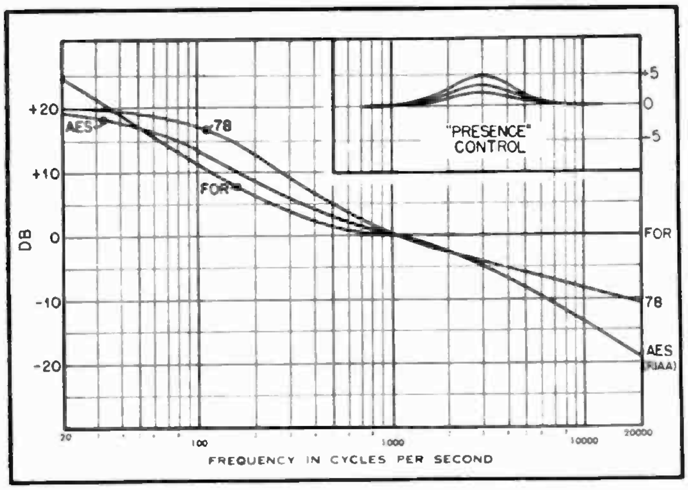


Fig. 7. Phono equalization and presence control curves.

the tube sockets and the capacitor wafer on the two tube brackets, and then assemble them to the wiring panel. Then insert all the components and solder in place, followed by the wire connections to the sockets. Leads should be connected to the controls and the latter mounted onto the front plate before attaching the plate to the brackets. The wires are then simply inserted in the proper holes and soldered in place. Small holes along the ends of the tube brackets are for leads running from one "compartment" to the other.

The back plate is next assembled with the two dual phono jacks and the single phono jack and the leads are connected in place before attaching to the tube brackets; the connections are then made and soldered. Because of the small space, the Z729 must be inserted in its socket and the shield applied before inserting the 12AX7 opposite it.

The voltages at the various points in the circuit are shown in italics in *Fig.* 3 for a supply voltage of 250. The circuit works satisfactorily with any supply voltage from 225 to 275, and if a higher voltage were to be used a series resistor should be connected between the source and the amplifier unit to reduce the voltage to a nominal value of 250. Total current drain is approximately 6.5 ma at a 250-volt supply. Filament requirements are 0.8 amps at 6.3 volts.

Performance

A 1-volt output is obtained with both volume and loudness controls at maximum—bass and treble controls flat—at an input of 0.09 volts from the high-level inputs, and from an input of 1.6 mv on phono at the AES position. The other two phono positions give a 1-volt output from an input of only 1 mv. This is somewhat more sensitivity than usual, but with two volume controls in series —one the volume control and the other the loudness control—some additional gain is required to provide sufficient range of control. IM distortion at the 1-volt output is approximately 0.4 per cent, rising to 1.5 per cent at an output of 5 volts, which is more than is likely to be used with modern power amplifiers.

Response curves are shown in *Figs.* 7 and 8, the former being a composite of the phono curves and the presence control. The maximum range of the loudness control on the Centralab unit is 35 db, which prevents the user from turning the level down to an absolute minimum with this section. In listening tests it is found that with the volume control set at its midposition, the output level through a usual amplifier is about normal for average use, and the loudness control operates to give suitable control over the listening range.

Figure 8 shows the range of the tone controls, together with the effect obtained with the Baxendall circuit when using the Centralab components indicated. The loudness compensation is shown in dotted lines.

PARTS LIST

C_1	100 µµf, silver mica, El Menco type CM-15
C_2	50 µf, 6 v, electrolytic, Sprague TVA-1100
C_3, C_4, C_{11}, C_{13}	0.1 µf, 600 v, metallized paper
C_5	750 µµf, silver mica, El Menco CM-19
C_6	200 µµf, silver mica, CM-15
C_7	270 µµf, silver mica, CM-19
C_8	56 µµf, silver mica, CM-15
C_9	1000 µµf, silver mica, CM-19
C_{10}	.015 µf, 400 v, molded paper
C_{11}	300 µµf, silver mica, CM-15
C_{12}	.0047 µf, 400 v, molded paper
C_{14}	0.22 µf, 400 v, molded paper
C_{16}	10-10/450, electrolytic, Sprague TVL2750
J_1	Single phono jack, Cinch 81A
J_2, J_3, J_4, J_5	Double phono jacks, Cinch 81B (2 req.)
L_1	1.0 Hy, Aladdin CS-1051
$P_{1a,\ b}$	Volume-loudness control, Centralab C2-100
$P_{2a,\ b}$	Tone Control unit, Centralab C3-300
P_3	50,000-ohm potentiometer, linear (combined with Sw_1) Centralab SPB-3001
R_i	47,000 ohms, ½ watt
R_1, R_{10}	0.1 meg, ½ watt
R_2, R_{11}, R_{13}	1500 ohms, ½ watt
R_3	1.0 meg, 1 watt
R_4	0.22 meg, 1 watt
R_5	0.47 meg, ½ watt
R_6	2.7 meg, ½ watt
R_7	0.33 meg, ½ watt
R_8	6.8 meg, ½ watt
R_9	0.68 meg, ½ watt
R_{12}, R_{14}, R_{17}	0.1 meg, 1 watt
R_{15}	6800 ohms, ½ watt
R_{16}, R_{18}	56,000 ohms, 1 watt
R_{19}	22,000 ohms, 1 watt
$Sw_{1a,\ b,\ c,\ d}$	See P_3
V_1	Z729 high-gain, low-noise pentode Genalex
V_{2-3}	12AX7
V_{4-5}	12AU7
	1—9-pin socket, molded, with shield
	2—9-pin wafer sockets

"Kit" Availability

This circuit can be constructed in conventional form, if desired, but it is somewhat simpler and more compact when assembled on the printed circuit panel with the chassis parts shown. The four chassis parts are of 18-ga. cadmium plated steel and are drilled and punched to fit the tube sockets and capacitor mounting plate, as well as for all wiring and mounting holes. The printed circuit panel is 1/16-in. phenolic. Also furnished is the 1.0-Hy. choke, which fits in place on the printed circuit panel, as indicated. These five items, together with specific instructions, are now available and the author will be pleased to advise interested readers the sources.

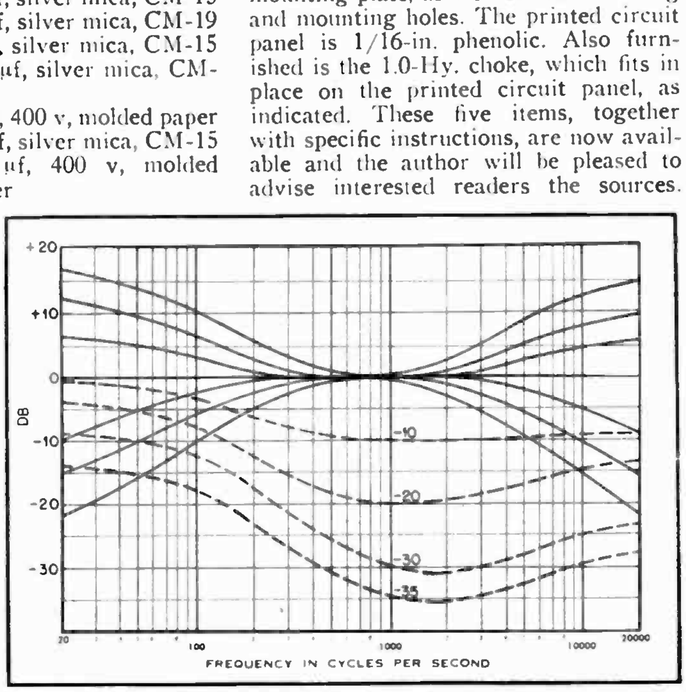


Fig. 8. Tone control and loudness contour curves.

New Approach To Miniature Preamplifier Design

ARTHUR J. ROSE*

Simplification of design requirements is said to yield a small and inexpensive unit with performance just as satisfactory to the ear as more standard products.

A study of manufacturers' specifications and of the technical literature will reveal that the major differences between phonograph preamplifier designs is mainly a matter of styling. The general preamplifier design recipe as usually followed calls for the following:

1. Equalizers for Recording Characteristics.
2. Means for switching inputs.
3. Gain control.
4. Frequency correction for room acoustics, speaker system variations, and human hearing discrepancies.
5. Means for overcoming system flaws, e.g. turntable rumble and record noise.

A restudy of the manner in which these goals are reached has shown that much can be done toward simplification of circuits, based on eliminating what is really superfluous even though "necessary" from the standpoint of convention. Once the absolute minimum requirements have been established, it is only a matter of fulfilling them with as little extravagance as possible. The problem of educating those who will benefit from the simplification is another, more difficult, matter.

In accord with this thinking—and in flagrant violation of convention—a pint-sized preamplifier was developed by the writer. It fulfills admirably the purpose generally served by larger, more complex units. The reduction in the number of parts brings the cost to an uncommonly low figure.

Frequency Correction

Bass and treble controls are usually thought necessary because of room acoustic and speaker-system variations. These are unchanging once the system has been installed in a particular location. Why, then, is it not possible merely to fix the positions of these controls after installation and then rely solely upon the record equalizer to match each record type to the corrected system? Overlooking the loudness versus hearing difficulties for the moment, the answer is rather obvious: equalizers do not adequately correct from record to record.

What record manufacturers do to their products is purely a matter of speculation. All the user has is a *recommended* playback curve. Before the average record sounds right the listener must do more than just switch in this particular curve. That in itself suggests that equalizers need more flexibility.

Fortunately, the type of correction afforded by certain bass and treble tone controls closely parallels that of the record equalizer. It appears that equalizers that do not have the flexibility of continuous variation around a particular characteristic might just as well be replaced by a fixed reference curve and then adjusted to suit the circumstances with a properly designed separate set of bass and treble controls.

This seemingly unscientific fact has met some stiff resistance from those who claim they must have *some* place in the preamplifier where the signal is exactly the same as that entering the recording microphone. Their point is admittedly only an academic one, since they also admit that this condition has little bearing on the correctness of reproduction—particularly if tone controls are not constantly adjusted. The solution to the matter really lies with the problem of "a properly designed set of bass and treble controls."

Another point—more practical but also more subjective—lies in the fact that most records worth playing through a high-quality system in the first place are of recent vintage. The majority of

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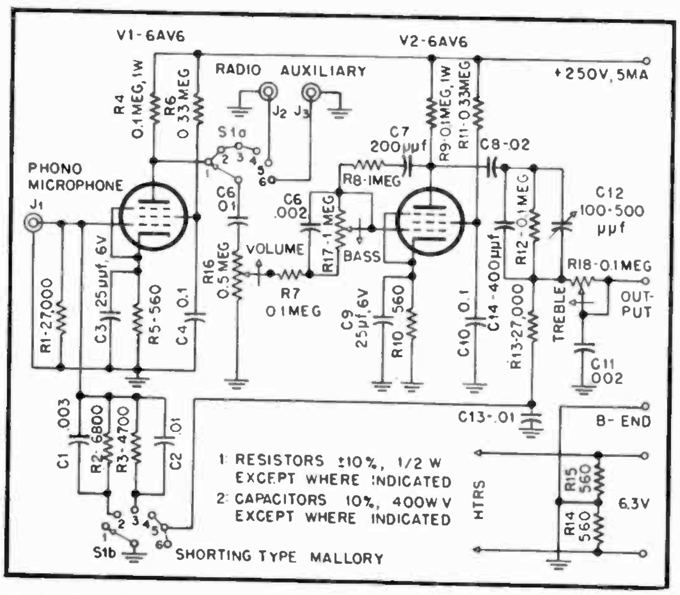

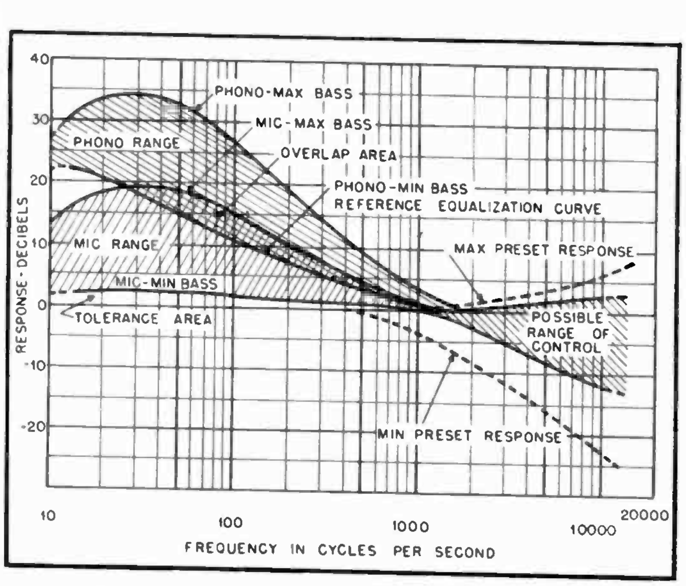

Fig. 1. (left) Schematic diagram of the 2-tube miniature preamplifier. Bass and treble controls are controlled by concentric shafts as the knob arrangement of Fig. 3 indicates. Fig. 2. (right) These curves show the maximum equalization ranges of the preamplifier and the reference curve discussed in the text.

the current recordings are very similar and there is promise that the RIAA "standard" may really *become* a standard!

Most recordings are made with a 500-cps bass turnover. In some cases there is bass pre-emphasis that requires a flattening of the playback curve below 50 or 100 cps. Treble pre-emphasis is usually at a fixed rate approaching 6 db per octave, with time constants ranging from 50 to 100 microseconds. For reasons made evident later, the writer's miniature preamplifier uses a straight 500-cps turnover without de-emphasis as a fixed reference curve.

To reduce the circuit to a minimum, only those functions that can be justified as essential are included. There is only a rare need for bass rolloff, especially at low and medium volume levels. A fortuitous circuit arrangement permits some bass droop at higher volume settings, but requires only the components for boost.

Need for large amounts of bass boost at low volume levels to produce proper scaling of sounds that were originally loud has given rise to loudness controls. Although their use has been justified more by the whims of bass-crazed audio hobbyists, it is desirable to have sufficient boost available to cover extreme conditions. The actual loudness correction to be used is subject to variation depending upon the relationship between the playback and the original levels, with due consideration to individual hearing. It is evident that there are a great number of possible circumstances and any compensation fixed solely as a function of playback level would prove to be incorrect in most of them.

Using terminology which divides hearing and acoustic corrections into two functions is paradoxical. It would be interesting to learn how someone in average circumstances can make a purely acoustic correction without benefit of hearing. Even sound-level meters are fitted with frequency-weighting networks to simulate the response of the human ear. Both acoustic and hearing correction can be and are made simultaneously, provided the maximum bass response is adequate.

The similarity of characteristics used as a basis for lumping corrective measures is more than just fortunate. It suggests to the writer that some types of correction may be imaginary and other types need stronger emphasis. Or viewing from another angle, there can be only one ideal resultant correction curve for a given situation, and a much smaller range of variation is needed to cover most situations than would be supposed.

Obtaining a large amount of bass boost can prove to be difficult if one wishes to reduce circuitry to a very simple form. Making allowance for normal boost requirements plus that needed for low levels has been solved by the writer with a simple circuit that applies a sort of automatic correction so that theoretical limits of boost are actually attained. This circuit has been used with success in supplying an 8-in. speaker with sufficient low bass to rival much more formidable systems. In itself, this fact shows that the need is only for improvement in circuit efficiency in most cases where bass is lacking, rather than for additional circuits.

In accord with this trend of thought, all functions related to the bass region are combined in the miniature preamplifier in one circuit, with a single control governing whatever adjustment is necessary. *Figure* 2 shows its range.

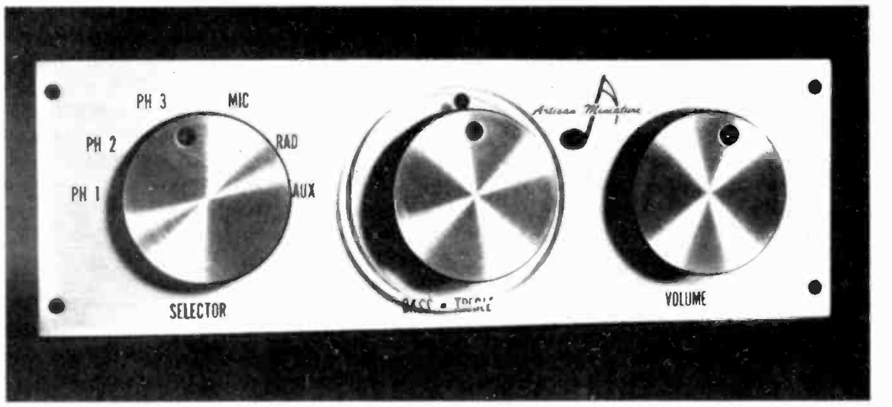

Fig. 3. As constructed by the author, the preamplifier is built in an aluminum box with this aluminum panel on the front. The outer shaft of the concentric bass-treble control is operated by a Plexiglas disc.

Treble Correction

Treble compensation can be simplified in like fashion. It has been general practice to apply 50 to 100 microseconds de-emphasis in the equalizer, and then by use of the treble control apply up to 75 microseconds further de-emphasis or emphasis. It is easy to see that some saving can be made by not including de-emphasis in the equalizer. This entails applying a variable de-emphasis to a response with fixed treble boost.

For example, a recording with 50 microseconds pre-emphasis initially has treble boost to the extent of 10 db at 10 kc. Why de-emphasize it to a flat response and then boost or droop it ±15 db? The same effect can be had without initial de-emphasis if the signal is fed into a system with a +5-db boost. Then, application of only rolloff to a maximum of 30 db will produce the desired result.

That sounds simple enough but there are some apparent drawbacks. De-emphasizing to the extent of 30 db in a simple circuit involves reducing amplitudes below the midfrequency region. This occurs because a single R-C network has a maximum slope of 6 db per octave. Another seeming drawback is that recordings vary in the amount of pre-emphasis, and radio inputs and some recordings have no pre-emphasis at all.

To justify intuition on the part of the writer, listening tests were conducted using a wide variety of recordings and program material with a considerable variety of associated equipment. In no case was there need to vary the treble response more than 15 db! It appears that all that is required for most purposes is a certain maximum response depending upon the associated

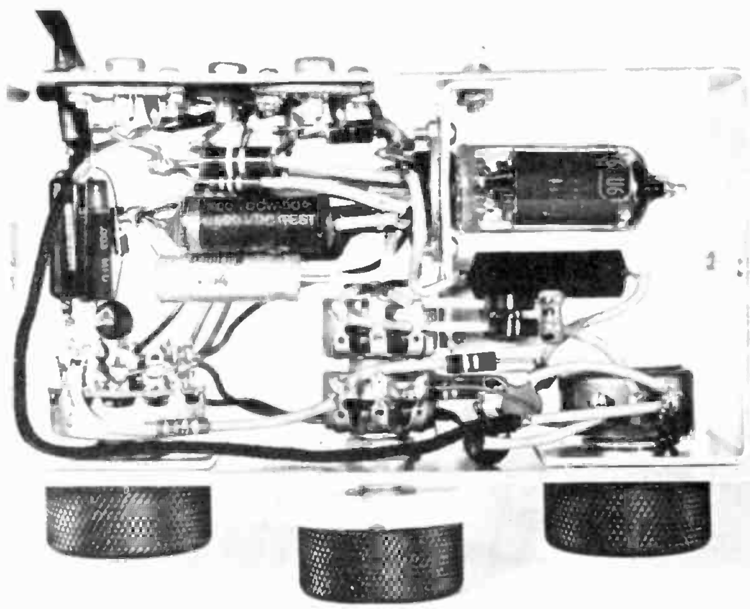

Fig. 4. Interior of the preamplifier. Size can be judged by the tubes. Even though the panel is just about large enough to accommodate the controls, there is plenty of space inside for components and wiring.

equipment. Therefore, it becomes entirely feasible to use only a treble rolloff control with a maximum droop of 15 to 20 db in a system with an *adjustable* boost. Once set for a given equipment lineup, the boost adjustment need not be altered. The range of net adjustment can be seen in *Fig.* 2.

The Miniature Preamplifier

The simplifications made thus far can be carried out in a spectacular manner. To illustrate, a very satisfactory preamplifier was constructed with a single tube! However, two tubes are used in the unit shown in *Figs.* 3 and 4 to bring the midband gain up to a more acceptable level for general use. Output with the two-tube arrangement is 2 to 3 volts with a 10-mv input, as compared to 0.5 volt output with a single tube (triode-pentode).

At first glance, the schematic of *Fig.* 1 may be somewhat confusing. Aside from the obvious functions that a closer study will reveal, there are several *hidden* effects derived from careful component arrangement. Perhaps the best manner of description would be an over-all picture and then a clarification of the details.

Three input jacks are provided. One is for a magnetic cartridge or microphone and the other two accommodate normal high-level, high-impedance devices. The 6-position selector switch performs the following:

Pos.	Name	Function
1	Phono 1	Normal phonograph position
2	Phono 2	Phonograph position for moderately noisy records
3	Phono 3	Phonograph position for very noisy records
4	Microphone	Position for using a low-level microphone in phonograph jack
5	Radio	Radio tuner
6	Auxiliary	Television sound or tape recorder

Positions 1, 2, and 3 retain the amplification of the first stage, keep a reference equalizer in the circuit and control the upper limit of frequency response. Position 1 yields a flat response, while 2 and 3 furnish cutoffs at 12 db per octave starting at about 6 kc and 4 kc respectively, by resonating the capacitor switched across the input grid with the inductance of the pickup.

Although one is inclined to shudder at the prospect of a 4-kc limit to response, the effect is not the same undesirable one that would be obtained with a filter of much higher slope—say 30 db per octave. Using 4 and 6 kc with the simple, well known filter arrangement shown in the schematic affords most effective removal of record noise and distortion with a minimum apparent loss of highs.

Position 4 retains only the 200-times, flat-frequency-response amplification of the first stage. The remaining two positions, 5 and 6, switch the radio and auxiliary inputs directly to the second stage. The reference equalizer is not effective in these last three positions.

Bass Circuit

Most of the burden is carried by the second stage. Variable bass boost is achieved by selective feedback from plate to grid. The plate circuit is loaded with the series arrangement of R_{14}, R_{15}, and C_{15}. Output is taken through the treble-control potentiometer. This configuration forms the backbone of the circuit.

In the phonograph positions, C_{15} is effective. In other positions, it is shorted to ground. While in the circuit, it provides the reference equalization curve. Not so readily apparent is the fact that this capacitor in conjunction with R_{14} and R_{15} forms a *frequency-sensitive* load on the second stage. This causes the amplification to rise somewhat with lowered frequencies. Therefore, the feedback bass control is exceptionally efficient and gives about 6 db more boost than would normally be expected. As an added result, the equalization curve approaches the theoretical ideal. This gives additional boost with a straight-lined 6-db-per-octave slope instead of the usual rounded and depressed curve.

There is further advantage to this circuit arrangement. Treble rolloff is easily accomplished, but more significant, cable capacitance can be fully offset! This eliminates the need for a low-impedance output such as a cathode follower. Theoretically, the upper response can be in the megacycles if desired.

With an additional step, complete treble control is obtained. By overcompensating the cable capacitance, the upper frequency response can be boosted to any acquired amount. An adjustable trimmer C_{12} performs this operation. The rolloff circuit is designed to give a maximum attenuation of about 20 db at 10 kc. The range of control thus afforded is complete for most imaginable circumstances.

There are several other design features worth noting. Coupling constants are chosen to give a fairly steep attenuation below 20 cps, providing a good measure of rumble and flutter elimination. At higher volume-control settings, the balance between the two branches of the feedback bass-control network is upset and bass droop is effected at the extreme lower settings of the bass control. Should additional droop be needed for a phonograph input, the selector switch can be set to the microphone position. To place FM reception on a par with recorded material, the 75-microsecond de-emphasis network at the output of most receivers can be bypassed.

Since the input stage is non-frequency-sensitive, maximum signal-to-hum is realized. With the bypassed cathode arrangement, hum is extremely low. With careful wiring and appropriate regard to grounds placement (mainly a matter of returning the heaters to a separate and lower-than-signal ground) hum can be reduced to 60 or 70 db below signal level.

Pentode operation in the first stage favors signal-to-noise ratio as compared to an equivalent triode because of the low input signal. Distortion is somewhat higher. Even so, it is a matter of arguing trifles. Distortion is not discernible at any level. Measurements indicate the order of 0.25 per cent at the overload point and .05 per cent with normal settings.

Much remains towards simplification of the means of controlling sound reproduction—particularly for home use. It is hoped that future trends in audio equipment design will be to accomplish the necessary results in a more concise manner than has been observed to date.

Parts List

C_1	.003 µf, 400 v, paper
C_2, C_4, C_{13}	.01 µf, 400 v, paper
C_3, C_9	25 µf, 6 v, electrolytic
C_5, C_{10}	0.1 µf, 400 v, paper
C_6, C_{11}	.002 µf, 400 v, paper
C_7	200 µµf, 400 v, ceramic
C_8	.02 µf, 400 v, paper
C_{12}	100–500 µµf, adjustable
C_{15}	400 µµf, 400 v, ceramic
J_1, J_2, J_3	phono pin jacks
R_1, R_{13}	27,000 ohms, ½ watt
R_2	6800 ohms, ½ watt
R_3	4700 ohms, ½ watt
R_4, R_9	0.1 meg, 1 watt
R_5, R_{10}, R_{15}, R_{14}	560 ohms, ½ watt
R_6, R_{11}	0.33 meg, ½ watt
R_7, R_{12}	0.1 meg, ½ watt
R_8	1.0 meg, ½ watt
R_{16}	0.5-meg potentiometer, linear
R_{17}	1.0-meg potentiometer, linear
R_{18}	0.1-meg potentiometer, linear
	(R_{17} and R_{18} assembled from IRC Concentrikit components)
Sw_1	2 pole 6 pos. switch, Mallory 3126J
V_1, V_2	6AU6

A Preamplifier Switching, and Equalizing Unit for Critical Listening

M. V. KIEBERT, Jr.*

Given a "cleaned-up" output amplifier, the next step is to follow the same steps in optimizing the preamplifier and control unit section of the complete system.

FOR THE ENGINEER and/or golden ear who enjoys good music and must live with the distaff side of a family not inclined towards the intricacies of equalizers, equalizer settings, and similar gadgetry, it is necessary to design a high-fidelity system to take into account this point of view and enable tolerable program enjoyment without personal supervision or the teaching of a course in the fundamentals of aural perception and audio circuits.

From this writer's point of view, a typical system is conveniently broken down into four sets of basic elements for the purpose of design consideration. These are, namely, the disc preamplifier and recording characteristic equalizing assembly; the system switching and environmental equalizing facilities; the power amplifier; and the loud speaker and associated acoustical system. These are shown in the block diagram of *Fig.* 1.

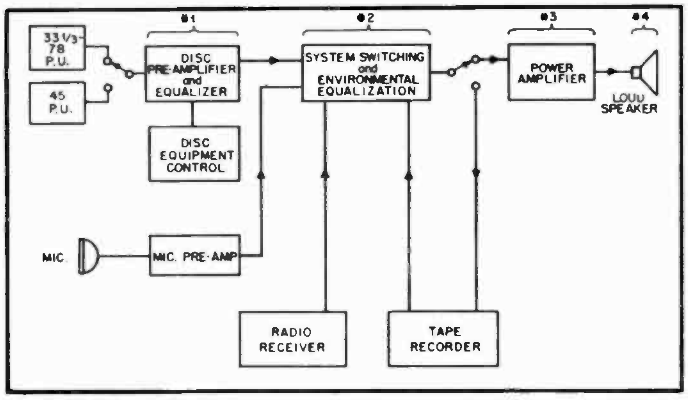

Fig. 1. Block schematic of the entire system.

Of these four elements listed above, only the preamplifier and recording characteristic equalizing assembly and the switching and environmental equalizing facilities pose a major audio design engineering job which is ordinarily under the control of the designer. This design problem is exacting due to the low levels involved, and to the fact that the switching controls and equalizing functions are most easily accomplished at relatively high impedance levels with their consequent sensitivity to noise and hum pick-up. Both of these latter functions are logically located in this control unit, "nerve center" position.

The loudspeaker and its enclosure—which constitute a complete acoustical system—together with the power amplifier, which in this particular application was a modified Williamson, are of essentially straightforward design or may be purchased units of basically linear characteristics and accordingly are assumed to require no additional equalization, nor to require other than an essentially flat frequency characteristic of good transient and intermodulation performance in order to provide satisfactory reproduction of the complex audio signal fed into these two elements of the system.

This article outlines the design, construction and performance characteristics of a combined preamplifier and recording characteristic equalizing assembly which is combined with switching and enviroment equalizer facilities. The final unit was engineered for installation in a coffee table, along with the radio receiver and both 33-1/3 and 45 r.p.m. turntables. The power amplifier, microphone preamplifier, and magnetic recording equipment, as well as the loudspeaker assembly, are separately located for convenience.

A separate power supply for this "nerve center" and the use of an output transformer provide for circuit isolation, thus avoiding stray circulating ground currents in the audio system. Residual hum and noise are therefore considerably below the threshold noise level of a quiet room, even when listening within one foot of the speaker.

The final unit more than meets the

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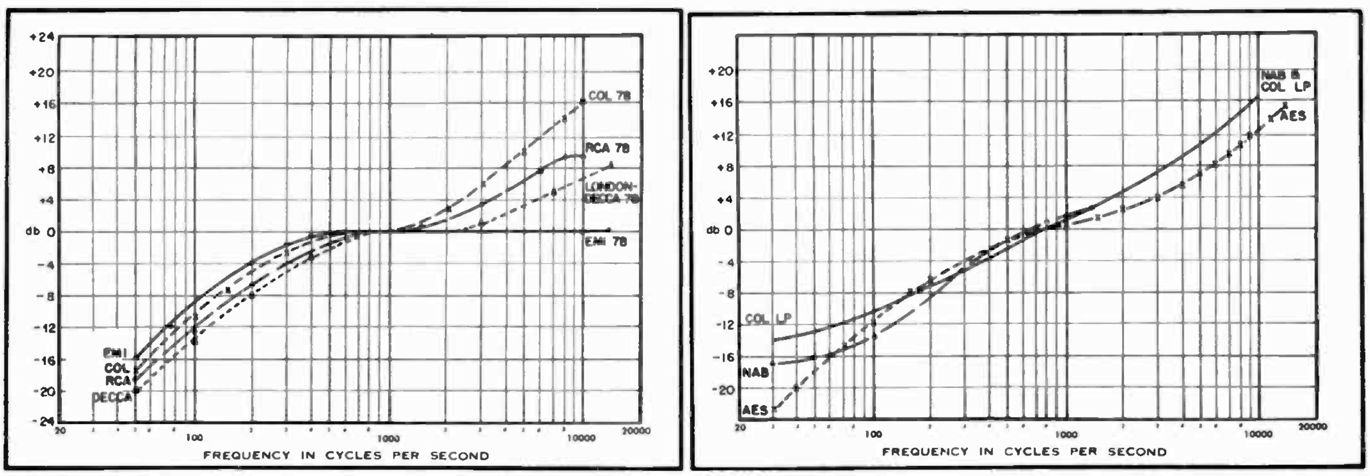


Fig. 2. Recording characteristics for which the preamplifier was designed. (A), left, those used by leading manufacturers of 78-r.p.m. records, and (B), right, those used for transcriptions and LP records, including the AES curve.

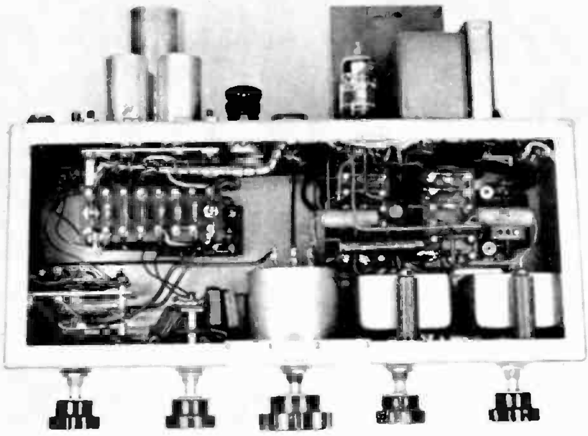


Fig. 3. The complete chassis of the preamplifier and equalizer control unit.

original design requirement of a system amenable to technically correct operation by unskilled personnel, while still providing adequate flexibility such that additional and adequate high- and low-frequency environmental equalization was readily available to satisfy individual listener preference and/or to compensate generally for various room acoustic anomalies.

Design Criteria

In considering this system, it was believed to be desirable that only one switch setting should automatically provide the proper and optimum low- and high-frequency compensation required for most discs, and that the midpoint setting of the environmental equalizer controls should provide a system which would have a "flat" over-all frequency characteristic at maximum volume setting with automatic compensation for the Fletcher-Munson curves at lower volume levels. It was desired that the environmental equalization be capable of either boost or attenuation of at least 20 db at both the high and low ends of the audio spectrum in order to provide for personal preference and/or area balance. Due to the fact that most natural phenomena do not change abruptly, and based upon many listening tests of various types and slopes of networks, the environmental equalization was based upon 6 db per octave (single degree of freedom) network swinging around a mid-frequency of approximately 1000 cps and more than critically damped in order to avoid difficulties.

In order to provide the optimum recording characteristic equalization, it was necessary to survey the recording industry both here and abroad, and secure the recording characteristics of each of the most widely used types of records. Accordingly, an investigation was made and reasonably accurate recording curves were made available from the major recording studios as follows:

EMI 78 (British Parlophone, Columbia. Brunswick, HMV, etc.)

Columbia 78 (U. S.)

RCA 78

London, Decca 78

Columbia LP, N.A.B., and the recommended A.E.S. curve

The recording characteristics for each of these types of discs are shown at (A) and (B) in *Fig.* 2. It is to be noted that while specifications and much of the literature indicate that the transition from a constant amplitude to a constant velocity curve is a discontinuous function, actual recording characteristics generally indicate coupled systems with only one degree of freedom with the attendant curves as shown at (A). Obviously and fortuitously, such curves are much easier to compensate than those which are discontinuous.

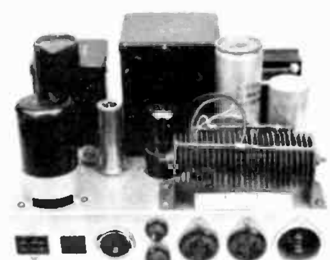


Fig. 4. Power supply used for the preamplifier unit.

In order to secure optimum signal-to-noise ratios with low-level pickups it was decided to utilize approximately 20 db of gain prior to any recording characteristic equalization. This was done in the first stage which, however, does have an output selector (Sw_1) in order that higher level pickups, such as the Pickering and Clarkstan units, may be used through the same equalizing stages. Use of a pre-preamplifier with a bypassed cathode resistor also helps the hum and noise problems which arise due to heater-cathode coupling. An equalizing preamplifier frequently recommended for this application gives rise to spurious hum and noise due to the unbypassed resistor in the input stage. When using the RPX040, RPX041 or RPX046 pickups, G.E. recommends the use of a 6SC7 with grounded cathodes and work-function bias, with equalization accomplished by means of a RC losser network between the preamplifier stages. This recommendation was probably based on their findings relative to unbypassed cathode resistors. However, test of the G.E. configuration as compared to the present unit or other units of more conventional design, does show poorer IM performance, which is probably a result of the greater non-linearity of their bias system.

Construction

Figure 3 shows the completed unit. The controls from left to right are as follows: Input selector Sw_3 (magnetic recorder, radio receiver, 45-r.p.m. record player, and 33-1/3—78 transcription table); record-equalization (record-type) control; Sw_2, loudness control; low-frequency environment equalization; and high-frequency environmental equalization.

The power supply for this unit, as shown in *Fig.* 4, has been built on a separate chassis in order to permit its location separate from the other unit.

In assembling components for use in the preamplifier assembly, shown in schematic form in *Fig.* 5, it was found necessary to utilize a bridge in order to select the correct values of capacitance for the particular value of equalizing feedback resistance utilized. Ordinary, stock type capacitors run approximately ±20 per cent tolerance, while ±10 per cent units appear to invariably run right to the limits of the tolerance and accordingly, the importance of utilizing the correct values of capacitance and resistance in the equalizing circuits cannot be too greatly emphasized if reasonably good compensation is to be obtained. Mica units of ±5 per cent tolerance have, however, been generally found to be very close to the correct value.

The other components of the system are not critical except that special attention should be given to the plate resistors in the first stages to insure that these units are of the "low-noise" type and that the resistors used in the feedback equalizing circuits have a negligible voltage-resistance coefficient.

Relative to low-noise-level and/or low-voltage-coefficient resistors, there are several points all too frequently overlooked by the casual audio experimenter—points which might well be outlined at this time.

For minimum noise, wire-wound units are always quieter than composition units, but before using wire-wound units in feedback circuits, make certain that they are non-inductively wound and in fact have a negligible reactance at frequencies up to 100 or 200 kc—otherwise excessive high-frequency phase shift will cause trouble.

In the present assembly a wire-wound, 1-watt unit used in the plate circuit of the pre-preamplifier improved the signal-to-noise ratio by 18 db when this unit was substituted for a ½-watt composition resistor—yet the composition resistor was being operated at only 20 per cent of its rated dissipation.

Composition resistors used in current-carrying loads such as plate circuits or unbypassed cathodes should be carefully suspect of being noise sources in low-level circuits. Where space and miniaturization appear to dictate the use of composition units, use 1- or 2-watt units, even though the dissipation requirement is only 0.1 watt or even

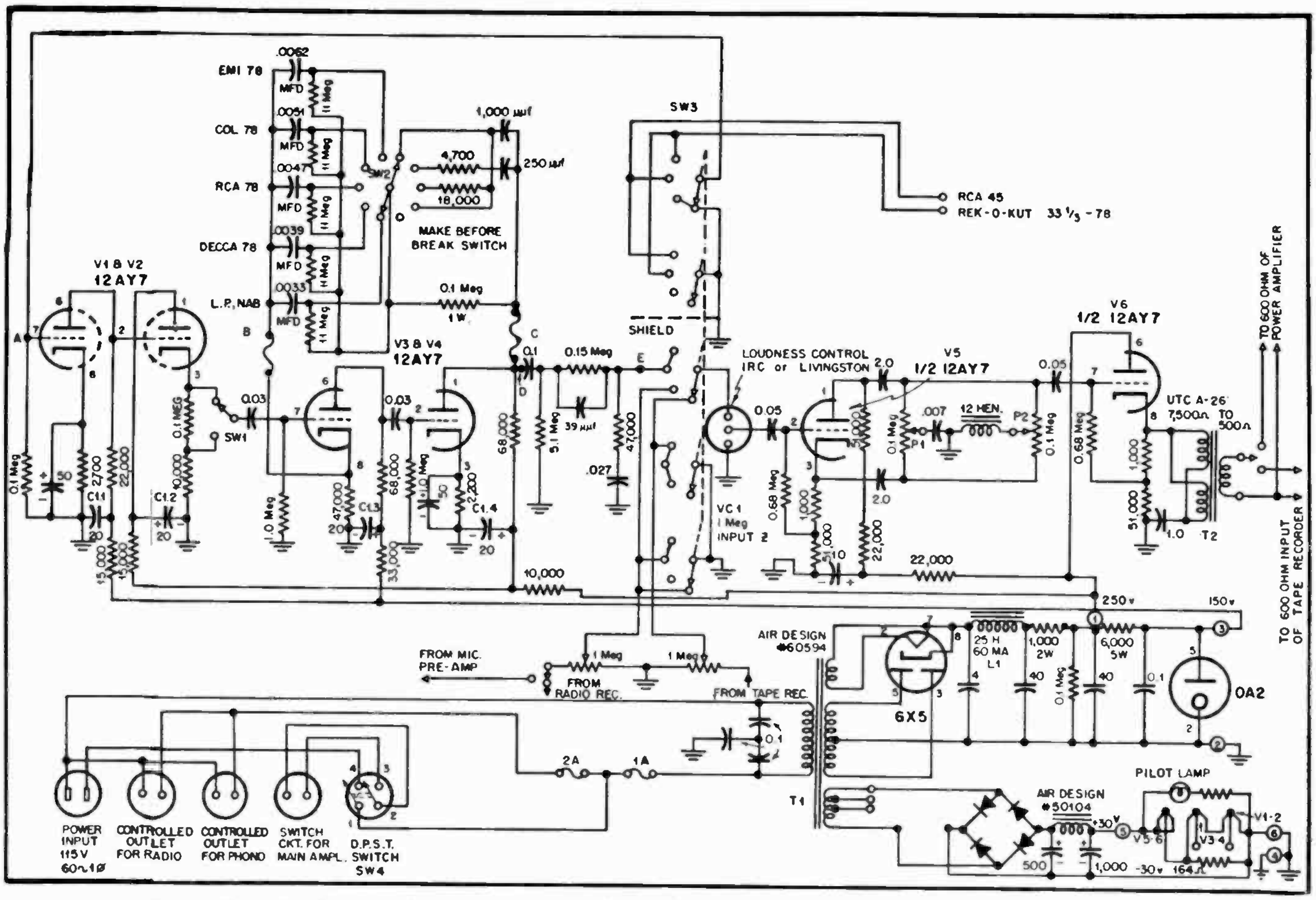


Fig. 5. Over-all schematic of the preamplifier and control unit together with the power supply shown in Fig. 4.

less. With these units used in place of the usual 1/3- or 1/2-watt units, it is frequently possible to pick up an extra 10 to 12 db of signal-to-noise ratio. Where appreciable d.c. currents are not normally present as in the case of the grid resistor and where the a.c. signals are not large, then the smaller ½-watt composition units are satisfactory.

In feedback circuits such as that found in the preamplifier recording characteristic equalizing stage, care should be taken to see that the feedback resistor and the associated cathode resistor are of the same type of composition material if space or component availability limits their items to composition elements. Under such a condition the two elements will act as a voltage dividing potentiometer and while the assembly may not have too good a voltage-coefficient characteristic, no appreciable difficulty will be experienced because they have a relatively high value of minimum bridging impedance—about 105,000 ohms as compared to the approximately 10,000 ohm output impedance (not considering equivalent output impedance due to feedback) of the 12AY7 output stage, across which this divider is placed. Were the 4700-ohm cathode resistor a wire-wound unit and the feedback resistor a composition unit, transient signals would be badly distorted—a point easily established when intermodulation tests are made on such a system.

The low-level coupling capacitors must likewise be carefully selected. Mica units or other high-quality types should be used. The miniature, metalized paper types are frequently found to be quite noisy when used in low level circuits.

It will be noted that the output of the recording characteristic equalizing amplifier contains an auxiliary low-frequency equalizing network to give a last fine touch to the low end of the recording characteristic compensation curve, a point which is all too often overlooked.

The output cathode follower circuit employs a bridging and isolating transformer in order to avoid the circulating ground currents normally encountered in single-ended outputs which usually give trouble in providing hum-free input to the power amplifier when this unit is located an appreciable distance away from the preamplifier. A bridging type of transformer is used in order that it may be terminated in a 600-ohm load without adversely affecting the distortion characteristics of the cathode follower. It is interesting to note that the 300-ohm output impedance of the cathode follower can deliver 4 volts r.m.s. into a bridging load with the IM kept to below measurable level. When this same stage is loaded by 300 ohms, then the output at 1 volt r.m.s. was found to have over 5 per cent IM distortion.

The pre-preamplifier and recording characteristic equalizer amplifier, along with all associated coupling capacitors and resistors have been mounted in a sub-assembly and this assembly shock-

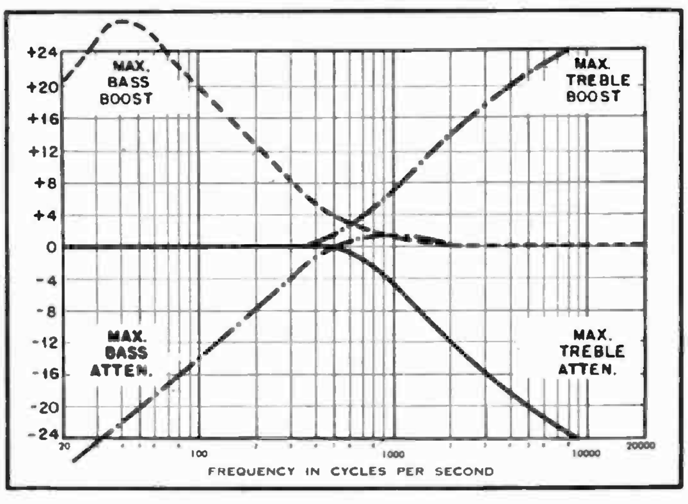


Fig. 6. Curves showing limits of equalization of the bass and treble tone controls.

mounted from the chassis with flexible leads for vibration isolation. This unit will be noted in the upper left corner of *Fig. 3*.

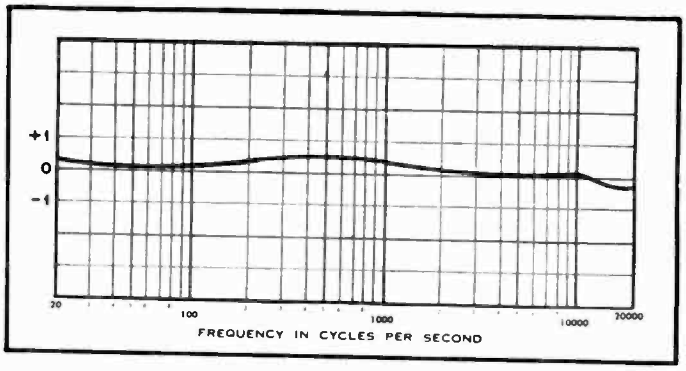
All the recording characteristic equalizing resistors and condensers have been mounted on Sw_2 with flexible leads at B and C extended to the shock-mounted pre-preamplifier and equalizing amplifier.

The "touch-up" low-frequency equalizing network between points D and E has been placed on the bottom of the chassis under the pre-amplifier sub-assembly.

Care was taken in wiring the unit to insure a single-point ground which would be located at the lowest signal level point. All input circuits were carefully isolated from ground with insulating washers used on the unbalanced input circuits. The actual ground point employed the chassis negative electrolytic bypass capacitor point which was located immediately and conveniently below the pre-preamplifier and recording characteristic equalizer-amplifier sub-assembly.

Care was also taken to insure minimum capacitance to ground of the interstage coupling capacitors between sections of V_5 and V_6 by mounting these two units on insulating stand-off pillars just back of the low- and high-frequency environmental equalization controls.

Fig. 7. Response through equalizer section with controls centered and with loudness control at at maximum level position.



Performance

Figures 6, 7, 8, and 9 show the actual reproduction characteristics, and confirm the fact that the recording characteristic equalizing circuits provide complementary equalization for the curves of *Fig. 2*.

Each element of this system was carefully investigated as to signal-to-noise ratios and overload characteristics.

The pre-preamplifier and associated disc-equalizing amplifier, for example, with a 15 millivolt (r.m.s.) input at 1000 cps provides 1.215 volts r.m.s. output to the loudness control with the noise level more than 80 db below this level. The overload point with the pre-preamplifier switch in the maximum gain position was at 80 millivolts r.m.s., which indicates that there is more than an ample overload margin when a G.E. pick-up is used under this condition of operation. The overload point in the minimum gain position, is of course, approximately 0.8 volt r.m.s. under the same conditions of operation.

The environmental equalizer section of the circuit was similarly tested, both at maximum boost and attenuation. Again by careful parts layout it was found that the noise level was more than 80 db below the normal signal level. The specific value was not measurable due to the fact that the vacuum tube voltmeter available for these tests could not read a lower level.

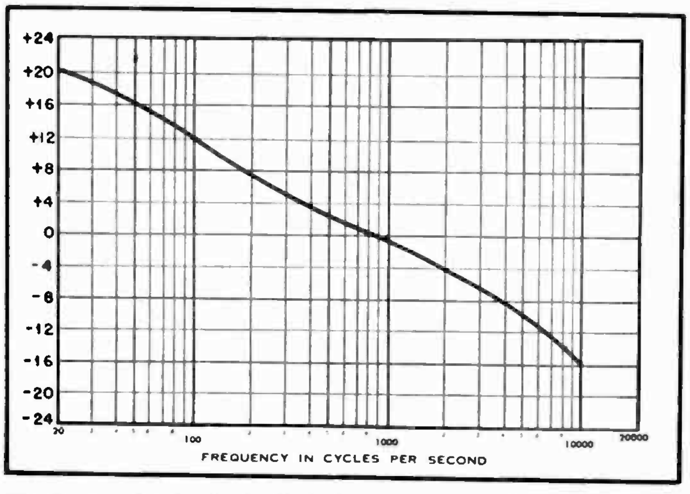
The environmental equalizer section was checked for overload characteristics and output voltage, both under the conditions of a terminating load and a bridging load. In the first case, 15.2 volts r.m.s. input appeared to be the overload point, which gave 5.0 volts r.m.s. into a terminating load, or 6.7 volts r.m.s. into a bridging load. With maximum boost and utilizing a 100 cps signal as a reference, the overload point was 10 volts r.m.s. input, which gave 2.55 volts r.m.s. into a terminating load, and 4.5 volts r.m.s. into a bridging load.

From the above, it is apparent that there is also ample margin between operating point and the overload point of the environmental equalizer section of this unit when it is noted that the maximum level feed into this section will normally be only 1.215 volts r.m.s.

Under normal conditions of operation, a 15-millivolt input signal as typical of a G.E. pickup, will provide a signal of approximately 0.55 volts r.m.s. across the 600-ohm output of the preamplifier, under which condition of operation a signal-to-noise ratio of better than 80 db is readily obtained. This level appears to be more than adequate as an input for a magnetic recorder and most conventional power amplifiers.

From past experience, it was believed to be essential that the excellent system performance should be obtained regardless of tube selection. Twenty 12AY7's in various combinations and of various previous operating histories were evaluated. Thanks to the type of circuitry, neither the gain, levels, nor frequency response changed more than 0.3 db regardless of their combination.

The writer wishes to acknowledge and credit George Beggs, Jack Shoup, Sherman Fairchild and Paul Landaman for many of the ideas and suggestions embodied in this unit and for suggesting the rather wordy descriptions of several design points often presented in the literature of this new field.



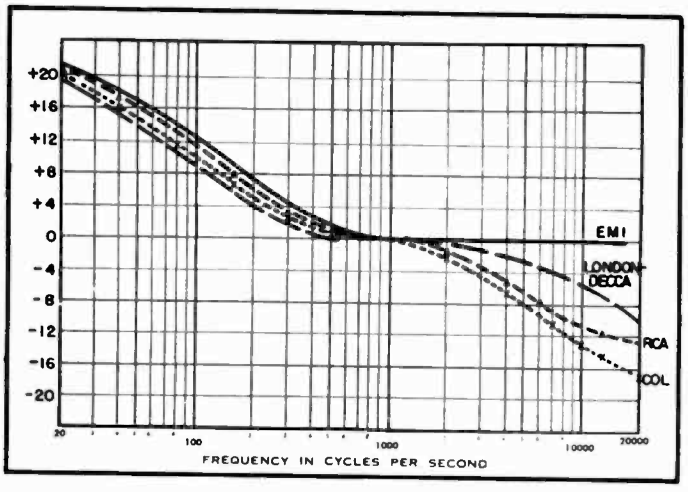
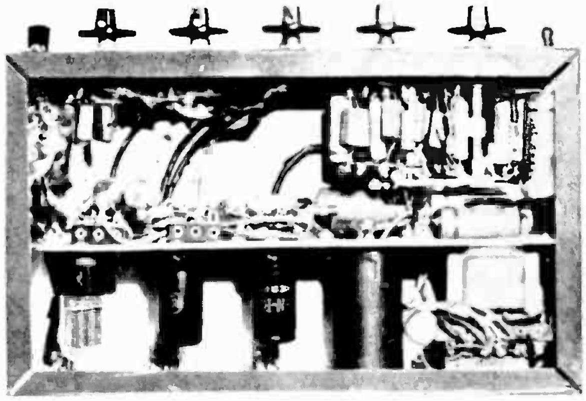


Fig. 8 (left). Measured playback characteristic for LP records. Fig. 9 (right). Measured playback characteristic at various settings of the controls for 78-r.p.m. records.



External and internal views of the author's self-powered control unit.

A Low-Distortion Tone-Control Amplifier

W. B. BERNARD, Cdr., USN*

A well-designed "front end" which should provide sufficient flexibility for anyone, since the curves can be tailored to fit by judicious selection of capacitors and resistors.

OVER THE PAST FEW YEARS the author has been working to develop an input amplifier and tone control circuit with fewer than the average number of faults. This article is a story of the evolution of such a circuit and *Fig.* 1 is the family tree of the circuit.

The circuit shown at (A) in *Fig.* 1 is a widely used circuit which provides for boost and cut for both bass and treble. There is no interaction between the two tone controls but the circuit requires that three grids be operated at low signal levels with the attendant danger of hum and at high levels the three triodes without feedback are likely to produce more distortion than the output amplifier that follows them.

The circuit of (B) is a simplification using one high-mu triode. Its only advantage is simplification since the distortion produced was quite high as shown in the author's article on distortion in voltage amplifiers[1] and the low- and high-frequency controls suffered from interaction.

To alleviate these difficulties the circuit of (C) in *Fig.* 1 was developed. The inverse feedback reduced the distortion to a reasonable level and the resistor between the sliders of the bass and treble controls minimized the interaction of these controls.

In general the circuit of (C) was a great improvement over anything previously tried but it was considered desirable to separate completely the bass and treble controls so the treble control was moved to the feedback network and the amplifier of (D) resulted. As the slider on the treble control is moved to the left the amount of high-frequency feedback is reduced and therefore the gain of the amplifier is increased at these frequencies. If the slider is moved to the right the amount of high-frequency feedback is increased thus reducing the treble gain of the amplifier.

All the foregoing circuits suffer from one fault. At the midfrequencies there are one or more points in the circuit where a tube is required to operate at 20 db above the desired output level in order to allow for a 20 db boost at high or low frequencies. This excess level adds considerable unnecessary distortion so the next step was to eliminate the necessity for operation at such a high level. The circuit shown at (E) in *Fig.* 1 was designed to produce all compensation but bass cut by variation of the feedback circuit. Because of the practical difficulties of obtaining bass cut by variation of

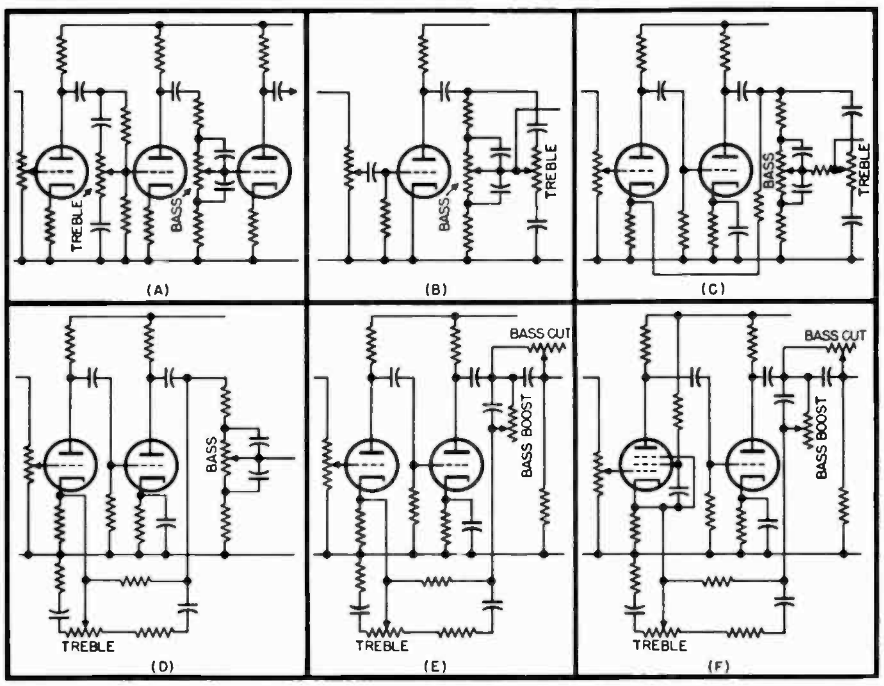


Fig. 1. Six steps in the development of the control unit described by the author. Each of the circuits from (A) to (E) had some faults; all are presumed to be eliminated in (F).

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[1] AUDIO ENGINEERING, Feb., 1953.

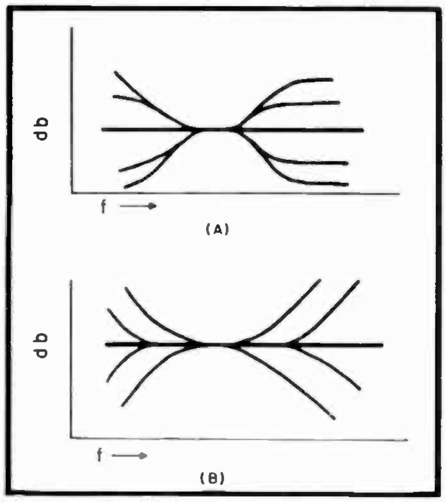

Fig. 2. Using a variable resistor and fixed capacitors in a tone control gives curves like (A). By changing capacitors, leaving the resistor fixed, variable-turnover curves may be achieved, as in (B).

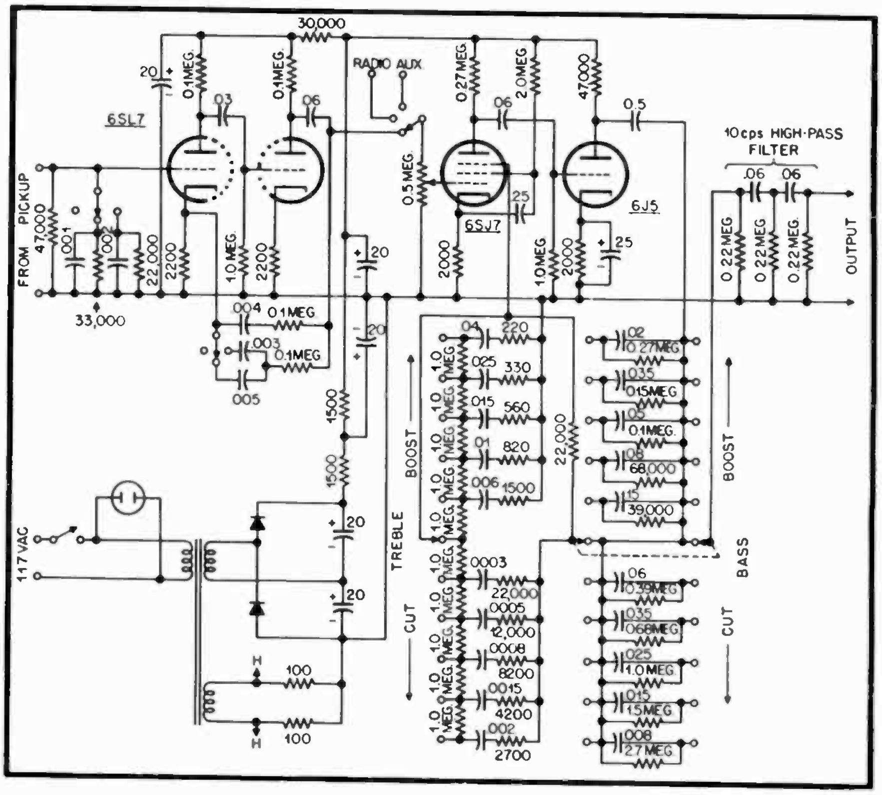

Fig. 4. Over-all schematic of the tone-control amplifier, combined with ordinary preamplifier.

feedback it is accomplished by an RC circuit connected to the output of the amplifier. For the sake of illustration separate bass boost and cut controls are shown but it is of course desirable to combine these on one shaft.

The circuit of (E) was a step forward but it still did not seem to be the optimum design. The requirements for a voltage gain of 10, tone compensation of 20 db, and about 20 db of negative feedback, were not compatible with the operation of the high-mu triodes with reasonable loads. These difficulties were overcome in the circuit of (F). Here a pentode voltage amplifier is used for the first tube and a low-mu triode is used for the second tube.

This circuit fulfills the requirements listed in the paragraph above and has additional advantages. The pentode grid which is fed from the volume control has a very low input capacitance when compared to a triode where the Miller effect has to be considered and the application of the inverse feedback further reduces this input capacitance. As a result, the major portion of the capacitance load on the volume control comes from the wiring and consequently there is no problem of loss of high-frequency response at mid-settings of the volume control. The output impedance of the low-mu triode is quite low and it is further reduced by the inverse feedback so any length of output cable within reason will have no appreciable effect on the high-frequency response. A calculation shows that under the worst condition (maximum treble boost of 20 db) the output impedance of the amplifier is about 600 ohms. This goes down to less than 100 ohms if flat response or treble cut is used. These impedances are as good as or better than would be realized by a cathode follower using the same tube.

The fact that a single potentiometer can not be used for both bass boost and cut may be considered to be a disadvantage but the use of a step-type control also has some advantages in that each step can be designed to give any desired curve up to the maximum of 6 db per octave correction. A variable resistor with one fixed capacitor gives curves which all have the same turnover frequency and initial slope as shown at (A) in *Fig.* 2. By changing the capacitor and leaving the resistor fixed we get curves which have the same slope but different turnover frequencies as shown at (B). Only by varying both *R* and *C* can we obtain the curves with different slopes which is the result usually desired. This makes it desirable to use a step-type control for treble boost and cut although in this case a single potentiometer would serve. By a special arrangement of switch wafers the resulting assemblies are quite compact. The maximum compensation and one intermediate curve are shown in *Fig.* 3.

Figure 4 shows the diagram of the complete unit which was constructed. The phonograph preamplifier section is conventional and will not be described in detail. In addition to the three choices of low-frequency turnover compensation, a sharp-cutoff filter as described by Pickering[2] is incorporated at the input of the preamplifier. The values shown are for a GE cartridge. A 10-cps high-pass filter is incorporated at the output of the amplifier to reduce turntable rumble and to prevent low-frequency transients from unduly disturbing the power amplifier or causing speaker cone excursions of undesirable amplitude. Such transients are prone to occur if a radio tuner which is operated into the system is tuned from station to station with the volume control turned up to a reasonable level. It may be stated as a general rule that there is no reason to feed into the main amplifier any

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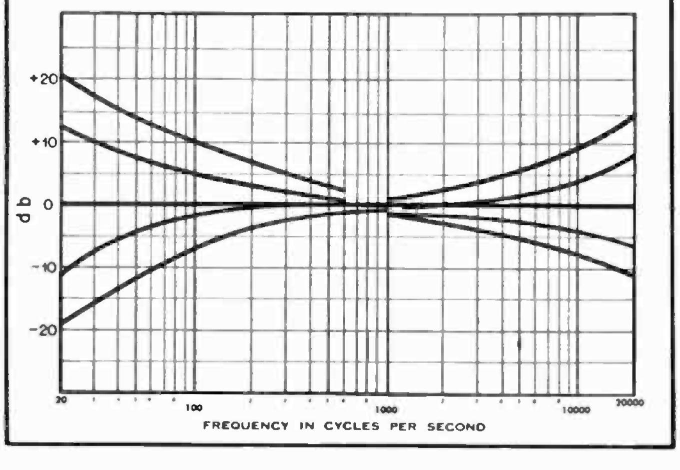

Fig. 3. Limit curves from the author's unit, with one intermediate curve shown to indicate trend of control operation.

[2] N. C. Pickering, "Effect of load impedance on magnetic pickup response," AUDIO ENGINEERING, March, 1953.

Record Improvement with H-F Cut-off Filters

ELLIOTT W. MARKOW*

Simplifying the design of practical low-pass filters for use with sound reproducing systems to eliminate unwanted high-frequency distortion and record scratch —with a few hints as to the construction of a simple and useful unit.

CONSIDERABLE EMPHASIS has been placed on the design of suitable circuits and equalizers to compensate for the original amplitude levels of recorded music. One very important factor which has received comparatively little emphasis is the advantage of being able to cut off at a certain frequency. In fact, inability to do this actually nullifies to a great extent the benefits which derive from equalization. Commercial equalizers and preamplifiers are available which provide quite adequate bass and treble equalization, but few which provide the additional refinement of sharp cut-off.

It is commonly known that one of the most important factors in the pleasing reproduction of all reproduced sound is the lack of distortion in the extreme upper and lower ends of the audio spectrum. This distortion is particularly noticeable and most common in the upper end of the spectrum, and it is usually here that the present day reproduced music contains most distortion and noise. This is particularly true of recordings (and broadcasts of recordings) because of the high noise factors and harmonic distortions which are overemphasized when any attempt at treble boost is made. These factors, unfortunately, combine to make adequate treble boost difficult for the ear to take even though the brilliance and "presence" of the recorded reproduction has been improved and the recording obviously has highs which could profitably be used if distortion and noise factors were removed.

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The solution to this apparent dilemma is obvious and is well known: use a sharp cut-off filter to eliminate all frequencies above a desired cut-off frequency—usually determined by the actual content of the record or program. This is rarely provided in commercial equalizer-preamplifiers designed for home use but is very necessary to get the utmost from recordings. An elaborate well equalized system that lacks this necessary feature is still incomplete.

It is quite important to emphasize the words "sharp cut-off." If sharp cut-off is not provided the circuit provides essentially only roll-off, which does not approach the effectiveness of sharp cut-off. Most so called "scratch filters" provide only a 6 db per octave roll off and are not recommended for a high-quality system.

It is the purpose of this article to emphasize the desirability of adding a sharp cut-off filter to an otherwise well equalized system and discuss the design factors involved so that those interested can add a suitable filter to their present equipment or build it into new equip-

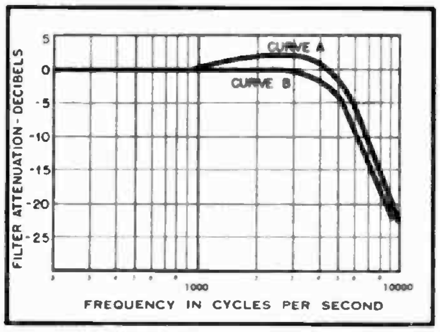

Fig. 1. Cut-off characteristic of a single-section filter designed for cut-off at 5000 cps. (A) represents the response of the filter using values of R and L given by Fig. 3; (B) response for values of R reduced to 2/3 of the Fig. 3 values or L increased by a factor of 1.5.

ment. None of the mathematics or theory involved will be discussed; just presentation of design curves and information. Filters of this type do not come as cheaply or as easily as simple resistance-capacitance networks. The necessary parts will cost somewhere around $15, but this seems small, in the writer's opinion, compared to the results obtained.

Proper cut-off makes many recordings of normally poor quality sound surprisingly good and quite enjoyable. It will not remove distortion occurring in the transmission band but it will remove a

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LOW-DISTORTION AMPLIFIER

(*from preceding page*)

low-frequency signal that the speaker cannot handle.

In many cases the low-frequency coupling afforded by supplying the preamplifier plate power from the output amplifier is sufficient to make the whole system just marginally stable, or even unstable, so the power supply for this unit is self contained. This independent power supply also makes the whole unit much more versatile. The power transformer is a 40- or 50-ma shielded unit designed for use in TV boosters. This transformer is operated into a voltage-doubler rectifier and an RC filter.

The photos show the appearance of the finished unit, which is built in a 7 × 12 × 3 chassis. A sub-chassis of 3/32 aluminum is fastened inside the chassis to hold the tube sockets and power supply components.

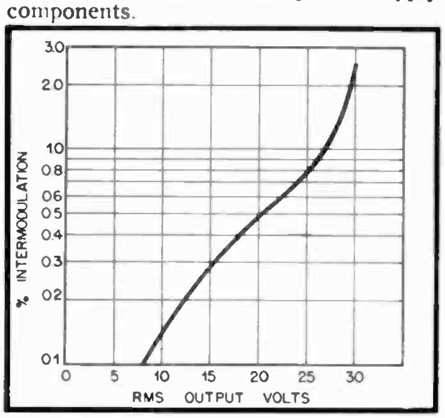

Fig. 5. IM distortion curve for outputs in excess of 8 volts rms. Distortion below this value is almost unmeasurable.

The IM distortion *vs.* volts output is shown in *Fig.* 5. This curve was taken with 5 feet of shielded cable connected to the output. Although a very long cable may be used without affecting the high-frequency response the plate current required from the output tube to produce a given high-frequency voltage is increased and with the increase in current comes an increase in distortion. Since the power of high frequencies in the usual program material is quite low the greatest difficulty in the use of a very long output cable will likely be hum voltages.

This article is intended to be a general discussion of the principles behind the design of a class of circuits and not a step by step constructional article so no dimensional drawings are included. It is thought that the amplifier pictured will fill the usual requirements found in high-fidelity home music systems and that the general principles outlined will be useful in a wider field.

lot of noise without removing a noticeable amount of music. Hi-fi extremists will argue that this is not high fidelity, but it is far better that the sounds please the esthetic sense than that a few extra cycles, with attendant noise and distortion, be reproduced to satisfy a theoretical curve. The pros and cons of this subject have been well covered in numerous other articles and need not be elaborated upon here. Let us simply say that personal preference dictates your course of action. In every case your non-technically minded listener—those listening just for the sake of the music—will invariably prefer a reasonable cut-off on imperfect sources.

Here is an important point: AN EQUALIZED SYSTEM WHICH CUTS OFF AT 5000 CPS AND WHICH DOES NOT OTHERWISE INTRODUCE DISTORTION WILL SOUND OF SURPRISINGLY HIGH QUALITY AND WILL BE ESTHETICALLY VERY SATISFYING. A 7000-cps system will sound somewhat better; a full 10,000-cps system will sound uncannily realistic and will approximate the "presence" of a live performance. Anything above this comes dearly and, with normally available music sources, very infrequently.

Filter Design

The amount of cut-off provided by a single-section filter described here is shown in *Fig.* 1. The same general shape and amount of cut-off per octave prevails regardless of design cut-off frequency. Experience seems to indicate that cut-off frequencies of 5000, 7000, and 9000 cps are adequate for any home system, and for practically all available recordings. It is unfortunately much easier to design a filter on paper than to execute it in terms of actual equipment. A filter is not a complicated device, but in order for it to work properly it is quite fussy as to operating conditions—in particular the input and output impedances seen by the filter section and the relationship between the inductance in the filter and these impedances. The input and output resistances "seen" by the filter section should be equal; furthermore there is a rather

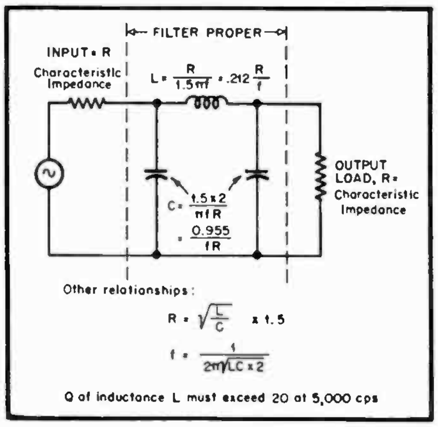

Fig. 2. Single-section low-pass filter which provides a slope of 22.6 db per octave attenuation beyond cut-off.

strict relationship between these values and the value of inductance required. The input and output resistances are commonly known as the characteristic impedance of the filter. It is around these values and the desired values of cut-off frequency that the filter components are determined. Fortunately the components need not be of close tolerance.

Without going into a lot of mathematics, one very practical circuit from the standpoint of transient response, sharpness of cut-off, and reasonableness of component values is that shown in *Fig.* 2. With correct selection of components this will provide about 22 db of attentuation per octave with only 3 db of attenuation at the design cut-off frequency. The precise sharpness of cut-off is, to a slight degree, a function of the "Q" of the inductance *L* but for values of Q above 20 there is very little increase in cut-off sharpness. More complicated circuits will give sharper cut-off but require additional components and usually have undesirable transient characteristics. The design formulas for this filter are also given in *Fig.* 2. *Figures* 3 and 4 show the relationships between characteristic impedance *R*, inductance *L*, and capacitance *C*.

This filter can be put in either the plate circuit or the cathode circuit of any medium- or low-gain triode amplifier or cathode follower and still have reasonable values of inductance and capacitance if certain factors are considered. Because of the loading effects of the filter it is almost impractical to insert the filter in the plate circuit unless the load resistance of the tubes is very low, so if at all possible it is much more preferable to operate the filter from a cathode-follower stage where loading effects can be practically eliminated. Where this filter can logically be inserted in an existing system is a matter of available gain, tube line-up, and physical layout. It is best designed into a new preamplifier and fed from a cathode follower, but it can be added to most existing amplifiers particularly if the preamplifier and power amplifier are connected through a cable. There are advantages in operating at as low an impedance level as practical and this usually means operating from a cathode follower output. Hum pick-up and practical values of capacitance are the major reasons for using a low impedance. With a toroid coil the problem of hum pickup is avoided. It is quite important that the filter characteristic impedance present a negligible load on the driving source, and as a rule of thumb the characteristic impedance should be a minimum of about 10,000 ohms.

There are several approaches to the selection of component values. Examination of *Fig.* 3 will show the relationship between inductance *L* and characteristic impedance *R*. In the usual case the filter is designed around a commercially available value of *L*, chosen to operate from some arbitrarily chosen characteristic impedance. Values of input and output resistance and capacitance are switched as necessary to change the cut-off frequency and the characteristic impedance. If we restrict ourselves to reasonable values of characteristic impedance and tuning capacitance, as can be determined from *Figs.* 3 and 4, it will be seen that the useful range of inductance values at 5000-cps cut-off varies between about 200 millihenries and 3 henries, all commercially available values. Characteristic impedance varies

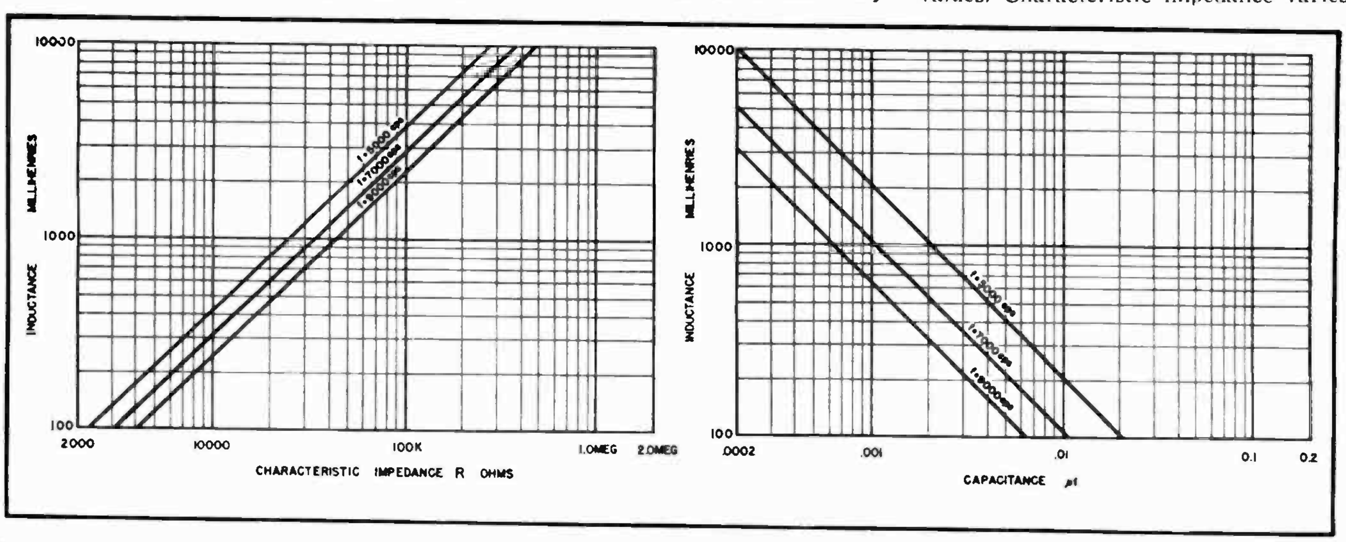

Fig. 3. (left). Chart showing relation between inductance L and characteristic impedance R. Fig. 4 (right). Chart showing values of capacitance C for indicated values of inductance L in Fig. 3.

between about 10,000 and 100,000 ohms.

Suitable inductors are available from several manufacturers in a large range of values. One value of inductance L chosen to operate at the desired impedance level can serve for all three frequencies by merely switching resistors and capacitors as illustrated in *Fig.* 5. The inductor should have a Q of at least 20 at 5000 cps. It will cost somewhere between $4 to $10 depending upon value, type of winding, and shielding. Toroids are recommended, especially for high-impedance circuits.

Practical Filter Example

A typical circuit is illustrated in *Fig.* 5. Values of capacitor C may be taken directly from *Fig.* 4. Values of input resistance are not critical but selection should be made on the low side of the design value rather than exceed the design value. As an example of how this filter was calculated, suppose a filter is required to operate in the cathode circuit of a triode, as in *Fig.* 5. The input impedance of the cathode circuit is usually given approximately by the simple equation $Z = 1/G_m$; however, it should be remembered that this condition is true for an unbypassed cathode resistor and any capacitance in parallel with and approximating the value of the cathode resistor will change this relation. The characteristic impedance of the filter should therefore be higher than the cathode resistor to avoid by-passing. A good working value is to choose the characteristic impedance of about 20,000 ohms, as has been done in the example. If this value of characteristic impedance is chosen the input impedance of the cathode follower remains approximately $1/G_m$ (about 500 ohms for most triodes) and no consideration need be given to either the actual cathode resistor or input impedance. High values of cathode resistance are usually used to keep the gain close to unity.

Having arbitrarily selected an approximate value of 20,000 ohms for the characteristic impedance, refer to *Fig.* 3 and select the nearest commercially available value of inductance which corresponds to a characteristic impedance of approximately 20,000 ohms, which in this case is 750 millihenries for the 5000-cps cut-off frequency. Selection of the inductance should be done on the basis of the lowest desired frequency of cut-off so that all succeeding impedances will be larger than that at 5000 cps. Values of characteristic impedance for frequencies of 7000 and 9000 cps are read directly from *Fig.* 3 and are seen to be 24,000 ohms and 33,000 ohms respectively. Similarly values of capacitance required for this value of inductance and for the chosen cut-off frequencies are read directly from *Fig.* 4, and are found to be .0027 µf, .0014 µf, and 850 µµf.

Let us examine further the selection of coupling capacitor C_c. This capacitor is used in a moderately low impedance circuit, therefore its capacitance must be large if the bass response is to be adequate. For a 3-db loss at 50 cps 20,000-ohm circuit, C_c must be 0.17 µf. By the selection of the correct value for C_c, bass response correction to correspond to high-frequency cut-off can be accomplished. It is well known that a system sounds best when the upper and lower frequency limits are both restricted so that the products of the limits equals approximately 400,000. As an example, if the treble response extends to 10,000 cps the bass response should extend down to 40 cps for the system to sound right. If bass is restricted the system sounds "tinny." If treble cuts off at 5,000 cps and bass continues flat to 40 cps the system sounds "boomy." The rule is that the product of bass and treble cut-off frequencies should equal 400,000 approximately. Therefore, for a system which cuts off at 5000 cps the bass should also cut off at 80 cycles.

The way to accomplish this partially is to select C_c so as to be just large enough to pass 40 cps (with only a 3-db loss) with the values of characteristic impedance chosen for 9000-cps cut-off. Using the example of *Fig.* 5, if C_c is chosen so that its reactance at 40 cps equals 33,000 ohms the attenuation will be 3 db. This same value of C_c will however provide an attenuation of 3 db at 100 cps (reactance equals 18,000 ohms at 100 cps) if the characteristic impedance is 18,000 ohms as for the 5000-cps filter, giving a product of upper and lower limits of 500,000 which will sound quite properly balanced. Similarly, balance is maintained for the case of the 7000-cps filter.

If plate coupling is selected, the selection of components is made in the same way. In this case the characteristic impedance of the filter should be high compared to the load resistor of the tube. A characteristic impedance as high as 75,000 ohms at 5000 cps still gives practical values for inductance and capacitance; however at these high impedance levels the capacitance of the circuit and inductance itself becomes an appreciable percentage of the tuning capacitor and must be considered, particularly for the 9000-cps filter.

Because of the filter, there is an insertion loss which becomes noticeable as a difference in volume level as the filter is switched out of the circuit. To compensate for this difference in level the use of resistors R_o and R_4 are suggested. The insertion loss will be about 6 db so that ratio of R_o and R_4 should be about 1 to 1. If R_4 is chosen to be about 33,000 ohms, R_o will also be 33,000 ohms. Since this is the value for R_1, the switch is wired so as to connect to R_1 for positions a and b. This amount of loss may need some adjustment depending upon your personal preference regarding relative loudnesses with and without filters.

One other point of importance: with the input and output impedances given by *Fig.* 3, a treble boost of about 2 db is introduced to give slightly flatter overall response up to the cut-off frequency. It is fairly important that some form of roll-off equalizer precede the filter to reduce this slight peak. If no equalizer is used the values of inductance as specified by *Fig.* 3 can be increased by a factor of 1.5 (or the characteristic impedance reduced) to give a somewhat smoother roll-off at frequencies before cut-off. In either case the cut-off is of the order of 22 db per octave for a set of values correctly selected as described, as is illustrated in *Fig.* 1.

No other single factor will improve a high-quality, well-equalized system as much as a well-designed filter. It is difficult to appreciate just how good some of your older records can sound (and some radio broadcasts improved) when all the noise and distortion above a reasonable cut-off frequency are eliminated. Install one in your equipment and listen for yourself.

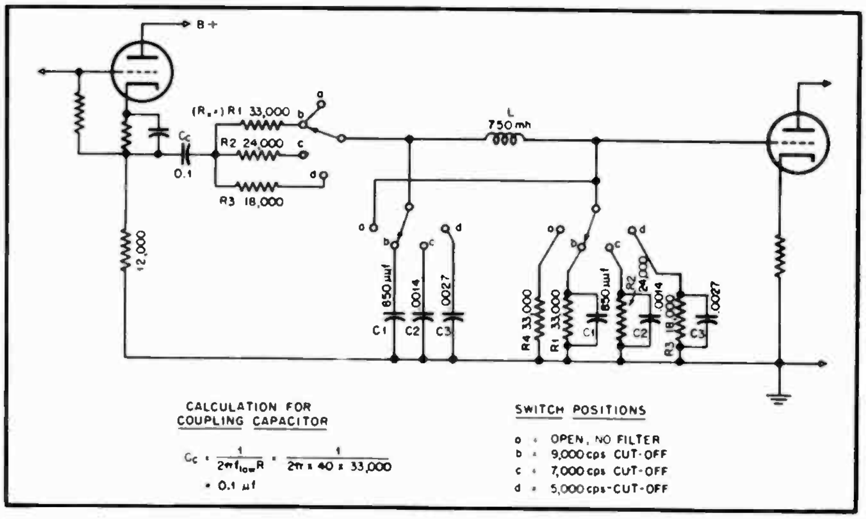

Fig. 5. Schematic of suggested filter to be connected between a cathode follower and the succeeding tube to provide cut-offs at 5000, 7000, and 9000 cps, together with a flat position.

Cathode-Follower Audio Control System

G. B. HOUCK*

The connection of existing equipment to a typical high-quality amplifier and speaker system often poses difficulties in switching. The author outlines a method which makes it possible for anyone to operate the "system" without excessive indoctrination.

In the construction of many home audio systems, basic design fundamentals are often overlooked. Not only are some of these installations very expensive, but frequently difficult to operate and inflexible in use. Several designs have been illustrated recently which feature a confusing profusion of controls, located either where they are not easily seen, or where their manipulation is awkward.

In the past, when audio systems were designed for hobbyists and technicians, there was little objection to laboratory-type installations.[1] When non-technically trained persons such as professional musicians came in contact with audio they usually found it necessary to learn enough electronic and acoustic fundamentals to control the medium effectively.

Today the audio field has expanded, largely because of the availability of high-quality, low-cost components, and an increasing awareness among consumers that "hi-fi" systems can be much more satisfying than ready-made units. Apparently, however, the demand for such ready-made units has not diminished to the point where manufacturers are unduly concerned. On the contrary, the present market for consoles and combinations suggests possible advantages over those of custom-built systems.

Of course there are many factors influencing sales of ready-made units and audio components. Perhaps the features dwelt upon in this discussion are not those immediately apparent to the prospective customer. Nevertheless, as factors of good system design, they cannot be disregarded.

The designer of audio systems must realize that simplicity is not only a virtue of sound design, but a public demand. This fact has been demonstrated repeatedly in the sales of competitive merchandise. For example, a TV receiver was put on the market a while back which featured "one knob control." Regardless of its engineering merits or deficiencies it enjoyed a good sales record, primarily on account of its control simplicity. Whatever the psychological implications may be, the fact remains that simplicity of operation is a most important consideration in audio system design. This may be qualified by stating that simplicity must be achieved provided it is not at the expense of quality. The qualification immediately reveals the weak point of most ready-made units, particularly in the case of tone-compensation control (a design drawback, not necessarily an engineering one).

One other point regarding the relative merits of ready-made and custom built systems should be brought out at this point in the discussion. Too many custom designers have the notion that the only obstacle in the way of increased sales and installations is a mere problem of packaging improvement. Perhaps this idea is prevalent for the reason that the expansion of the home audio business has coincided with an increasing proficiency in the techniques of installation both in furniture and in structures. Indeed, the ability of some designers in rendering electronic gear elegantly inconspicuous is legerdemain. Unfortunately, many installations are perpetrated by persons not primarily concerned with audio. If the decorator and cabinetmaker have no understanding of the operating requirements of sound systems, or if the professional designer considers them less important than novel cabinet design, it can be expected that the results will be anything but satisfactory. Controls will be located close to the floor,[2] labels will be non-existent, record changers will be inaccessable, etc. There is always room for improvement in packaging design when it meets the requirements of human engineering. The term in its truest sense implies an approach in which basic design fundamentals are applied in such a manner as to cause response and control by the instrumental and human mechanisms to be smoothly coordinated. The designer should understand that achievement of this goal is possible only when all the factors are carefully considered. The best designs will feature careful choice of control types, clear control identification, proper arrangement, and most important, convenient operating location.

Design Considerations

When the actual construction and installation of an audio system is contemplated the usual thoughts are directed to a design which presupposes starting from scratch. This may be the wrong approach. It effectively tags the owner's present equipment for disuse or even the junk-yard. It is also somewhat misleading in that it attempts to convince the owner that there is nothing worth retaining in his present equipment which may include a TV receiver, FM-AM radio, phonograph etc., of fairly good quality. These units may also represent a considerable investment. Furthermore, the custom-built system, when the costs of cabinetry are considered, is often prohibitively expensive, and usually cumbersome or simply immovable.[3] As such it may become a source of frustration to some homemakers, forever obsessed with the urge for periodic rearrangement of furnishings.

The approach to this problem presented here in its broadest scope is one which offers the builder the double advantage of retaining not only the operating features of his present equipment, but the actual equipment as well. At the same time, an increase in audio quality is made possible. These three features represented the initial goal when development of the cathode-follower control system was first undertaken. Other advantages, made apparent with the final perfection of the technique, will be discussed and evaluated later.

With the decision made to construct a high-fidelity audio system utilizing existing receivers, the task became to investigate the quality of the receiver units. Measurements of a typical AM table model radio indicated an audio frequency response of 20 to 7000 cps excluding the output transformer. Similar measurements with TV receivers

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[1] M. P. Johnson, "Multipurpose audio amplifier," Audio Engineering, August 1947.

[2] Jeff Markell, "Cabinet design for Hi-Fi systems," Fig. 4, Audio Engineering, May 1952.

[3] William C. Shrader, "Audio in the home," Audio Engineering, May 1952.

and FM tuners led to the assumption that except for their audio output sections, most of these receivers could be classed as "hi-fi." The next step, then, was to devise a method of bypassing the audio output sections, apparently the principal sources of poor frequency response and distortion. This would be accomplished with an absolute minimum of circuit changes in the receivers. It was also obvious that some means should be available for using one high-quality audio power amplifier and speaker system for all of the units. The final step involved design of an adapter device for linking the receiver front-ends with the amplifier. With the completion of these design steps, several unique advantages became immediately apparent. There would be no modification of existing unit controls nor any change in operating precedures. No rebuilding of receivers nor extensive circuit changes would be necessary.[4] The various units could be located wherever desired. Total cost would be no more than necessary to purchase or construct the power amplifier, speaker and several adapters.

Four steps are involved in the realization of such a hi-fi system. The engineering and craftsmanship required are a minimum. The first step involves the addition of a few components to the power amplifier, as shown schematically at (A) in *Fig.* 1 and pictorially in *Fig.* 2. The second step is the "adaption" of the units. The third step is the construction of the simple adapter devices. The fourth step is the installation of the complete system.

Circuit

Before describing these four steps in detail, it is desirable to consider the circuitry involved, including purely electronic functions. The essence of the system is the connecting link between the receiver unit and the hi-fi amplifier. Basically, it is a cathode-follower device which effectively shorts out the regular output transformer and provides special low-impedance output from the cathode.[5, 6] The pin connections shown are for output tubes similar to those shown at (B) in *Fig.* 1. Other tube types may be used with appropriate connections. In the power amplifier the d.c. component of the signal is used to energize a 5000-ohm d.c. relay which controls primary power for the amplifier. The audio signal passes through a blocking capacitor and a potentiometer adjusted for the proper operating level. The usual power switch remains in parallel with the relay contacts, providing manual power control in the event high-impedance sources are connected to the amplifier. The input terminal jack is wired to accommodate either high or low-impedance sources. *Figure* 2 shows a Williamson type amplifier with relay installed.

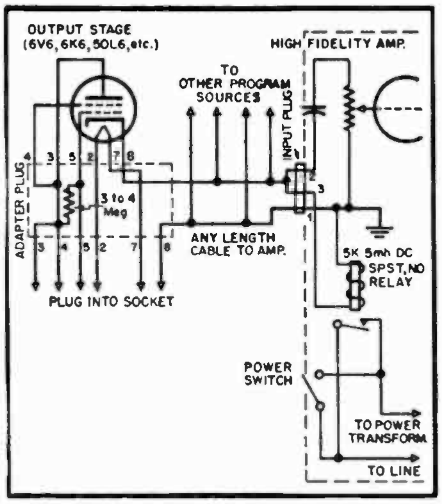


Fig. 1. Circuit schematic for the cathode-follower control system.

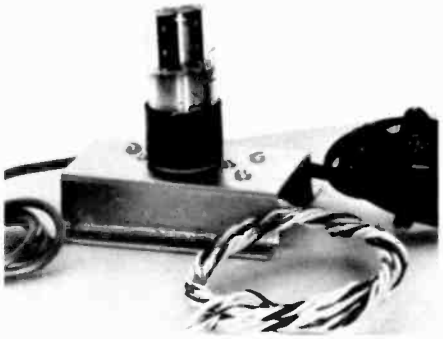


Fig. 3. Adapter chassis used with electronic organ, showing simple "add-on" unit which may be employed with tuner or any other sound source.

As outlined a few paragraphs above, no modification of the receiver units is necessary in most installations. There are, however, a few cases where a simple addition to a receiver may be needed. If the tuner has no audio output section or terminates, for example, in a detector or discriminator output, a unit similar to the one shown in *Fig.* 3 may be constructed easily. The 6SN7 assembly illustrated is one particular type which was added to an electronic organ for an echo amplifier and speaker arrangement. Miniature tube types such as the 12AU7, 12AT7, etc., result in a space saving. *Figure* 4 shows the schematic of such a unit. The assembly is simply mounted by fastening it to the chassis or securing it to the cabinet. The only caution to be observed is the need for an adequately filtered plate supply, as any hum present will be fed into the audio output. It is also recommended that an isolating transformer be installed in a.c. transformer-less sets to eliminate possible shock hazard.

Construction of the adapter plugs is not difficult. The socket should be similar to the type illustrated, in *Fig.* 5. The plugs, if unavailable from parts distributers, can be fabricated from old tube bases (containing a sufficient number of pins). The resistor and pigtails are soldered to the socket as shown in the photo and drawn through the plug pins along with the output leads which are passed through a small hole drilled in the side of the plug.

Installation of the complete system is simplicity itself. Since the line impedances are low, the interconnection may be made using ordinary zip-cord or bell wire. For under-the-carpet wiring, TV antenna lead-in ribbon is recommended, since it is flat and causes no lumps or bulges.

Phonographs

So far, nothing has been said about the use of a phonograph with this system. There are few hi-fi record playing devices these days which do not require preamplifiers and some means of equalization. In the case where the phonograph player has its own preamp and audio output, any of the previously mentioned adaption techniques can be similarly employed. If the builder has nothing more to start with than the basic player mechanism and pickup, he is advised to construct a preamp-equalizer, as shown in *Figs.* 6 and 7. The three requirements for such a device are proper load for the pickup, sufficient gain, and cathode-follower output

Fig. 2. Williamson-type amplifier with cathode-follower control system installed.

[4] Ulric J. Childs, "Why not use your present tuner?" Audio Engineering, April 1952.

[5] G. B. Houck, "Gain chart for cathode followers," *Tele-Tech*, August 1947.

[6] Audio Design Notes—"The cathode follower," Audio Engineering, June 1947.

impedance. The design illustrated provides a minimum of control. If the builder desires a more elaborate equalizer, several excellent designs have been presented in previous issues.[7]

Location of Units

There are practically no restrictions on the location of the units of the complete audio system. Because of the low-impedances and the need for only two conductors, the units may be placed as far apart as desired, without undesirable effects such as high-frequency attenuation or hum-pickup. In general, the TV receiver should be located where it can be viewed easily. In the case of the speaker, a corner location is usually preferred because loading is improved. There is certainly no reason why a TV receiver cannot be located directly above or even near the center of the sound source.[8] One serious objection to the

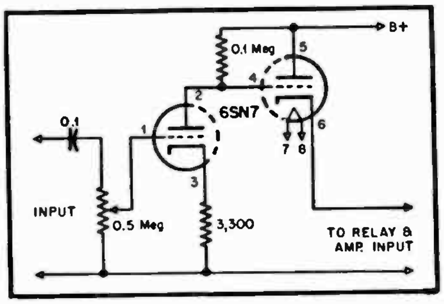


Fig. 4. Schematic of adapter chassis shown in Fig. 3.

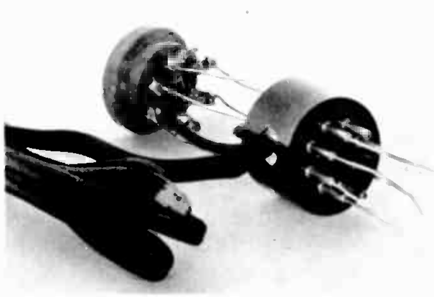
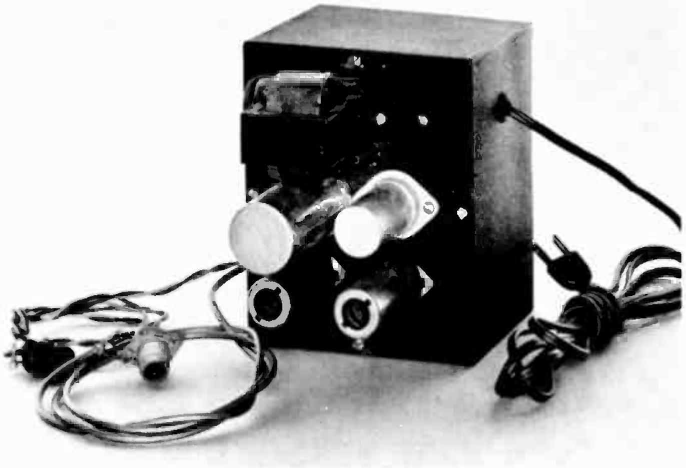


Fig. 5. Assembly suggested for adapter plug used with typical output tubes in AM, FM, or TV receivers.

location of any units close to the speaker cabinet is that it results in poor control. If the operator is forced to stand less than a foot or so from the speaker when adjusting the volume control, it is quite likely that the level will be unsatisfactory at other distances. As a matter of fact, it is most desirable to have the controls as near the listening position as possible. This, of course, is one of the principal features of the cathode-follower control system. It should not be difficult to place the FM or AM radio next to a comfortable chair, or the phonograph near the record storage space. With the power amplifier and its power supply located in a closet, in the cellar or attic, heat

Fig. 6. Compact self-powered pre-amplifier-equalizer designed to feed a phonograph signal to the power amplifier in the same manner as described for other sound sources.



dissipation problems are solved, and there is little chance of hum produced by power transformer interference.

Performance

In discussing the performance of the system it is important to emphasize the functional aspects as well as electronic. Since the design calls for bypassing those portions of the receivers which represent the chief source of poor quality, it is obvious that better performance is achievable. It should be reasonable to expect the audio quality to be limited only by the capabilities of the power amplifier, provided there are no losses in the connecting link. If an amplifier comparable in quality to the Williamson is employed, it seems safe to assert that high-fidelity will be obtained. As for the connecting link, its impedance is generally less than 500 ohms. If the average level is approximately 1 volt and the hum and noise level as much as 100 microvolts, or 80 db down, it is obvious that no loss of quality is observed. Although the output lines of every unit are wired in parallel, no harm is done if more than one unit is turned on at a time. In this case, the bias on the cathode-follower may be upset and the resulting sound is somewhat disagreeable, if not confusing.

The distinct advantages of this system are revealed when functional performance is evaluated. The control features of the separate receivers are retained, control of hi-fi audio selection is completely automatic, and there is very little chance of control confusion. Since most persons are familiar with the control arrangement of conventional receivers, the builder need not consider preparing an instruction manual for the system. Whereas there are some audio systems which can be operated only by their designers, the cathode-follower audio control system installed in the author's home and in several others is used with equal facility by visitors and family, including his five-year-old son.

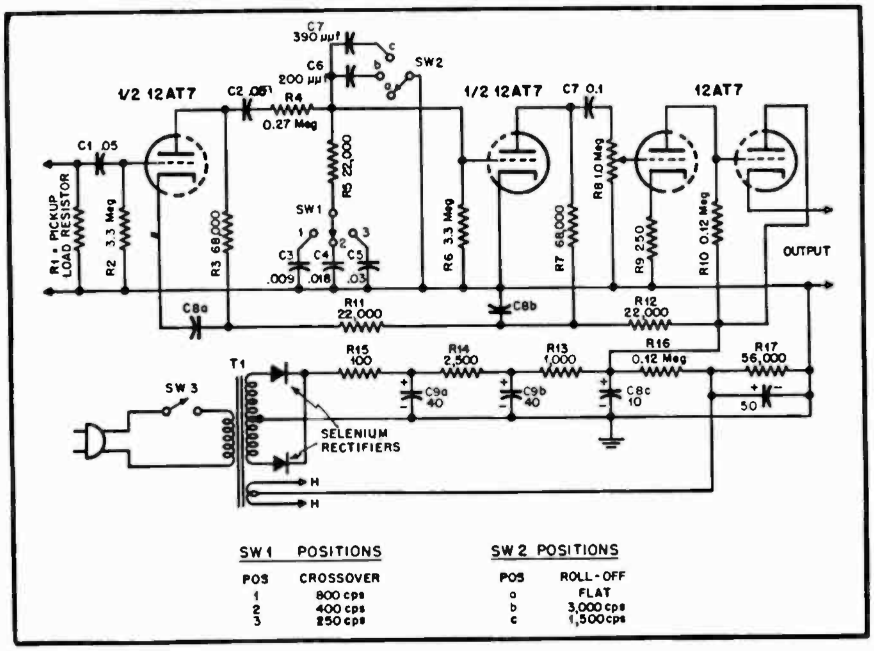


Fig. 7. Circuit of preamplifier pictured in Fig. 6.

[7] St. George and Drisko, "Versatile phonograph preamplifier," Audio Engineering, March 1949.
[8] C. G. McProud, "A new corner speaker design," Audio Engineering, Jan. 1949.

Simplified Equalizer Design

GEORGE A. DOUGLAS*

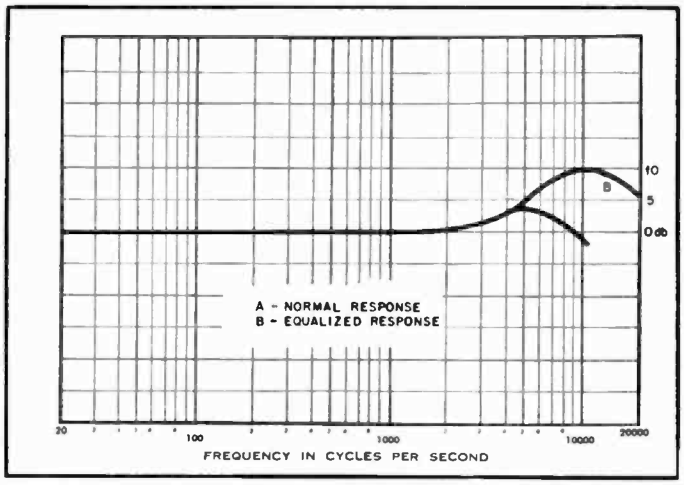
Charts and tables to reduce complication—and construction hints to ease building.

In applications requiring exact modifications of the audio spectrum, constant-resistance equalizers are capable of doing the job with maximum efficiency and convenience. The correction of cutter peaks and tape droop at the high end, the incorporation of pre-emphasis, and numerous other alterations of response can readily be accomplished with the design data that follows, and which involves only simple arithmetic to calculate values of components; or, with the use of a reactance chart, a few steps of multiplication.

Figure 1 shows the four configurations with their transmission characteristics used to cope with the equalizer problems commonly met in audio work. The circuits of (a) and (b) are shelf suppressors, or conversely, low and high boost; (c) and (d) are peaking equalizers. As it is inadvisable to use more than 20 db of attenuation in a single equalizer, it will be seen that the shelf suppressors serve where a gradual curve is required not exceeding 6 db per octave. Networks may be cascaded to obtain a steeper characteristic, but this requirement is usually met by applying (c) or (d), in which case it is possible to control the slope.

The first step in the design of an equalizer is to draw on logarithmic paper the actual and desired response of

Fig. 2. Method of adding pre-emphasis to recording characteristic to eliminate a cutter peak, for example. Curve A represents normal response, while curve B represents desired response. The required equalization is represented by the difference between the two curves.



* *51 E. 42nd St., New York 17, N. Y.*

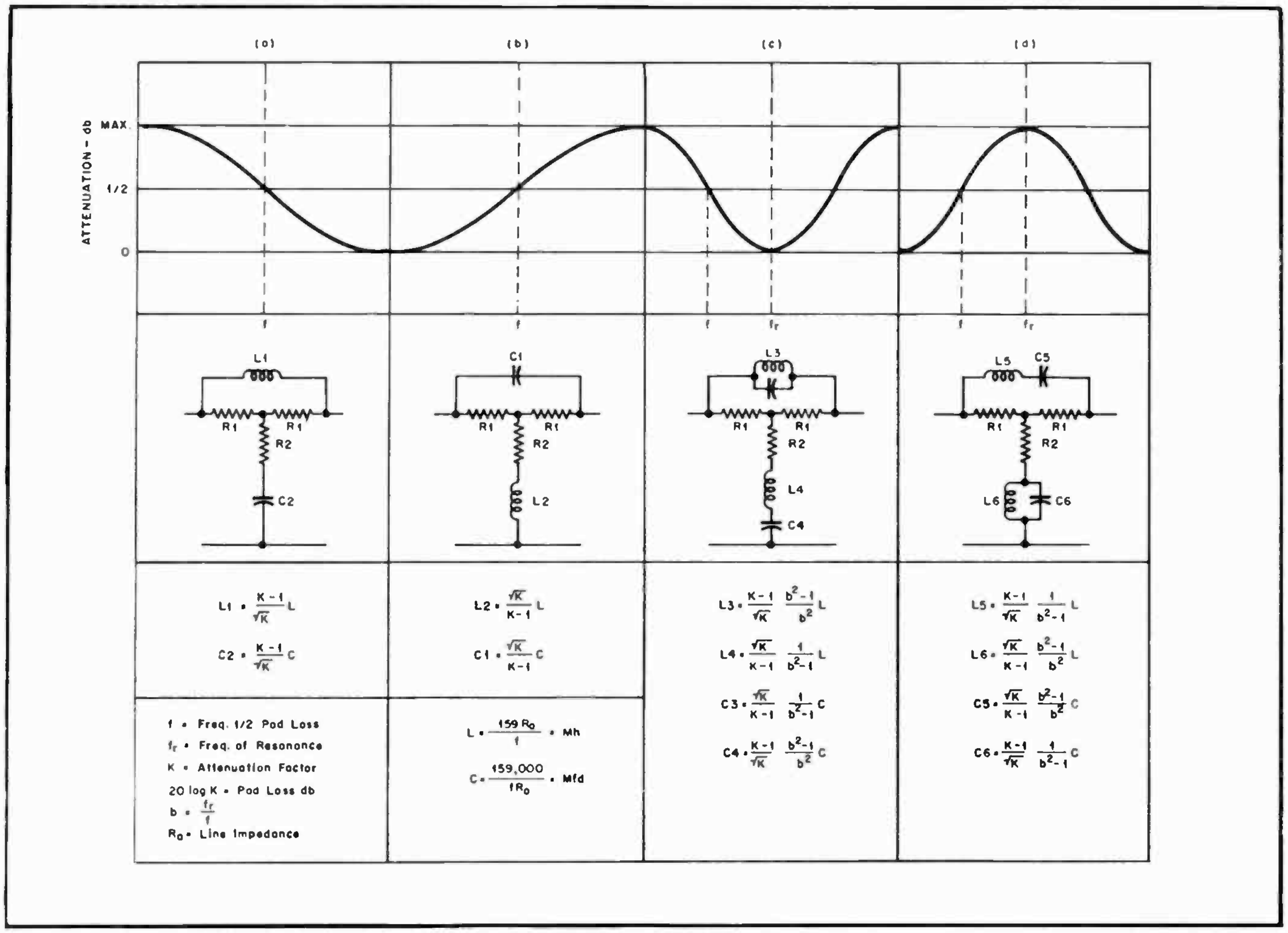


Fig. 1. Chart showing the configurations of four different types of equalizers, together with formulas for determining the reactive elements.

the equipment requiring modification or correction. This will furnish information to construct the transmission characteristic of the equalizer. On this curve note the design frequency on the transmission characteristic at the line of half the total attenuation. With the design frequency f and the impedance of the line in which the equalizer is to work, calculate L and C, or take their values directly from a reactance chart. L_1 to L_6 and C_1 to C_6 may now be ascertained by substituting in the equations figures taken from the accompanying tables.

For the shelf suppressors only Table 1 is used. The values for $\frac{K-1}{\sqrt{K}}$ and $\frac{\sqrt{K}}{K-1}$, and for R_1 and R_2 for an impedance of 1 ohm, are located opposite the degree of attenuation desired. For other values of R_1 and R_2, multiply by selected R_0. The data in Table 2 determines the slope of equalizers (c) and (d). The ratio in column 1 corresponds to f_r/f (the resonant frequency divided by the frequency of one half the attenuation). Opposite this are found the solutions of $\frac{b^2-1}{b^2}$ and $\frac{1}{b^2-1}$

Example:

Curve (A) of *Fig. 2* depicts a typical cutter response above 1000 cps. It is desired to design a 600-ohm equalizer for a recording characteristic with 10 db of pre-emphasis at 10,000 cps. Ordinarily the low-frequency shelf suppressor could be used, but in this case the cutter peak would be undesirably increased. It is apparent that the rising slope of the peak can be continued to the 10,000-cps, 10-db point; therefore the obvious solution is to design the equalizer with the configuration of (d) with a resonant frequency of 10,000 cps, and an attenuation[1] of 10 db. From the curve, the point of half the attenuation, or 5 db, is found at 7000 cps, the design frequency. L and C can now be ascertained:

$$L = \frac{159 \times 600}{7000} = 13.62 \text{ Mh}$$

$$C = \frac{159{,}000}{7000 \times 600} = .038 \ \mu\text{f}$$

Now, since

$$L_5 = L\frac{K-1}{\sqrt{K}}\;\frac{1}{b^2-1}$$

$$L_6 = L\frac{\sqrt{K}}{K-1}\;\frac{b^2-1}{b^2}$$

$$C_5 = C\frac{\sqrt{K}}{K-1}\;\frac{b^2-1}{b^2}$$

$$C_6 = C\frac{K-1}{\sqrt{K}}\;\frac{1}{b^2-1}$$

From the tables:

$$\frac{1}{\sqrt{K}} = 1.213 \qquad \frac{f_r}{f} = \frac{10}{7}$$

$$\frac{K-1}{\sqrt{K}} = .824 \qquad \frac{b^2-1}{b^2} = .512$$

$$R_1 = .520 \qquad \frac{1}{b^2-1} = .952$$

$$R_2 = .704$$

Substituting, we now have:

$L_5 = 13.62 \times 1.213 \times .952 = 15.7$ Mh
$L_6 = 13.62 \times .824 \times .512 = 5.8$ Mh
$C_5 = .038 \times .824 \times .512 = 0.16\ \mu$f
$C_6 = .038 \times 1.213 \times .952 = .043\ \mu$f
$R_1 = .520 \times 600 = 312$ ohms
$R_2 = .704 \times 600 = 423$ ohms

For many applications in the upper frequency range, r.f. chokes can be utilized, providing the d.c. resistance is not excessive. The photographs, *Figs.* 4 and 5, show a completed equalizer with 5-mh. coils of about 15 ohms. To provide proper inductance, select capacitor(s) of correct value for C in *Fig.* 3. Set the oscillator to the desired resonant frequency, with more inductance in the circuit than is needed, and remove turns until maximum attenuation occurs.

Coil resistance can be compensated for to some extent by including it in the pad. Thus, when an inductance appears in the shunt arm, its d.c. resistance should be deducted from R_2, and a resistor selected which has an actual

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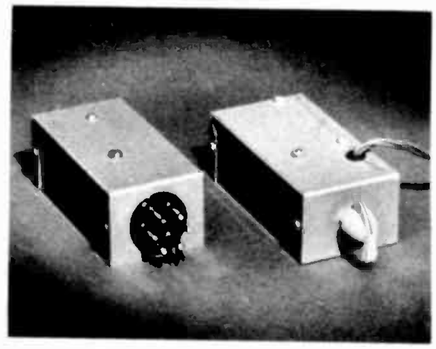

Fig. 4. Two equalizers constructed by the author—left, a plug-in unit employing a fixed amount of equalization, and right, a unit providing step-switch control of attenuation.

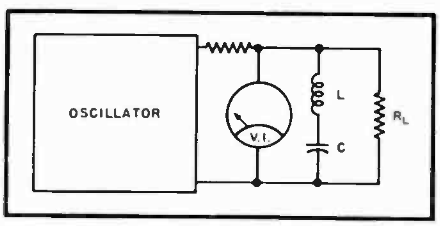

Fig. 3. Circuit arrangement for determining resonant frequency of LC combination.

TABLE 2
EQUALIZERS

$\frac{f_r}{f}$	$\frac{b^2}{b^2-1}$	$\frac{1}{b^2-1}$
10/9	.187	4.348
10/8	.359	1.785
10/7	.512	.952
10/6	.637	.568
10/5	.750	.333
10/4	.840	.190
10/3	.910	.099
10/2	.960	.042

TABLE 1
EQUALIZERS

db	$\frac{K-1}{\sqrt{K}}$	$\frac{\sqrt{K}}{K-1}$	R_1	R_2
1	.113	8.833	.057	8.68
2	.232	4.307	.114	4.32
3	.345	2.902	.171	2.84
4	.468	2.135	.226	2.20
5	.586	1.705	.280	1.646
6	.709	1.410	.332	1.34
7	.826	1.193	.382	1.118
8	.949	1.053	.430	.946
9	1.082	.923	.476	.812
10	1.213	.824	.520	.704
11	1.356	.737	.560	.602
12	1.500	.667	.598	.536
13	1.644	.608	.634	.472
14	1.790	.558	.668	.416
15	1.949	.513	.700	.368
16	2.116	.453	.728	.310
17	2.285	.437	.752	.268
18	2.461	.406	.776	.246
19	2.654	.376	.798	.224
20	2.848	.351	.818	.202

[1] While the term "attenuation" usually refers to a loss, it has a slightly different meaning when used with an equalizer of any of these types. In the case of equalizers (c) and (d), attenuation refers to the difference in the transmission at the resonant frequency to that at frequencies remote from the resonant frequency. For those equalizers of types (a) and (b), attenuation refers to the difference in transmission at the two extremes of the frequency spectrum. In both cases, this "attenuation" is determined by the loss in the resistive network, which is normally referred to as an attenuator or pad.

Pickup Loading and its Effect on Frequency Response

W. A. FITZMAURICE* and W. JOSEPH**

If you notice a difference in quality when you move your phono equipment ten feet or so from the amplifier, do not be surprised—there's a good reason for it. The authors explain why and tell how to avoid it.

In home audio installations employing magnetic pickups, the cartridge is usually located somewhat remote from the preamplifier by distances ranging upward from one foot. Because of the capacitance associated with the interconnecting cables, the question is raised as to the effect of this shunt capacitive loading on the response of the cartridge. In addition, there seems to be considerable doubt as to the exact effect of resistive loadings as recommended by various sources.

To answer these questions the authors carried out an investigation to determine the response of variable reluctance cartridges under varying conditions of loading.

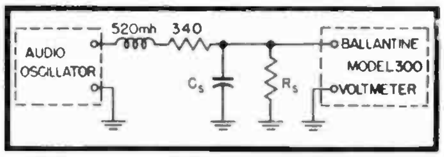

Fig. 1. Equipment set-up for measuring cartridge response.

The measuring arrangement used is shown in *Fig.* 1. The output of an audio oscillator was connected in series with an inductance of 520 millihenrys and a resistance of 340 ohms. This simulates the General Electric variable reluctance cartridge, type RPX-040, RPX-041, or RPX-050. Various combinations of *C* shunt and *R* shunt were used. The shunt capacitance consisted of mica capacitors, the input capacitance of the Ballantine Model 300 a.c. voltmeter, and stray wiring capacitance. The audio oscillator output was maintained constant at all frequencies at 0.01 volts.

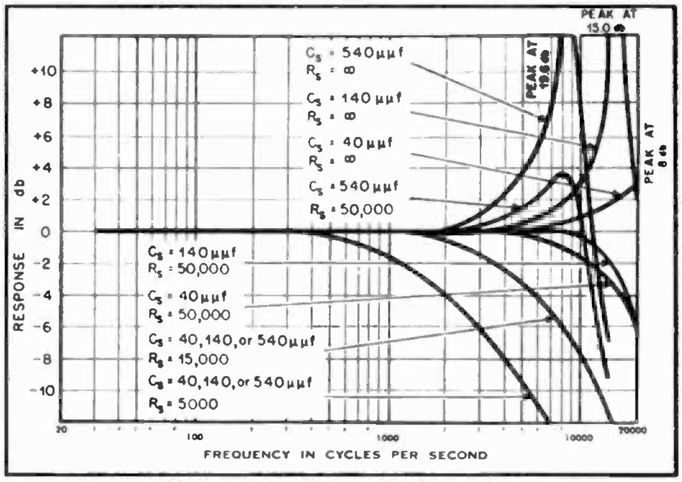

Fig. 2. Curve showing change in cartridge response with change in load resistance.

The results obtained were plotted and are shown in the curves of *Fig. 2*. It will be noted that the effect of shunt capacitance is quite marked when the shunt resistance is in the neighborhood of 50,000 ohms or greater With no load resistor (infinite R_s), pronounced resonant peaks are obtained for all values of C_s measured. With C_s of only 140 μμf, a 15-db peak is encountered at about 16,000 cps. Such peaks tend to accentuate noise and produce "dirty" highs.

The damping effect of R_s (decreased Q), may be seen by the reduction of a 19-db peak at 7500 cps with C_s of 540 μμf and R_s of infinity, to a 3-db peak with the same C_s and R_s of 50,000 ohms. With C_s of 40 μμf and 140 μμf and R_s of 50,000 ohms, the resonant effect holds the response level to approximately 10,000 cps. The "water fall" drop thereafter is a characteristic of the effect of resonance.

As R_s is decreased, the effect of C_s becomes progressively less, and drooping response is obtained. With R_s of 15,000 ohms, the response is down 3 db at 5000 cps and 8 db at 10,000 cps. With R_s of 5000 ohms, the response is down 3 db at 1500 cps and down 8 db at 4000 cps, independent of all values of C_s used.

Below resonance, the performance characteristics may be analyzed by referring to to *Fig.* 1 and noting that the output of the audio oscillator (equivalent to the internally generated voltage of the cartridge) is impressed across a voltage divider consisting of *L*, *R*, and the parallel combination of R_s and C_s, all in series. The output is taken across the parallel combination of R_s and C_s. Since the impedance of *L* increases and the impedance of C_s decreases with an increase in frequency, a voltage divider exists whose output is a function of, and decreases with, an increase in frequency.

The data given applies specifically only to the cartridge simulated. However, from Table 1 listing the impedance of the more popular types of variable reluctance cartridges, it can be seen that the same problem exists to some extent in all cases.

A survey of the types of cable com-

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* *316 Milford Ave., New Milford, N.J.*
** *52-18 19th Ave., Brooklyn 4, N.Y.*

EQUALIZER

(*from preceding page*)

value equal to R_2 minus the coil resistance.

As a general observation, r.f. chokes can be used in applications requiring inductances of less than 15 mh. While a high Q is desirable, it can nevertheless be discounted to the extent that it is possible to attain the necessary objective. For extremely "sharp" curves, best results can be obtained by the use of toroids, but for most applications it will be found that r.f. chokes can be used, and at considerably less expense.

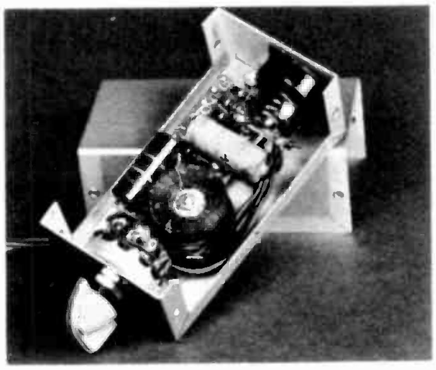


Fig. 5. Internal construction of the equalizer shown at the right in Fig. 4.

Motor Rumble Reduction in Wide Range Phonographs

JOHN R. CATER*

A negative-feedback-type R-C filter, the author finds, is the only way to banish table rumble permanently and completely, short of acquiring a broadcast-studio unit.

MUSIC LOVERS, who have striven to approach perfection in their reproducing systems and have lavished much money and mental anguish in the effort may wonder to what end they have achieved excellent bass response when it brings up motor rumble to an intolerable level. Of course, there are turntables available with extremely low rumble, but they are bulky and they cost several hundreds of dollars. Even some turntables selling for more than $100 have very annoying rumble.

The nature of rumble suggests a solution. Many phonograph motors have four poles. These motors rotate at 1800 rpm if they are synchronous, and around 1725 rpm if they are induction motors. Rumble results from mechanical and electrical imbalances in the motor which cause a variation in speed once per revolution, or 30 and 28.7 times per second. It is a rare record indeed (not counting test records) which has such low musical notes. A narrow-band frequency-rejection filter centered on the rumble frequencies will wipe out the rumble without affecting the useful output of the vast majority of records.

Figure 1 shows such a filter. It employs a cascode amplifier V_1 with a parallel-T network in a feedback loop. It is absolutely essential that the components of this twin-T be both accurate and stable. It is well to measure them rather than trust the marked or coded values. Since the ganged potentiometers R_4 have only 10 per cent of the values of the fixed resistors with which they are in series, they may be considerably less ac-

* *108 Warren St., Nutley 10, N. J.*

PICKUP LOADING

(*from preceding page*)

monly employed shows the minimum capacitance per foot of high-grade rubber microphone cable to be about 40 μμf. and for the smaller phono cables, about 60 μμf. Coaxial cables are available with capacitance per foot as low as 13 μμf. Of course, there is always the foot or so of high-capacitance phono cable incorporated in the record player arm to be contended with.

Cathode Follower Helps

When cable capacitance is added to the input capacitance of the preamplifier, it is unusual to find C_s lower than 100 μμf. From the foregoing it may be seen that with such capacitance it is possible to obtain very satisfactory response from the cartridge up to 10,000 cps, and de-emphasis may also be provided to compensate for today's recording practices. This, however, means very careful juggling of C_s and R_s and requires measuring equipment not available to most experimenters. In addition, it is often desirable to incorporate variable de-emphasis networks into the system with the preamplifier remotely located from the record player with a consequent introduction of a C_s of 500 μμf or more.

With these factors in mind, the authors believe that higher quality installations should incorporate a cathode follower immediately adjacent to the cartridge.

The simple cathode follower shown in *Fig.* 3 was set up and tested. The effectiveness of this arrangement is brought out by the response curves of *Fig.* 4. Even with a C_s of 2500 μμf, a negligible effect on high-frequency response is

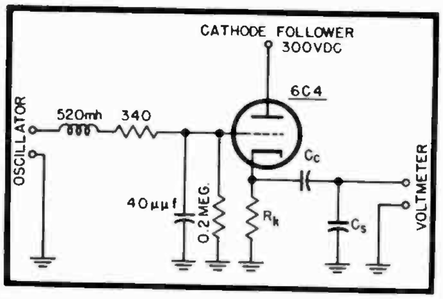


Fig. 3. Equipment set-up for measuring cartridge response when feeding a cathode follower.

noted. The effect of various values of coupling capacitors, C_c, on the low frequency response was explored. A C_c of 0.05 μf with an R_k of 5000 or 10,000 ohms produces a droop of only 1 db at 20 cps.

A 0.2-meg R_s was employed to eliminate any resonant effect arising from phono arm cable capacitance as represented by C_s of 40 μμf.

When operating with 300 volts on the plate and a 5000-ohm, ½-watt cathode resistor, a 6C4 draws 3.5 ma. Virtually the same results may be obtained with (½) 12AU7, (½) 6SN7, 6J5 or numerous other triodes. Different values of plate voltage have little effect on the results. As careful attention must be paid to the filtering of the cathode follower B+ supply voltage as for a preamplifier.

When a follower is employed, it is recommended that de-emphasis be accomplished after the output of the follower. An excellent example of such a system may be seen in an article by P. W. St. George and B. B. Drisko entitled "A Versatile Preamplifier" in the first Audio Anthology.

TABLE 1

Manufacturer	Cartridge Model	Approx. L mh	Approx. R ohms
General Electric	All home types	520	340
Audak	L-6	750	600
Pickering	120, 140	165	600
	260	130	750

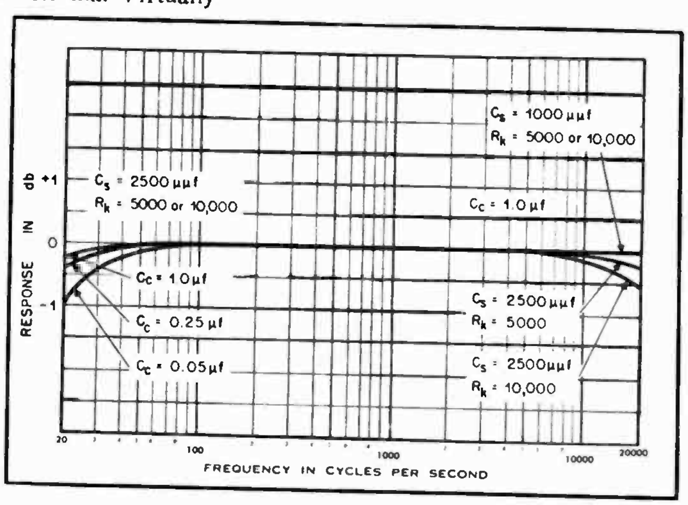


Fig. 4. Curve showing change in cartridge response with change in a load isolated by a cathode follower.

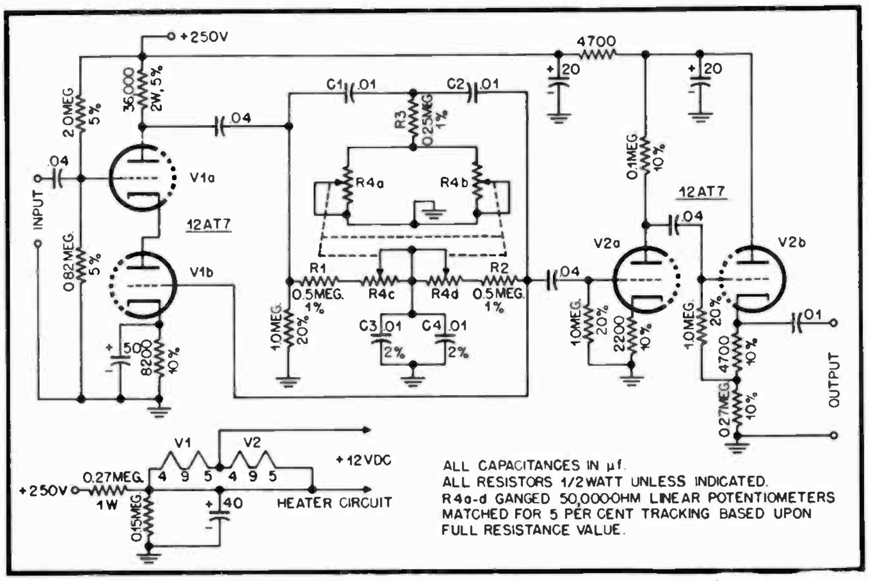


Fig. 1. This is the rumble filter, turnable to between 28.9 and 32 cps. Note carefully that the ganged potentiometers must be matched for tracking. R_1-R_2 and C_1-C_2 should also be checked, preferably with a bridge.

curate and stable without causing trouble. From a group including one IRC Q11-123 potentiometer, and six M11-123 "multisections," the writer was able to select a set with a maximum tracking error of 5 per cent (based on full resistance value). Two 50,000- and one 25,-000-ohm potentiometers are required. Less tracking trouble will be experienced if two 50,000-ohm units are connected in parallel to obtain the 25,000-ohm value. R_1, R_2 and R_3 are 1 per cent tolerance deposited-carbon-film resistors. Most composition resistors are likely to age too much to be satisfactory. C_1, C_2, C_3 and C_4 are silver mica capacitors chosen for 2 per cent tolerance.

The values chosen permit the center frequency to be adjusted sufficiently in most cases by simply setting R_4. Referring to *Fig.* 2, the null frequency for this network is $1/2\pi R$, and for the values selected, ranges from 32 to 28.9 cps. If it desired to lower this range of frequencies, it may be most conveniently done by adding parallel capacitors across C_1 and C_2 in *Fig.* 1. Since the parallel capacitors will be small in comparison to C, they need not be silver micas. If the components give too low a frequency when properly matched, the null frequency may conveniently be raised by shunting R_1, R_2 and R_3 (*Fig.* 1) by appropriate resistors.

In the cascode amplifier V_1, the gain for signals into the grid of V_{1a} is equal to $\frac{\mu R_L}{R_L+[(\mu+2)R_p]}$. For signals into the grid of V_{1b}, the gain is approximately equal to $G_m R_L$ and is much larger in value than for V_{1a}. Since the signal applied to V_{1b} is degenerative, there is a very appreciable loss in signal level at the output of the filter, but the effective Q of the twin T is greatly increased. Because of the loss (approximately 30 db), the signal from a low-level pickup must be passed through a preamplifier first, though the output from high-level units such as the Weathers FM pickup oscillator may be passed through the filter without further amplification.

It is usually desirable to bring the level up again immediately. One triode section is sufficient (V_{2a}) and if double triode is used, there is one section available (V_{2b}) for a cathode-follower so that output cable capacitance may be neglected.

Cascodes and cathode-followers place an undue burden upon heater insulation unless appropriate steps are taken. The heater-cathode rating of most tubes is 90 volts. If the heater supply is biased 90 volts above ground, grounded cathodes are still within rating, and cathodes operated at high voltages may reach + 180 volts without exceeding the tube's rating. Such biased heater supplies should be heavily bypassed to ground.

With the equipment available the writer was not able to measure attenuation at the rejection frequency because of distortion in the audio oscillator used. It was estimated to be in excess of 25 db below the region of flat response. Using again the response in the flat part of the range as zero level, the output was down approximately 1 db at 36 cps and 6 db at 32 cps when the rejection frequency was adjusted to 30 cps.

Higher-Q filters employ cathode-followers in the feedback loop were tried, but were too sharp in cutoff unless two or more were stagger-tuned. With just one such filter, the rumble frequency would vary enough to rear its ugly head periodically on each side of the rejection frequency.

A twin-T used as a brute force filter is unsatisfactory because it has an effective Q of only ½ and affects the response up to several hundreds cps.

The circuit presented here, incidentally, permits one to use test records for adjusting equalization without having rumble render the results unintelligible.

It is fortunate that the major component of rumble, for the motors considered, is so low, because any attempt to "poke a hole" in a pass band does give transient troubles. As has been pointed out, very few records (or for that matter, very little music) have frequencies down to 32 cps. So if we eliminate a few cycles starting at this point, we are not tampering with the really usable band.

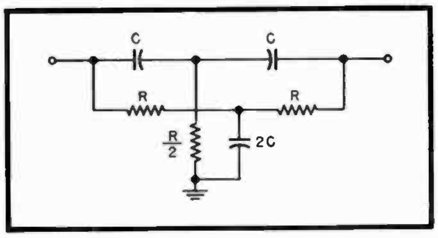


Fig. 2. The parallel-T network.

Though some purists may insist that because they can hear down to 15 cps and their speaker has useful output this low, they want to equalize correctly to such a point, the simple fact is that they will probably have to use signal generators for such dubious aesthetic flights. For those whose turntables have two pole motors (thus placing the rumble in the neighborhood of 57 cps) we can only offer sympathy.

Reference

Valley and Wallman, "Vacuum Tube Amplifiers," New York: McGraw-Hill Book Co., Chap. 10.

Two-Channel Conversion for AM-FM Receivers

C. G. McPROUD

Details of simple changes which give the advantage of two separate receivers to tuner chassis which employ independent AM and FM channels but with coupled tuning capacitors. Ideal for binaural or stereophonic broadcast programs.

SOME TIME AGO, the writer made a few changes to a standard AM-FM tuner chassis in order to be able to listen to one program on FM while recording another on AM, or vice versa, all without realizing how advantageous this modification would be when radio stations started to use their two transmitters to radiate the two-channel signals for binaural or stereophonic programs.[1]

The whole idea is so simple and straightforward that anyone faced with the problem of providing both AM and FM programs simultaneously from a single tuner should come up with the same solution. However, since such a solution has not yet appeared in print, this one is offered as one way to accomplish the desired end.

At the time the modification was planned, there were no stations regularly broadcasting two-channel programs, but it must be admitted that there are times when two programs of interest are on the air at the same time, and the human hearing and interpreting mechanism is unable to cope with both at once. For this problem, the simplest solution is to record one of the programs. As one element of the writer's home system, a wire recorder was employed for just this purpose. The quality left something to be desired, but it was at least good enough to convey the information contained in the program—probably as well as any of several million table-model sets that serve as the sole entertainment medium in many households. At one time, this system employed two separate tuners—originally a wide-range t.r.f. tuner of the Miller type, with modifications of course, and the other an inexpensive FM tuner with a ratio detector.

The next step in the program of improvement included the design and construction of separate AM and FM tuner chassis, both tuned by means of push buttons. The FM tuner used relays to switch adjustable capacitors in the oscillator circuit, while the other used relays to switch the plate supply to any one of four single-channel fixed-tune chassis. This entire system was described in an earlier series of articles.[2]

The next step in the program was to secure a combination AM-FM tuner of good quality, with the idea of separating the sections so as to have the advantages of two-channel reception if desired. Because of its physical construction, the Browning RJ-12B was selected. In this receiver—as in several other types—the tuning is accomplished by two entirely independent variable capacitors, coupled together only by means of a dial cable. The modifications are now reasonably obvious. With the advent of two-channel broadcasting for binaural or stereophonic programs, the receiver becomes doubly useful.

Before discussing the actual modifications to the RJ-12B, let it be said that the general idea of these changes can be adapted to any AM-FM tuner which uses separate devices to tune the two sections. The Browning RJ-20 is similar to the RJ-12, and the Meissner 9-1091 may be converted in a similar manner. The details may differ, but the principle remains the same. For simplicity, only the changes to the RJ-12 will be described, and they are divided into two parts—electrical and mechanical.

Electrical Changes

The electrical changes required are simple, and are made primarily to permit the simultaneous operation of both channels. In the system for which this modification was made, the switching between AM, FM, and Phono is accomplished in the control amplifier. Therefore, since the phono and the TV inputs to the tuner were never used, it was preferred to eliminate this switching facility from the tuner chassis. However, to reduce current drain and consequent heating, a selector switch was used to permit operation of either channel separately or both together. This switch has three positions—the center providing for both AM and FM operation, the left position energizing only the AM tuner, and the right position energizing only the FM tuner.

In addition, neither the volume control nor the power switch were needed on the tuner chassis, both being provided elsewhere in the system. Therefore both of these were removed. As a matter of fact, when the RJ-12 is to be used in a typical modern system which incorporates a control amplifier, both of these components may be removed, leaving only the two central knobs on the tuner chassis—one for tuning and the other as the selector. However, for the modification, two tuning shafts were needed in addition to the selector switch, so in order to preserve symmetry, the a.f.c. switch was wired to the front panel, which is somewhat more convenient than its normal back-apron location. Thus we have four control positions on the front of the chassis—a.f.c. switch, tuning, selector, and blank. The original controls were, in the same order, the power switch, tuning control, selector, and volume. *Figure* 1 shows the

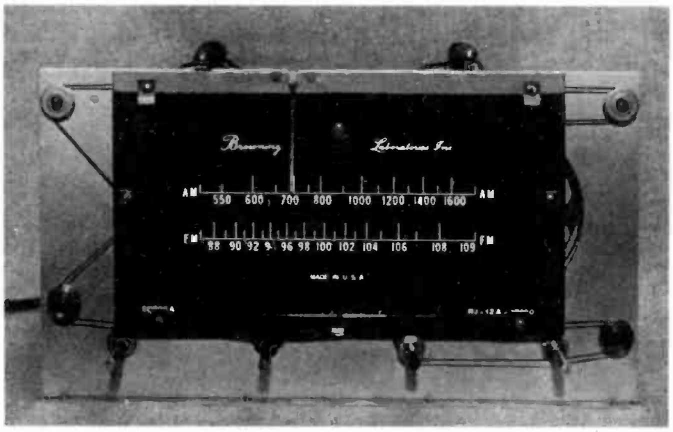


Fig. 1. External appearance of the converter Browning RJ-12B tuner, showing the two dial pointers.

[1] With no intention of getting into this controversy as to the correct name—at least not in a constructional article—the writer respectfully refers the reader to discussions on this subject by Tinkham on page 77

[2] C. G. McProud, "Elements of residence radio systems," AUDIO ENGINEERING, Sept.–Dec. 1948. Reprinted in AUDIO ANTHOLOGY.

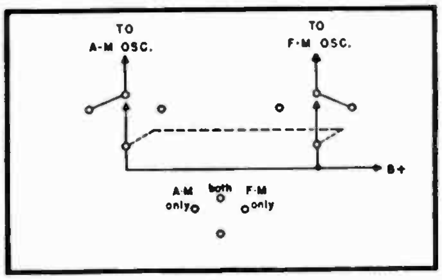


Fig. 2. Schematic of the switch connections to be made in making the conversion.

front of the modified chassis, which is but slightly different from a normal chassis. It will be noted that there are two dial pointers—the one at the top extends only to the AM dial scale, while the new one at the bottom extends upwards to the FM dial scale.

Taking the changes one by one, the following procedure should be followed. Remove the power switch and its associated wiring. Remove the volume control, leaving the wires disconnected but at the same location temporarily. One of the r.f. coils will have to be pushed slightly backward to permit the control to be removed. If done carefully, no damage should result. Now, remove the selector switch, leaving all its wiring intact. This will save a lot of tracing when connecting the new switch. All the shielded wires from the switch to the jacks on the rear apron may be removed, making it easier to get the switch out of the way. Next, remove the tip jacks marked AUDIO OUTPUT and RECORDER INPUT—the first and third from the left end of the rear apron when viewed from the back of the chassis. In the two holes, mount two 1.0-meg audio-taper potentiometers, wiring the arms to the adjacent tip jacks, and the ground end to a chassis ground lead. Connect a .05-μf capacitor from the junction of R_{11} and R_{13} (in the Browning diagram) —from which a shielded lead to the switch was removed—to the high end of the potentiometer nearest the AM end of the chassis. Connect another .05-μf capacitor from the junction of R_{17} and C_{11} to the high end of the other potentiometer. These two potentiometers permit adjustment of the outputs to equal levels.

Install a two-pole three-position switch (Centralab 1462) in the hole from which the selector switch was removed, and connect the new switch as shown in *Fig.* 2. The remainder of the leads should be connected so that the indicator tube works on both sections of the receiver at the same time. To accomplish this, connect the lead from pin 5 of the 6AL7 socket to the AM a-v-c bus; connect pin 4 to the ground end of the discriminator network; and connect pin 6 to the "high" side of the discriminator output, through a 1-meg. resistor. Thus the plate supply is connected to the AM oscillator in the left and center positions of the switch, and to the FM oscillator in the center and right positions.

New A.F.C. Switch

The normal a.f.c. switch in this chassis is a single-pole double-throw slide switch mounted on the rear apron. By prying up the tabs on the sides of this switch, the slider element may be removed and upon replacing the rear bakelite plate of the switch, the wiring is not disturbed and the termnials serve merely as tie points. Install a single-pole double-throw rotary switch on the front apron where the power switch was, and connect three wires from the old switch to the new one, making sure to maintain the same operating arrangement.

After cutting the shafts to the same length as the original tuning shaft, the electrical changes are completed. One hole remains unoccupied on the front apron—that from which the volume control was removed.

Mechanical Changes

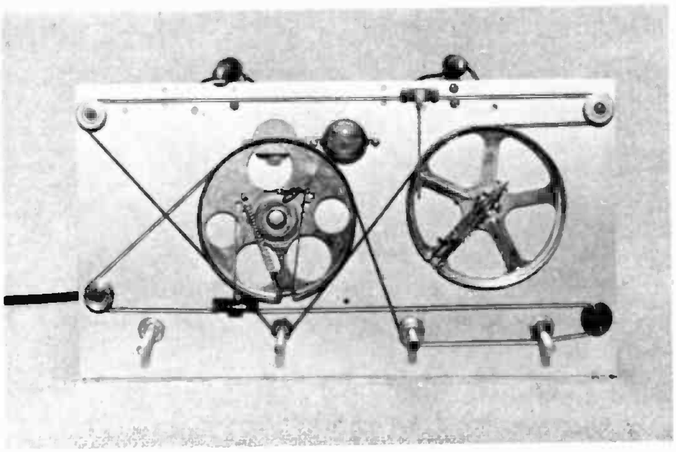
The first step is to remove the dial and all the dial stringing. Then remove the large pulley from the AM tuning capacitor shaft, and remove the smaller pulleys from both capacitor shafts. Mount a ¼-to-⅜ shaft extension on the FM capacitor shaft, and firmly attach a new 4-in. pulley to this extension, making sure that the dial cable opening in the pulley is at the bottom when the capacitor is half meshed. Remount the 4-in. pulley on the AM capacitor shaft, with the groove of the FM pulley about 1/16-in. further from the panel than the AM pulley. Mount a tuning shaft of the same type as the present one in the hole from which the volume control was removed, and mount two ¾-in. idler pulleys as shown in *Fig.* 3. These pulleys must be free to turn easily, and eyelets for mounting them are usually supplied with the pulleys. Note that the bottom of the left idler is on a line with the top of the right one, and that this line is slightly above the bottom edge of the glass dial plate. A third idler pulley should be reamed out to run freely on the ¼-in. selector switch shaft.

Following the diagram of *Fig.* 4, carefully restring the AM dial cord. This cord is shown in solid lines and should be drawn up so as to be quite tight, with the spring extended to about 1½ times its normal length. Where the dial cord wraps around the knob shaft, it is suggested that three turns be taken. Test the stringing before proceeding, making sure that the knob shaft turns freely and that the dial cord behaves properly in the shaft depression.

Following a similar procedure, string the FM cord, using the dotted lines of *Fig.* 4 as a guide, and again test the operation thoroughly before proceeding further.

It is suggested that two similar dial pointers be obtained—the writer prefers the type which are made of a fluorescent plastic as they are easily seen. Since one is to slide along the top of the dial plate and the other along the bottom, try them out first, and cut off short enough that they clear, yet long enough to reach the dial calibrations. Then remount the dial plate and attach the top pointer, using a few drops of radio cement in addition to crimping the back of the slider. The vertical line at the left end of the dial scale indicates the position of full meshing of the capacitor plates for both sections. Therefore, the pointer should be secured to the dial cord at this end of the dial and with the capacitors fully meshed.

The FM pointer is similarly mounted to slide along the bottom of the dial plate. This may take some careful adjustment to make sure that the pointer passes the other cords without catching, but it can be done with a little care. It is suggested that the sliders be given a light coating of Lubricate or some similar lubricant.



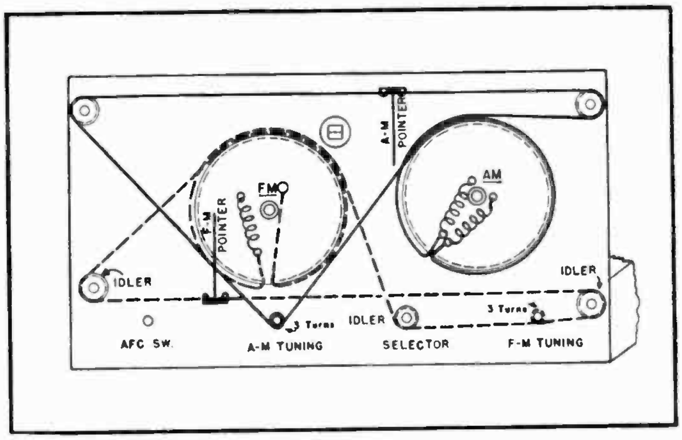


Fig. 3 (left). The tuner chassis with the dial plate removed to show the dial stringing. Fig. 4 (right). diagram of the dial stringing. The solid line indicates the path of the AM cord and the dotted line shows the path of the FM cord.

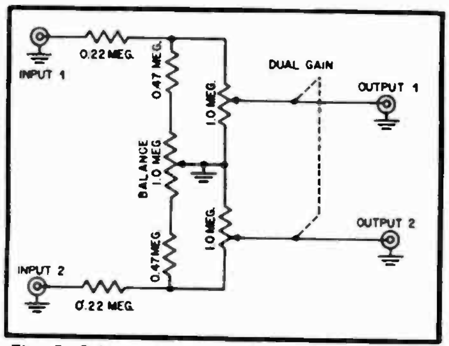


Fig. 5. Schematic of simple control for stereophonic programs.

Output Circuits

For use as a tuner capable of receiving two separate monaural channels, nothing further is required, since the two output jacks on the rear apron provide the AM and FM signals independent of each other. However, for binaural use, some means of controlling the volume of both channels simultaneously is an advantage. Therefore it is recommended that a separate control unit be provided. This unit can be as simple as possible, containing only a dual potentiometer, relying upon the two output controls on the tuner chassis for balancing levels between the two sections of the receiver. If a more elaborate unit is desired, so that either channel can be connected to either power amplifier or to both at will, it is equally possible to make such an arrangement. In any case, if the unit is to be used primarily for stereo programs, the balancing control is desirable. *Figure* 5 shows a simple control, without any provision for switching, yet equipped with the balancing feature. This entire unit can be mounted on a small bracket or on an unused space on the front panel of the installation. To reduce the number of *apparent* controls, it is suggested that the complete unit be assembled using an IRC Concentrikit, with the balancing control operated by the outside knob, and the dual potentiometers operated by the small inside knob. The parts list shows the components for either type of construction.

A more flexible control system is shown in *Fig*. 6. The selector switch has five positions—(1) both AM and FM channels feeding separate output circuits, with AM on output 1 and FM on output 2; (2) same as (1) except that AM is on output 2 and FM on output 1; (3) AM feeding both output circuits; (4) FM feeding both output circuits; and (5) stereophonic, with the dual volume control and the balancing control in the circuit. The only eventuality not provided for is the reversal of the stereophonic sources, which could become necessary in certain instances. This was not included because of the impracticality of locating a four-pole six-position switch of reasonable dimensions. However, it is probable that some standardization in channel usage will occur, so that the FM channel is always on the left and AM on the right. or vice versa. Since the normal orchestra arrangement places the high-frequency instruments predominantly on the left, it is probable that FM would be the left channel.

The balancing control provides a loss of about 3 db in each channel in the center position, with a variation of 3 db in the level of either channel as the control is rotated from one end to the other. If the control unit is to be used with amplifiers at any appreciable distance, it would be desirable to add two cathode followers, as shown in *Fig*. 7. There is some additional gain in this circuit so the balancing control has a mid-position loss of 6 db in each channel, with a range of 6 db from one end of the control to the other. The output from this arrangement is approximately the same for all positions of the switch, and one additional position has been added to permit reversal of the sides in the stereo position. The output impedance is sufficiently low that amplifiers may be located up to 20 feet from the control unit without appreciable frequency discrimination.

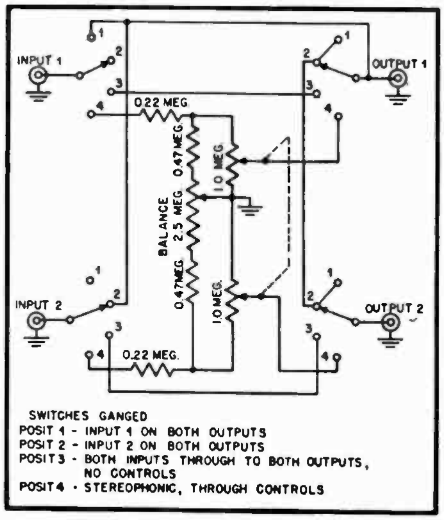


Fig. 6. The addition of a switch provides greater flexibility in the operation of the converted tuner. With the switch at A, input 1 is connected to both outputs; at B, input 2 is connected to both outputs; at C, each input is connected to a separate output, without the volume and balancing controls in the circuit; at D; the inputs are connected through the controls for binaural programs.

PARTS LIST

Receiver Modifications

1 2-pole, 3-position switch, Centralab 1462
1 SPDT rotary switch
2 1.0-meg audio-taper potentiometers, small size
2 .05-μf capacitors, 400 v., paper
1 Dial drive shaft with panel bearing
3 ¾-in. idler pulleys
1 4-in. dial pulley, with tension spring
1 Shaft extension, ¼-in. hole, ⅛-in. shaft
2 Slide-rule dial pointers
Nylon dial cord

For Figure 5

1 1.0–1.0 meg. dual potentiometer, audio taper
1 2.5-meg. potentiometer, linear or, if concentric control is used
  1 IRC K-2 Concentrikit
  1 IRC KS-2 Universal Shaft Kit
  1 IRC B11-239 base element
  1 IRC B13-137 base element
  1 IRC M13-137 base element
4 Input jacks, RCA Phono type
2 0.22-meg. resistors, ½-watt
2 0.47-meg. resistors, ½-watt

For Figure 6

Same parts as for Fig. 5, with the addition of
1 Centralab 1414 switch

For Figure 7

Same parts as for Fig. 6, except for the substitution of the following for the resistors listed:
2 0.27-meg resistors, ½-watt
2 0.39-meg resistors, ½-watt
4 0.47-meg resistors, ½-watt
2 0.1-meg resistors, 1-watt
2 10,000-ohm resistors, 2-watt
2 3900-ohm resistors, 1-watt
2 Noval sockets
2 12AU7 tubes
2 0.1-μf capacitors, 600 v., paper.

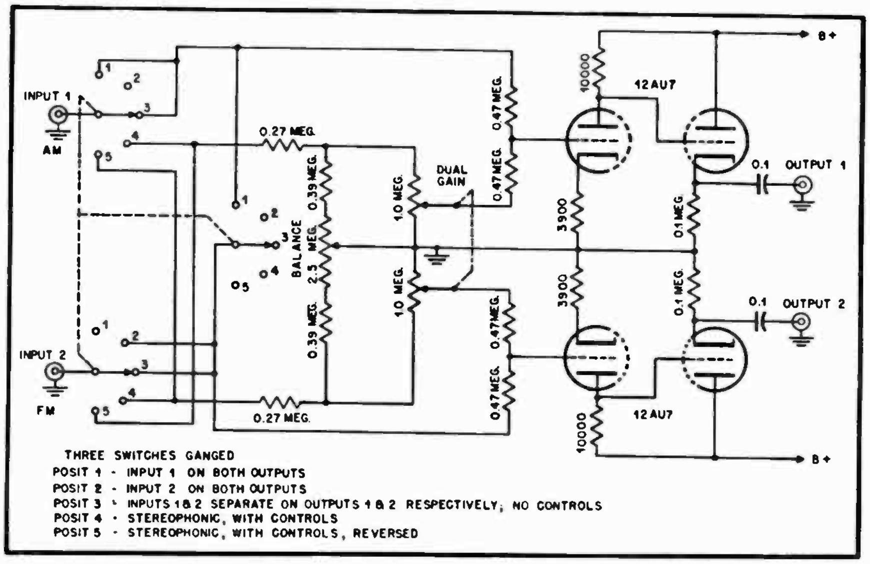


Fig. 7. Similar switching plus the addition of two cathode followers may be preferred because of the lower output impedance offered with this arrangement.

Binaural or Stereophonic?

R. J. TINKHAM*

The author clarifies the meanings of the two words which have been used to describe this latest of interesting innovations in the field of sound reproduction.

Some confusion apparently still exists in the minds of many regarding the definition and connotation of the words *binaural* and *stereophonic*. Binaural means, literally, "two-eared." It refers to the fact that the sense of hearing in both ears, plus the brain, makes it possible for us to analyze sound, to discriminate against unwanted sounds and, to a large extent, to localize the source of a sound. Stereophonic, on the other hand, means "three-dimensional sound," or a sound source "in the round," much as we perceive it in actual experience. It can be stated, then, that one uses his binaural sense to perceive stereophonic impressions. This is much like saying that one uses his binocular sense of vision to achieve stereoscopic or three-dimensional impressions. If we stop up one ear with a finger when listening to someone in a noisy place, we will observe how the ability to discriminate against the background noise disappears and the ability to localize the source of sound vanishes. Or, if we shut one eye, our judgment of depth is gone and, although we know better, all objects are as on a flat surface. Compare a monocular snapshot with the modern stereo-photo or, for that matter, with grandma's "Trip to Egypt" via the old-fashioned stereoscope. The difference is remarkable.

Modern tape recording has achieved, (a) an easy recording method, (b) the ability to record more than one channel at a time on the same tape, and (c) reasonable expense. It is inevitable that someone would think of trying to capture sound stereophonically. Our attention has again been directed to the Bell Labs experiments[1] of 1933 with the Philadelphia Orchestra performing at home and being reproduced in Washington with startling realism, facts lost sight of for nearly a generation. In 1948, Camras[2] gave a demonstration of three-channel recording for Armour magnetic recorder licensees. And in 1950 the author, then with another company, was asked rather bluntly at a Society of Automotive Engineers round-table discussion, why the recorders that he had sold some of the members did not reproduce properly, with the question, "Why does the recording of an auto going over a cobblestone pavement sound like a male singer with a cracked voice humming in the bathroom instead of an auto on cobblestones?" Money had been invested in supposedly good equipment, but the reproduced sounds were not considered natural or normal. Some form of distortion seemed to be present.

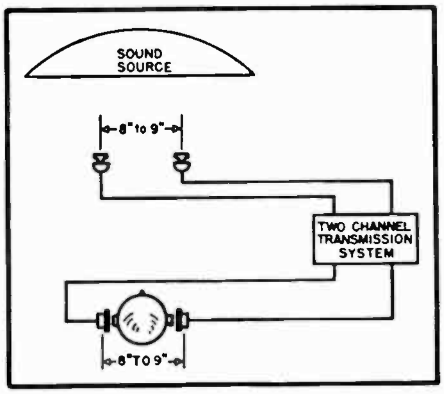


Fig. 1. The elements of a true binaural system.

If one reflects on the difficulties encountered in achieving "realistic" sound effects for ordinary broadcasting (monaural), where crumpled Cellophane sounds like a raging fire in the north woods and a cantaloupe dropped on the floor portrays the victim's head being bashed in, he realizes that all single-channel reproduction gives only an approximation of the original sound—our imagination fills in the deficiencies. For instance, a symphony over the network sounds like a symphony, with which we are reasonably familiar. But recordings of autos on cobblestones do not sound normal, for this sound is a bit unusual and our imagination does not help out much.

Let us consider the "auto-on-cobblestone" problem briefly. Since the one recording microphone, placed in the driver's position, had no analytical or discriminatory power of its own, it recorded faithfully what it heard and, for the first time (perhaps) the auto researchers heard, in truth, what they didn't hear—a hard fact to realize. Obviously, the solution was to restore the natural sense of binaural hearing by using two complete and separate sound transmission channels, from microphones to headphones, one channel to each ear. This was suggested, and one organization tried it with successful results.[3] Oddly enough, loudspeaker reproduction, at that time, was not successful for reasons which will become clear as we continue.

Microphone Placement

Apparatus intended for one field often ends up in another. Gene Carrington, educational director and lecturer for a large midwestern organization, learned of the foregoing and made orchestral recordings at Interlochen (Michigan) Summer Music Camp under the direction of Dr. Joseph Maddy. Mr. Carrington conducted experiments with respect to the placement of the two mikes with separations varying from many feet to about the distance of a human's two ears. Headphone listening showed that the sound was most realistic when the mikes were about 8 or 9 inches apart. And the demonstration was astonishing in its realism; one seemed to be standing right in the midst of the orchestra. During the past year, several demonstrations of "binaural" sound have been presented, using two loudspeakers—one for each channel—in a large room. The results were puzzling to many people who attended these demonstrations—until they heard similar demonstrations with headphones. Then they understood what was being attempted. Suddenly "three-dimensional sound," whether reproduced binaurally over headphones or stereophonically over loudspeakers, has become most popular. But this popularity has raised another question. Why don't these demonstrations work so well with loudspeakers? After all, it is mighty inconvenient to listen for any length of time to headphones, and loudspeakers reach larger audiences less expensively.

The answer is that loudspeakers do work very well when properly handled. And here entered the confusion between "binaural" and "stereophonic."

If we refer to *Fig.* 1, where we have two closely spaced microphones in a sound field, and a two-channel system connecting them to separate earphones, it becomes obvious that we are merely extending the diaphragms of our two ears electrically. Actually, of course, we are extending the diaphragms of the individual headphone receivers, electrically, to the physical placement of the diaphragms of the microphones. The time factor may be suitably delayed by the insertion of a simultaneous, two-track recorder in the bilateral transmission system. In other words, our ears seem to be where the microphones are physically placed. We have re-oriented our ears geographically and individually. The theorem of proportionality exists: mike-to-mike spacing is equal to the earphone-to-earphone spacing, and our brain functions normally to fuse the two

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* *Ampex Corporation, 934 Charter St., Redwood City, Calif.*

[1] *Electrical Engineering*, Jan. 1934, Vol. 53, No. 1, six articles.

[2] Marvin Camras, in *Proc. I.R.E.*, April 1949, Vol. 37, No. 4.

[3] *J. Acous. Soc. Am.*, Nov. 1952, Vol. 24, No. 6.

separate signals arriving from each of our two ears. The resultant sensory perception is astonishingly like listening in the spot where the mikes were placed. Of course, we are assuming that we have paid attention to the proper phasing of the mikes, amplifiers, and headphones, and haven't interchanged sides. Disturbing but interesting effects result from such maladjustments. And the quality of the reproduced signal, as always, depends on several factors present in the system: frequency response, distortion, noise, flutter, wow, etc.

But we soon discover that listening to headphones makes the sound appear to be behind, rather than in front of us. This is a fact brought out by Bell Telephone's "Oscar" exhibit at the 1933 Chicago Worlds Fair and now permanently displayed at the Chicago Museum of Science and Industry. And it is also just as tiring today to be squeezed by a headset as when the crystal radio was in vogue. The apparent answer to the problems is obvious. Reproduce the signals over loudspeakers, of course. This will put the sound in front of us, and will be more convenient for listening. But while the sound as we hear it from the two loudspeakers sounds different from single channel listening, it still doesn't sound quite right. Remember, we still have the mikes placed only 8 or 9 inches apart. We have suddenly invented a new system of listening: "bistereonauralphonic" or a cross between two different concepts—which is meaningless. See *Fig. 2*. The placement of the speakers is unimportant here, as no arrangement will give the desired effect. It doesn't matter whether we set the speakers at an angle so that their axes intersect, as at (A) in *Fig.* 2, or so that they are square with each other. Speaker manufacturers pride themselves today on the non-directionality and wide-angle dispersion of their products. What happens under these conditions of close mike spacing and wide speaker spacing? If we stand somewhere in front of the speakers, we no longer receive sound meant for the left ear only in the left ear, and right-ear sound only in the right ear, but rather we receive sound in both ears meant for individual ears only. Our original mike placement simulated our relative ear positions, not the loudspeaker positions. This results in a brand new sensory experience which puzzles that superb analytical instrument called a brain, for the sound is twice mixed: once in the air, because each ear hears sound from each speaker, not just from one as it should; and once in our brain. Since we hear left and right sound in both ears simultaneously, with head diffraction lags, etc., our brains are led to the mental conclusion that we hear two speakers. This leads to the inescapable conclusion that we are befuddled, and this type of reproduction becomes a new and novel experience to be sure. This is certainly not the type of reproduction for which we search, namely that of reproducing the original sound stereophonically (i.e. "in the round"). But with this system there is, however, one point or group of points lying on the perpendicular bisecting vertical plane between the two speakers where, due to phase relationship and standing wave patterns in the reproducing room, we can hear sound from the left speaker predominantly in the left ear and vice versa. Here the results are somewhat more satisfactory. A slight shift of the head will make this interesting phenomenon disappear, and again both sound sources will be heard in both ears simultaneously. This narrow central "binaural" plane limits the number of happy listeners to those who can stand close together in tandem.

Corrective Measures

The answer to this predicament is immediately obvious to anyone familiar with acoustics. In the case of the binaural setup of *Fig.* 1, we are not concerned with listening room acoustics, obviously, and this occasionally has its

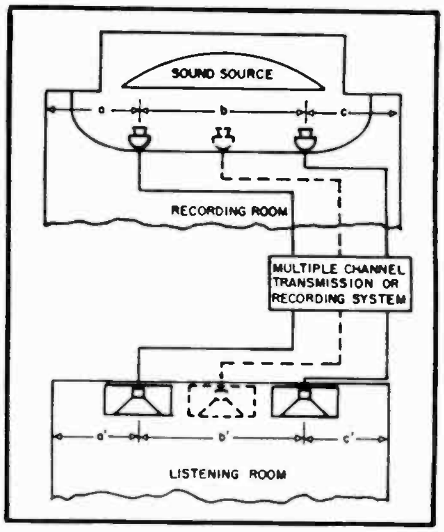

Fig. 3. The true stereophonic system. When three channels are used, distance becomes greater than when only two channels are employed.

advantages. Also, we are not too much concerned with the recording studio acoustics, since our built-in analyzer (brain) compensates for almost all of the acoustic shortcomings of that room, such as excessive reverberation, unwanted echo, etc. This we have learned to do unconsiously since earliest childhood. We use here our binaural sense in the proper and usual manner.

But now, since we wish to reproduce this sound over speakers, we must take the listening-room acoustics into consideration. In fact, the listening room becomes an extension of the recording room. Of course, this has always been true in the case of a one-channel system, but not nearly as obviously so as it now is in the case of the two- or three-channel stereophonic system. After all, we are now trying to re-create that special effect in the listening room which existed in the concert hall.

Let us suppose for a moment that we wished to reproduce a concert, by means of loudspeakers, in the same hall as that in which it had been previously recorded. In reality we should have an infinite number of speakers arranged across the stage and several layers high in order to reproduce a similar full wave front. But for the sake of economy we use only two (or three) and spread them, one to the left, one to the right (and preferably another in the center). If we space two of them at, say, the one-third points across the stage as in *Fig.* 3 where $a=b=c$), then, by proportionality, the microphones used to pick up the original sound should have occupied the same positions now occupied by the speakers. Sound is then radiated from the points where it was originally picked up. This is a most important factor, a factor relating to the theorem of proportionality. The recording system has merely delayed the element of time. The speakers then deliver approximately the same acoustic pattern into the room that was picked up at those points by the microphones. Sound from all over the stage reaches the various

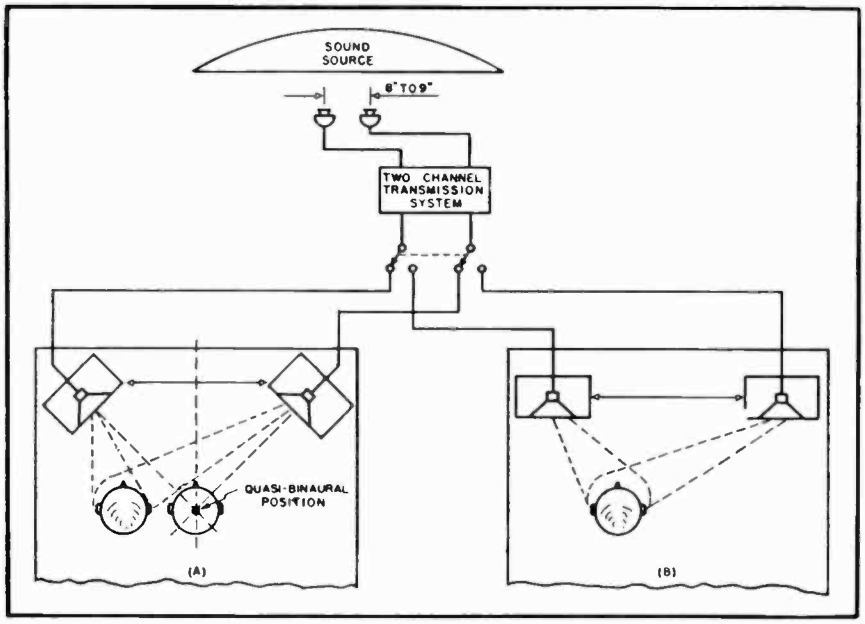

Fig. 2. "Bistereonauralphonic," or mixed and meaningless system.

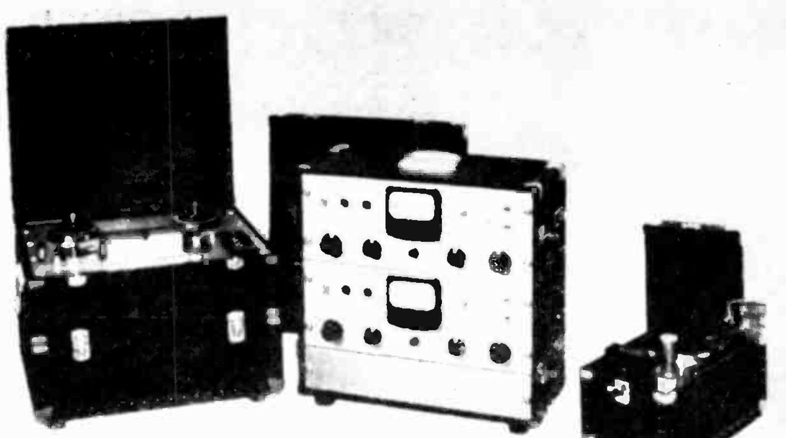

Fig. 4. A complete portable stereophonic recorder, with amplifiers and power supply.

microphones at varying sound levels and times. These factors are necessary to the proper re-creation of the speaker-reproduced pattern.

If the microphones had been spaced "ear distance" (i.e. about 9 inches apart), as was done in a recent demonstration, and the loudspeakers a similar distance apart to maintain proportionality, rather a neat trick which was not done. we would have had in effect a one-channel system. This would accomplish little toward our ends. But with the speakers spaced widely, we lose the necessary proportionality and achieve only our unrealistic "bistereonauralphonic" system. Conclusion: the mikes should be spaced as far apart for this type of recording (stereophonic) as the speakers are to be spaced. We must know beforehand what sort of reproducing system is to be used.

Next we add the problem of placing the speakers not in the original room where the sound was recorded, but rather in some other and usually much smaller room, perhaps a living room. whose major dimension is less, perhaps than the distance between the recording microphones. Should we, therefore, reduce the spacing of the mikes? Experiment says no. Experiment also shows that if we space the speakers in approximately the same general lateral arrangement in the smaller room as in *Fig.* 3, where $a' = b' = c'$, maintaining approximate proportionality, the results are satisfactory. In fact, the sensitive central plane of "binaural" position, referred to previously, disappears and the auditor is free to move anywhere within the listening room. He will experience a similar sensation as though he were to move to various parts of the original auditorium. Placing the speakers in the corners of the room facing diagonally inward or placing them close to side walls appears not to work as well as spacing them a little way in from the walls and square with the room, as in *Fig.* 3. In so doing apparently a more normal sidewall reflection and multiple image pattern is set up in the room, thus simulating more closely the recording set-up. This helps blend the sound pattern in the room. Corner speakers do not set up such an image pattern.

In general, it is necessary to know where and in what manner the sound is to be reproduced before the proper microphone placement can be stated. For headphone listening, mikes should best be spaced a person's head-width apart. A bag of sawdust between the mikes, simulating one's head. is of unknown

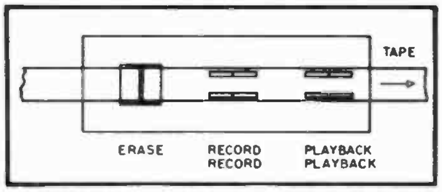

Fig. 6. Diagram of the arrangement of the Ampex stereophonic head assembly.

value. For loudspeaker listening the acoustics of both the recording and listening rooms must be taken into consideration without fail. A wide spacing of the mikes is indicated so that the theorem of proportionality will be maintained. When two channels are used, the mike spacing and position should be chosen with respect to the group size and arrangement so that all sections of the group will appear balanced in the reproduction. An overly wide spacing of the mikes will leave a "dead" spot in the pickup between the mikes (and speakers) at the center of the group. This is where the third channel, in the center, becomes helpful by really rounding out the stereophonic system. The Bell Labs experiments showed that three channels were an optimum minimum number, but that two would work satisfactorily, if properly handled.

The insertion of a dual AM and FM simultaneous broadcast radio link does not change any of the conditions set forth, obviously, but does limit the system to two channels. Separate-sideband AM transmission plus FM has been suggested for a three-channel system, but appears to be still in the future.

It should be repeated that the quality of the total system will limit the naturalness of the reproduction. This was clearly demonstrated at the November, 1952, concert of the University of Illinois Symphony Orchestra, conducted by Leopold Stokowski, at Urbana, Illinois, where the author was invited to transmit the concert stereophonically to an overflow crowd in a relatively large but acoustically different auditorium on another part of the campus. Stereophonic recordings were made simultaneously. During the rehearsal two cardioid mikes were tried, followed by two modern condenser mikes. The former, designed many years ago, have an increasingly attenuated response above 9,000 cps. The condenser mikes have good high end response. Several of the musical faculty present commented on the noticeable difference, and the more nearly true string and oboe tone resulting from the switch. No other elements of the system were changed.

One controversial point might be raised here. What mike pickup pattern should be used? The author prefers a cardioid pattern, but with a wide frequency response. This pattern seems to help stereophonic reproduction because the pickup thus is one-sided, while the speaker reproduction is other-sided. This would seem to help in making the listening room an extension in fact of the recording studio.

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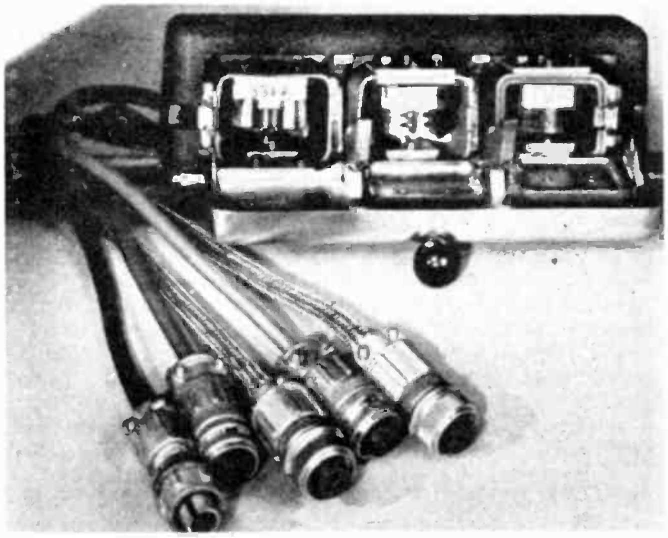
Fig. 5. Stereophonic head assembly, employing a full-track erase head at the left, a two-track record head, and a two-track playback head at the right.



Stereophonic Reproduction

JAMES MOIR*

A basic discussion of the reasons behind our ability to locate sounds simply by hearing them, together with practical requirements of stereophonic sound systems.

THERE IS LITTLE DOUBT that the best examples of current sound reproducer equipment meet most of the known criteria for a high quality monaural system and in consequence there does not appear to be a great deal of opportunity for further improvement in subjective quality if present techniques are merely subject to greater refinement. For example an amplifier having a frequency characteristic flat to .01 db and a distortion content below .01 per cent is not subjectively better than an alternative design flat to 0.1 db (or even 1 db) and a distortion content of 0.1 per cent. In spite of this state of (pseudo) perfection no competent critic would consider that the best possible monaural reproduction of anything but a soloist could be mistaken for the real thing, and until we can deceive most of the people for most of the time there is room for improvement in techniques.

The pleasure derived from listening to a live orchestra is compounded of many factors, most which are adequately dealt with by a laboratory type of monaural reproducer system, but if we are to have a *perfect reproduction* of the original there are many marginal factors that require further attention. An orchestra generally occupies a platform 60 to 100 ft. wide and 20 to 30 ft. deep and this spatial distribution contributes to the pleasure derived from listening. There is merit in mere size. An orchestra that makes use of all the instruments all the time is flat and uninteresting and is rarely employed. Instead the listeners interest is excited and maintained by constant change in the prominence given to the various instrumental combinations. Thus the centre of the listeners' interest moves about the stage, the remaining instruments forming a pleasant but unobtrusive background to the focal zone on which the listeners immediate interest is concentrated.

Reduction of Source Width

A monaural reproduction is completely unsatisfying in this respect, the whole of the 100 by 30 ft. source being compressed and strangled to emerge from an 8- or 10-in. diameter hole, with the result that there is no possibility of identifying or appreciating the individual instruments or sections of the orchestra on the basis of their spatial distribution. The pleasure to be derived from the movement of the sound source is irrevocably lost.

A similar result is obtained from a monaural reproduction of the normal movement of actors about a set or stage. Movement in depth is moderately well reproduced if the reverberation conditions are satisfactory but all movement across the stage is reproduced as movement in depth. Thus all the action appears to take place in a tunnel with a microphone at its mouth.

Current microphone techniques aim to hide this defect by such procedures as employing a microphone boom to support the microphone just out of the picture and over the head of the speaker—the standard film and TV practice.

A monaural system is at a further disadvantage in that a single microphone is unable to discriminate against room noise or reverberant sound, with the result that all recording and broadcast studios must be acoustically treated to obtain a reverberation time much below that known to be optimum if a tolerable result is to be obtained with a monaural reproducer system. Similarly, noise from the audience coughing and shuffling appear to be enormously enhanced when heard over a monaural system. A simple but remarkably satisfying demonstration of the magnitude of this effect can be obtained by anyone with normal hearing and access to a hearing aid of the normal monaural type. Conversation that is easily understood and appears to be without any noticeable background when using two ears, appears against a marked background of reverberation and other room noises requiring considerable concentration if it is to be understood when the monaural hearing aid is used. Kock has shown that the human hearing mechanism automatically discriminates against noise when it approaches the head from a direction

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BINAURAL OR STEREOPHONIC?

(*from preceding page*)

Something should be said about suitable tape recording equipment. *Figure* 4 shows a typical two-channel portable recorder. One case contains the tape transport mechanism another contains two separate record and simultaneous playback amplifiers, one set for each of the two tracks. The third contains the power supplies for the amplifiers. The erase/bias oscillator, mounted in one amplifier, supplies bias directly to one channel and through a buffer amplifier to the other, so that the same bias frequency is used in each recording head, thus eliminating the beat frequency which would result from using two separate oscillators of approximately the same frequency. *Figures* 5 and 6 show the head assembly with a full track erase head, two side-by-side record heads which record two simultaneous tracks, running parallel along the length of the tape, and two appropriate pickup heads. Tape motion is from left to right across the heads. Thus the tape is first erased, then recorded upon, and then monitored a fraction of a second later over the playback system. The record heads are well shielded between the two and have 50 db attenuation on cross-talk between channels. The gaps of both the record and playback structures are critically aligned in parallel so that the time error between the two separate tracks is less than .00003 sec. at 15 in./sec. tape speed, or .00006 sec. at 7½ in./sec. This has an interesting sidelight. Tapes recorded stereophonically (or binaurally) on this type of inline gapped head may be played back on a standard full-track reproducer in an entirely satisfactory one-channel manner. This makes it possible to produce pre-recorded stereophonic tapes which may be played on either a stereophonic or a standard playback machine.

It will be noted that binaural headphone listening to orchestral music, for example, appears to give a much greater sensation of a completely "new" way of listening, and that loudspeaker stereophonic listening (correctly done) is not as astonishing. Headphone binaural listening is best for many special investigative problems. But speaker reproduction should sound more natural than the headphone method because we are used to being in a room-acoustic environment. In fact, it is for this very naturalness that we search.

The author is indebted to his early teachers in acoustics, his many customers, his associates, and several midwest university musical organizations for background and experimental set-ups which have led to the conclusions set forth above. Some of the conclusions reached might be somewhat controversial. Much three-dimensional acoustics still is. As one acoustican has said: "In no branch of science is the theory so simple and the measurement so difficult."

which differs from that taken by the desired sound. This binaural discrimination in favor of the wanted sound amounts to as much as 10 to 15 db, and is entirely lost when a monaural system is used.

A further and somewhat unsuspected result of this binaural discrimination is a marked increase in the clarity of speech and an apparent decrease in intermodulation distortion when reproducing music from a large source such as an orchestra. In fact it gives to the reproduction of a large orchestra the degree of clarity characteristic of a small orchestral combination.

It is suggested that Chinn's preferred frequency range tests using reproduced program material cannot be compared with Olson's tests using live material as in the latter tests the audience were listening "binaurally." The difference in conclusions largely represents the difference between monaural and binaural reproduction.

It is apparent that at the present time few of the advantages of a stereophonic system can be expressed numerically and recourse must be made to expressing opinions until such time as we have some system of indicating the overall subjective appeal of a reproducer system. A number of organizations have worked on the problems of stereophony both in Europe and in America and have recorded their opinions on the advantages of stereophonic systems.

Thus, tests by Bell Labs indicated that a stereophonic system having an audio bandwidth (per channel) of 3750 cps was considered by an audience to have the same aesthetic appeal as a monaural system 15,000 cps wide. J. P. Maxfield of Bell Labs has stated "I would rather hear a two channel reproduction flat to 6,000 cps than a single channel system flat to 15,000 cps; it is more pleasing, more realistic, more dramatic." This is a concise indication of the writer's opinion after more than a year's work on the subject, with the opportunity of hearing the systems developed by three of the leading European concerns.

Explanation of Stereophony

Without further discussion of the virtues of a stereophonic system, the mechanism of the stereophonic effect will be discussed as this is probably the best approach to an understanding of the technical requirements of a stereophonic system.

Mother Nature has provided us with two ears and these enable us to sample the sound field at two points spaced apart by the width of the head. From the differences that exist at the two ears and with the benefit of long experience, the ear - nervous system - brain combination can estimate the position of a sound source with remarkable accuracy, giving the same three-dimensional significance to the acoustic environment as the possession of two eyes gives to the visual environment.

For any sound source not in the median plane the sound at the left ear will differ from the sound at the right ear in three major respects.

1. Referring to *Fig.* 1 it will be seen that a sound from a source on the right side will strike the right ear before the left ear, the time difference being a maximum when the source is on one side and in line with the two ears. In this position the time difference is .00063 sec corresponding to an ear separation of 21 cm.

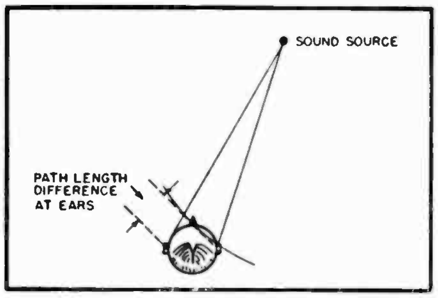

Fig. 1. Difference in path length and time of arrival when sound source is not in median plane.

2. There will be an intensity difference at the two ears, this difference being a function of the frequency of the sound. The intensity difference is frequency dependent as it is mainly the result of diffraction round the head.

3. Most everyday noises have a complex frequency spectrum and as the diffraction losses are a function of frequency the frequency spectrum at the two ears will also differ.

These differences justify further discussion. The reason for the difference in time of arrival at the two ears is evident and requires no further explanation, but the question immediately arises as to which part of the sound-wave cycle is accepted by the ear as determining the time of arrival at that ear. On an impulsive sound having a steep wave front it may be assumed that the arrival of the wave front is recognized, but on a repetitive waveform there is difficulty in understanding just how the ear recognizes the difference between successive cycles with identical waveform. A high-frequency wave passing from right to left will have several cycles pass the right ear before the first cycle reaches the left ear, and the right ear may not know just how many cycles have passed at the instant the first cycle reaches the left ear. This rather suggests that there may be difficulty in fixing the position of a high-frequency source having a frequency such that more than half to one cycle of the wave can be accommodated in the space between the ears. Taking the velocity of sound as 33,000 cm/sec and the ear spacing as 21 cm. it might be expected that frequencies above 800 cps (half wave = 21 cm) might present difficulties in location and it is worth noting that this is found to be the case in practice.

Using a very elegant test technique, Galambos has taken simultaneous photographs of the sound waveform and the nerve response that results from the sound, and these show that the nerve discharge always occurs at the first positive peak. This suggests that the brain has adopted the first positive peak as a reference point and that in measuring time intervals to fix the position of a sound source in space, it notes that interval between the first positive peak arriving at the left and right ears. This process must be repeated at fairly frequent intervals if a moving source is to be followed and it is suggested that the intervals between syllables might well form the convenient gaps from which to commence each new measurement of "time of arrival" difference.

The loudness difference at the two ears is mainly due to the presence of the head between the ears. Any obstacle placed in a sound field distorts that field, producing an increase of pressure on the side facing the oncoming sound wave and a decrease in pressure on the reverse side—a process known as diffraction. The pressure difference between the two sides is a function of the ratio of obstacle diameter to wavelength of incident sound, and for a given size of obstacle will increase as the frequency of the incident sound rises (i.e., wavelength falls). An exact calculation of the field distortion is a problem of great difficulty but as we are only interested in diffraction around a human head, Weiner's measured results are satisfactory. These are shown in *Fig.* 2 and

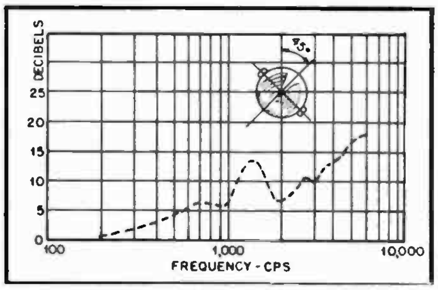

Fig. 2. Ratio of sound pressure (in db) at ears for an angle of 45 deg. to the source.

indicate that the pressure difference at the two ears has risen to 8 db at 1000 cps and 17 db at 6000 cps. Pure tones are not of significant importance in everyday life where the usual sounds—such as speech, music, and noise—have energy scattered throughout the whole frequency spectrum. On a complex sound the resultant pressure difference at the two ears will obviously depend upon the frequency spectrum of the energy in the sound. Steinberg has calculated the pressure difference and hence the loudness difference for normal speech and his results are shown in *Fig.* 3 from which it will be seen that up to angles of 40 deg from the median the loudness difference in db is almost directly proportional to the angle turned through by the head.

The third major difference noted is that as the diffraction effects are a function of frequency, with a resulting difference in the frequency characteristic of a sound at the two ears. With experience the brain may be able to use this difference to provide a clue to localization.

For sounds originating in the median plane all these differences vanish as the

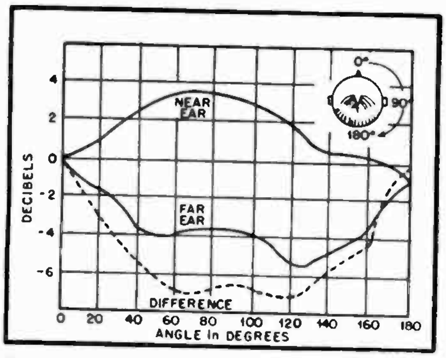

Fig. 3. Variation in loudness as a speech source is rotated in a horizontal plane around the head.

source is then symmetrically disposed with respect to the two ears, a condition that holds at all angles of elevation. It is to be expected that the precision of location in the vertical plane would be poor and in practice this is found to be the case. Discrimination between back and front is also found to be poor unless the head is free to make some exploratory movement. The slightest rotation of the head provides adequate discrimination between front and rear, presumably as a result of the brain noting the direction of the change in the time differences at the two ears.

It will be seen that there are three main differences in the sounds at the two ears—a time of arrival difference, a loudness difference, and a frequency-characteristic difference. At present there is no conclusive proof as to which of these provides the real clue to localization in practice. It seems highly likely that all three make contribution, with time and loudness differences providing the major clues.

Practical Stereophonic Systems

Consideration can now be given to methods of achieving stereophonic reproduction via an electrical reproducer system, knowing that we must maintain the time and amplitude differences present at the pickup points in the studio. There are two approaches to this problem, the first that of taking two samples of the sound field and transferring these two samples to the remote listener's ears through completely separate electrical systems and two headphones as shown in *Fig.* 4. Two entirely separate systems are obviously necessary as the left- and right-ear signals cannot be allowed to contaminate each other. At the transmitter end the sound pickup consists of two high-quality microphones mounted in a space model of the human head to simulate the acoustic field distortion produced by the head in practice. After amplification the signals are conducted by two separate channels to the two earphones. The results of this are extremely impressive but the necessity of wearing headphones militates against its use and it would appear unlikely to find favor with the general public unless the headphone cords are dispensed with. This could be accomplished by introducing two local (domestic) low-power radio transmitters with miniature receivers mounted on the headband—a procedure fairly common in film and TV studios for transmitting instructions to the operators of mobile cameras monaurally.

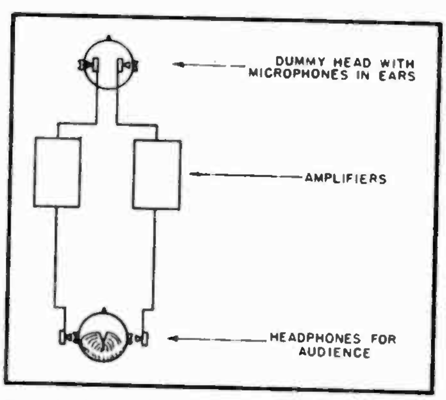

Fig. 4. Stereophonic reproduction with headphones.

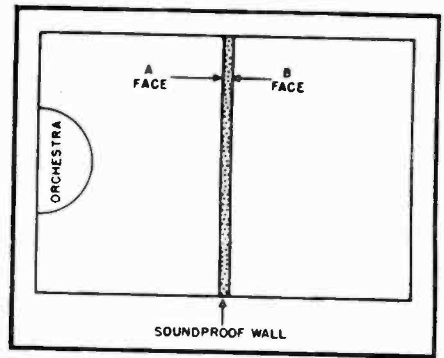

Fig. 5. A wall can be rendered sound transparent if face A is covered with microphones connected through individual amplifiers to loudspeakers on face B.

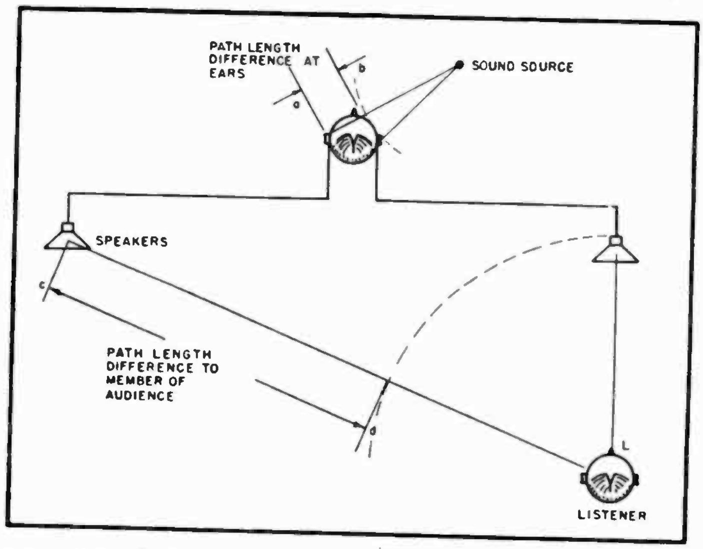

Fig. 6. Path-length discrepancy using closely spaced microphones and two-speaker system.

If the listener's ears cannot be transported to the studio it is possible to adopt the alternative approach and transport the acoustic field to the listener. The principles involved will be understood by referring to *Fig.* 5 showing a long hall divided into two separate sections by a soundproof transverse wall at A. This wall can be rendered sound transparent (unidirectionally) by covering the A face with a large number of microphones each connected through an amplifier to a loudspeaker on the B face of the wall. Any sound field approaching the A face would be reproduced on the B face and the audience in the B section would be unaware of the dividing wall. While this procedure would be reasonably effective it is commercially impractical, as a separate line and amplifier are required between each microphone and its associated loudspeaker. Some means of reducing the number of microphones and loudspeakers is required.

In the vast majority of stage plays and most films the action is largely concentrated at ground level and as localization in the vertical plane is rather poor in any case it seems reasonable to assume that the "vertical" information need not be transmitted. This eliminates the need for all the loudspeakers except the bottom row, a very considerable simplification. With present techniques two or three separate channels are all that can be accommodated on tape, disc, or film, so it appears advisable to check the performance of two- and three-channel systems.

In a simple two-channel system, localization is weakest in the center just where it is desirable that it should be at its best and the addition of the third microphone and speaker bridged across the two outer channels improves the performance in this respect, though none of the combinations tried are as good as the three-channel layout.

Trial Attempts

There have been various attempts to combine the advantages of the dummy-head microphone mounting with the use of loudspeakers, though so far without any outstanding success. From the point

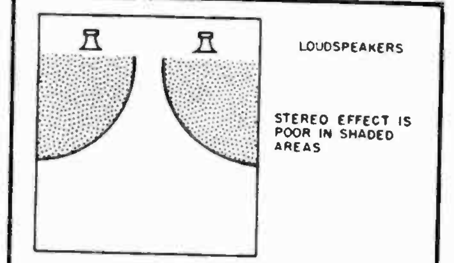

Fig. 7. When close-spaced microphones are with loudspeakers, the stereo effect is limited to the clear area.

of view of microphone technique the dummy-head mounting has the great advantage of simplicity, two microphones in a common mount being no more difficult to handle than a single microphone. The limitations appear when consideration is given to the time differences which result in seating positions off the room axis. Referring to *Fig.* 6 it will be seen that at the microphone the position of a sound source is characterized by a time difference proportional to the path-length difference, and this holds for all positions on a line midway between the loudspeakers. At a point such as L well off the center line the original time difference *a-b* is completely swamped by the time difference *c-d* which is characteristic of the relative positions of the listener and the loudspeakers, and has no relation to the position of the source. It might be expected that the stereophonic effect would not be quite so obvious in position off the axis of the room and this is found to be the case in practice. *Figure* 7 is indicative of the area in which effective stereophony is obtained. Though this appears to be a fundamental weakness of all systems using close-spaced microphones, the advantages of simplicity in the microphone technique justifies further work on the system.

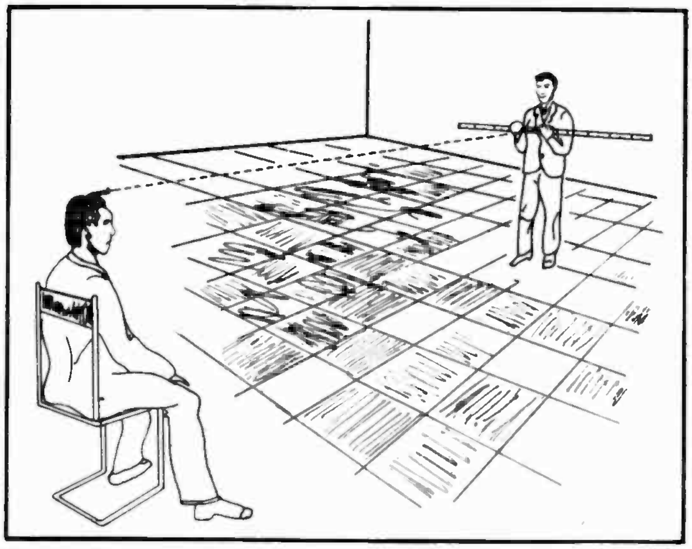

Fig. 8. Method of error measurement in sound location tests.

Figure 8 indicates the technique used in all the accuracy tests. A caller held a horizontal stave marked off in each direction from the center, and stood about 25 ft. from the subject under test. The subject wore a light head harness carrying a horizontal sighting rod which was visually lined up on the center zero of the stave at the commencement of the test. The caller took up a position on a 25-ft. radius from the blindfolded subject and read passages from a book until the test subject indicated that he was "on target" at each position. A third assistant read off the error in position on the reader's stave, along the sighting rod on the subject's head. This procedure was repeated at ten different angular positions with five different subjects and the errors averaged. For the indoor tests the same group of observers repeated the procedure in a small theater, the listening position being 55 ft. from the caller. The results of the first series of tests are set out in Table I and it will be noted that accuracies of the order of 1 deg. can be achieved both indoors and outdoors. A comparison with the errors made when attempting to align the eyes on the stave zero showed that the accuracy of visual alignment is not more than twce that of an aural alignment. This is surprising, as the ears are rarely used for the purpose of position fixing, a duty left to the eyes.

During the indoor tests the opportunity was taken to check the effect of frequency-range restriction on the accuracy of location. The results also serve to indicate the information carried by the various regions of the frequency spectrum. The technique remained as in the previous tests but the source was replaced by a high-quality loudspeaker reproducing speech picked up by a microphone in another studio. Three filters having the characteristics shown in *Fig.* 9 were used to define the frequency range in the three tests, the results being shown in Table II. It will be seen that the majority of information on source position is carried by the frequency components above 500 cps though the ear can find adequate information in the small amount of energy that remains in the region above 3000 cps or below 500 cps. With the male speakers employed in these tests, speech intelligibility was zero when the band below 500 cps or the band above 3000 cps was being used.

In tests we have noted that dynamic localization (the localization of a moving sound source) appears to be appreciably more accurate than the localization of a stationary source.

The final point to be considered is the technique to be used in obtaining two or three channels between studio and reproducer, probably the most difficult problem of all. For ¼-in. magnetic tape, two solutions are fairly straightforward, the first being the use of two or three separate tracks on the same tape. Double track tapes are currently available and can be used for a two-channel system without difficulty. Three tracks each .05 in. wide are possible, though agreement will have to be reached on the relative position of the three tracks, as it is probably impossible to mount three recording or scanning heads side by side. Earlier it was noted that two channels 6,000 cps wide, or three channels 4,000 cps wide, gave results which were subjectively assessed as being equal or better than a monaural channel 15,000 cps wide, and in consequence it becomes necessary to consider how best to use a medium such as tape which can deal with a frequency band perhaps 20,000 to 25,000 cps wide. Nothing is to be gained by reproducing frequencies above 15,000 cps, and a more pleasing result can certainly be obtained by dividing the band into two or three separate channels each 6000 or 7000 cps wide and using them for a stereophonic system.

Phonograph records present a more difficult problem and as they may be superseded by tape it may never be necessary to solve that problem, but one neat solution due to A. D. Blumlein of E M I is worth noting. Blumlein proposes to record a two-channel stereophonic signal on an ordinary disc by simultaneously using lateral and vertical modulation of the single groove. Equipment was built and succesfully demonstrated that this was possible. E M I have demonstrated 78 r.p.m. recordings flat to 20,000 cps and these would permit two or three channels 6000 to 9000 cps wide as suggested for tape.

Broadcast-band AM radio poses on

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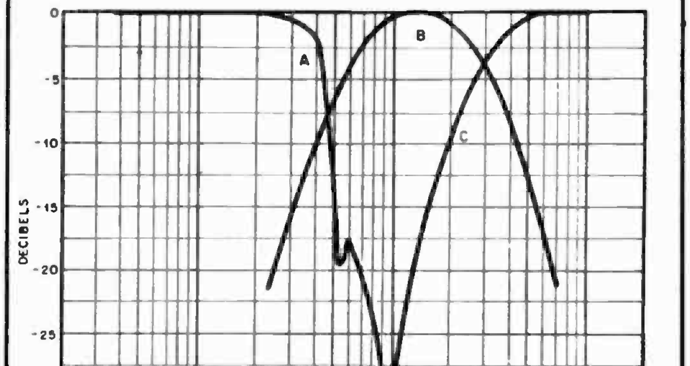

Fig. 9. Filter characteristics. (A) low-pass filter; (B) band-pass filter; and (C) high-pass filter.

A Discussion of Dividing Networks

J. P. WENTWORTH*

Presenting two simple dividing networks—one of which provides a connection point for negative voltage feedback which will equalize over-all amplifier-speaker response.

IT IS GENERALLY ACKNOWLEDGED that a single loudspeaker unit can not reproduce optimally the entire audible frequency range. For clean reproduction of high-fidelity music, at least two speakers are required, each transducing signals in a limited frequency band. To insure that each speaker receives only signals within its own frequency range, it is customary to use a dividing network or networks to distribute the signal among the speakers.

Perhaps the neatest way to accomplish such signal distribution would be to separate the frequency bands at a low power level, and to use an individual power amplifier for each speaker. However, since the output stage and transformer represent a considerable part of a high-quality amplifier, in terms of cost, power consumption, and weight, it is usually more expedient to make the separation at the secondary side of the output transformer.

An ideal multispeaker system would give the impression that the sound emanated from a single super-speaker, which was able to handle all frequencies To accomplish this aim, the network should satisfy the following criteria:

1. The network must so distribute the

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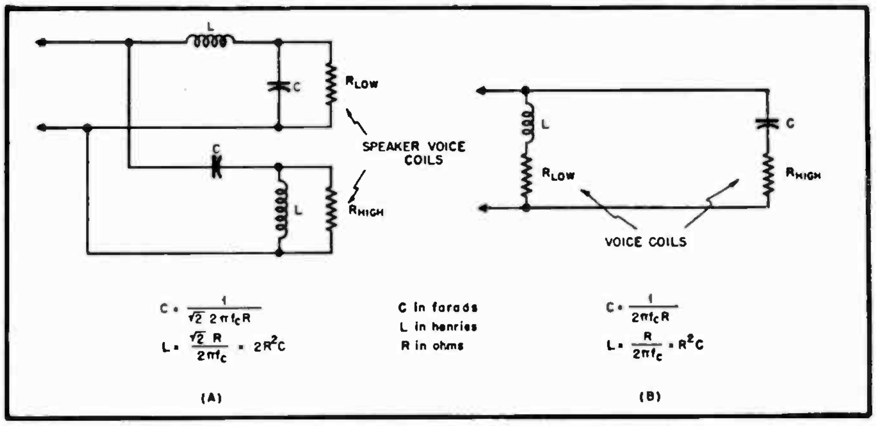

Fig. 1. (A) Parallel network, constant resistance type, giving a cut-off slope of 12 db per octave; (B) Half-section network giving cut-off slope of 6 db per octave.

STEREOPHONIC

(*from preceding page*)

even more difficult problem, particularly in Europe where all stations are on a nominal spacing of 9 kc. Single sideband transmission of two channels is possible as the data in Table II shows that frequencies below 500 cps are of no great importance to the stereophonic effect, permitting a double-sideband transmission of frequencies up to this point, with single-sideband transmission of the left and right signals on the upper and lower sidebands, as shown in *Fig.* 10.

On short waves there is no technical difficulty. The audio bandwidth of a short-wave transmitter is sufficient to provide two or three channels as suggested for magnetic tape, or the carrier may be simultaneously modulated by FM and AM signals to give two channels.

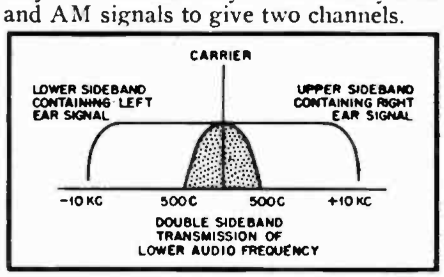

Fig. 10. Radio bandwidth usage for stereophonic transmission on AM.

The renewed interest that has been shown both in Europe and America in the problems of stereophony indicates that it will be the next major step towards obtaining a *perfect reproduction*. Under the stress of competition from TV, stereophonic sound may appear first in sound movies and may be accompanied by the introduction of a wider screen to take advantage of the new freedom.

TABLE I

Comparison of the Accuracy of Location in Indoor and Outdoor Environments

Sound Source—Original Male Speech

	Mean Error	Standard Deviation
Indoors at a spacing of 55 feet	1.04 deg.	1.15 deg.
Outdoors at a spacing of 25 feet	1.2	2.7

TABLE II

Accuracy of Location using Bands of Filtered Male Speech

Frequency Range	Avg. Error	Std. Deviation
50-500 cps	3.8 deg.	3.55 deg.
500-3000	0.9	3.8
3000-7000	0.5	3.4
50-7000	0.7	4.7

REFERENCES

1. W. E. Kock, "Binaural localization and masking," *J. Acous. Soc. Am.*, Nov. 1950.
Details of an experimental determination of the amount of discrimination against noise that is given by our binaural hearing system.

2. Harvey Fletcher, "Hearing, the determining factor in high-fidelity transmission." *Proc. I.R.E.*, June 1942.
An excellent paper which reviews most of the factors known to be of importance in obtaininig a high-fidelity reproduction. This paper is important to anyone interested in the subject of high-fidelity.

3. Galombos and Davies, "The response of single auditory fibres to acoustic stimulation." *J. Neurophysiology*, 1943, No. 6.
Describes nerve voltage measurements that appear to have an important bearing on the mechanism of the stereophonic effect. Most interesting reading.

4. F. M. Weiner, "Diffraction of a sound wave by the human head." *J. Acous. Soc. Am.*, Jan. 1947.

5. J. C. Steinberg and W. B. Snow, "Auditory perspective physical factors." *Elec. Eng.*, Jan. 1934.
One of a group of six papers devoted to stereophony. That loudness difference is the main factor in binaural localization is not now so widely believed as in 1934, but the group of papers is required reading.

6. J. Moir, "Stereophonic sound," *Wireless World*, March, 1951.

7. J. Moir and J. A. Leslie, "The stereophonic reproduction of speech and music." *J. Brit. I.R.E.*, 1951 Radio Convention.

8. K. de Boer, "*Stereophonic sound reproduction*," Philips Tech. Rev., April, 1940.
An account of the factors that are of importance.

signal among the speakers that the acoustic power output at every frequency is the same as that radiated by the single hypothetical ideal speaker.

2. All frequencies should be radiated with the same phase relationships as those that would exist in the case of the single ideal speaker. This objective is a rather minor consideration in minimizing phase distortion; however, if inverse feedback voltage is to be taken from the speaker voice coils, it becomes a requirement of paramount importance.

3. The speaker system should present to the amplifier a constant and purely resistive impedance, in order to minimize distortion in the output stage of the amplifier.

4. In the interest of economy of power, it is desirable that the network be non-dissipative; i.e., that it be made up of purely reactive impedance elements.

5. In order to provide effective damping, the source impedance seen by each speaker should be as low as possible.

It is evident that conditions (1) and (2) will be implicitly satisfied if requirements (3) and (4) are met.

Two networks that will fulfill requirements (1) through (4) are shown in *Fig.* 1, where R is the impedance of the speaker, and f_c (the "crossover" frequency) is the frequency at which the radiated power is equally divided between the two speakers. The input impedance of either of these two networks is equal to R, and is independent of frequency, if the following conditions are met: (a) the impedances of the speakers must be equal, and (b) the impedance of each speaker must be a pure resistance.

Circuit A of *Fig.* 1 provides a considerably sharper frequency division of the signal than does circuit B; the power radiated by each speaker of circuit A falls off at a rate of 12 db per octave in the cut-off band, whereas the slope for circuit B is 6 db per octave. Which of these circuits is the more desirable depends on several factors. Use of the sharper slope, for instance, reduces the range of frequency over which any one speaker has to handle an appreciable amount of power. However, unless great care is exercised in locating the speakers, this same sharpness of crossover may destroy the illusion that the sound is being radiated from a single source, since the fact that the sound is being produced in two frequency bands is more obvious under these conditions.

Relative Advantages

From the point of view of the hobbyist, circuit B offers certain advantages over the more complicated circuit A. It is true that a saving in complexity of the circuit is probably of negligible importance, *per se*; and certainly any reduction in size and weight of the circuit will be a small percentage of the volume and weight inherent in the speakers themselves. However, there is a material saving in the cost of the components required—circuit B requires only about 36 per cent of the inductance, and some 70 per cent of the capacitance required by circuit A for a system having the same crossover frequency and the same speaker impedance. Since the capacitors used in a high-fidelity system must be of the high-quality, oil-filled variety, and since the values of capacitance required in a practical system are quite large, a reduction by 30 per cent in capacitance represents a very real reduction in cost. Moreover, use of a smaller inductance will result in a lower resistance in the inductors, with a slight resultant gain in efficiency, and with a concomitant improvement in damping.

In circuit B, any resistance in the inductor winding can be written off as part of the speaker resistance, and can easily be balanced by addition of a padding resistance in the other half of the circuit. However, a reasonable amount of resistance unbalance can be tolerated, except when feedback is used from the voice-coil windings. In the latter case, phase shift may become excessive if balance is not maintained.

Similarly, inductance in the speaker winding can be regarded as part of the inductor in one branch of circuit B. Unfortunately, however, since this speaker is carrying the low-frequency components of the signal, the inductance in this speaker is of less importance than it is in the other speaker, which is not compensated by the circuit configuration. By application of Thévenin's theorem, one may see that it is possible to regard shunt capacitance around the high-frequency speaker as included in the series capacitor, although this capacitance should be negligible, even at the highest audio frequencies.

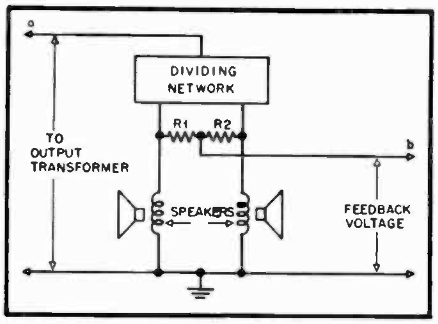

Fig. 2. Circuit arrangement for obtaining feedback voltage from voice coils of two-speaker system.

So far, nothing has been said about requirement (5) above, i.e., that the speakers should see a low-impedance source. Unfortunately, both of the above circuits are weak in this respect, circuit B perhaps having a slight advantage over circuit A. Moreover, the source impedance—and hence the speaker damping—will be a function of frequency. If speaker resonance occurs at a frequency at which the damping is ineffective, "muddy" performance is inevitable.

However, a low source impedance can be provided by applying the voltage across the voice coils as an inverse feedback signal. Feedback voltage may be developed by the circuit shown in *Fig. 2*. For "flat" output, R_1 and R_2 should be equal. A certain amount of tone control can be exercised by varying the ratio of R_1 to R_2, thereby vary-

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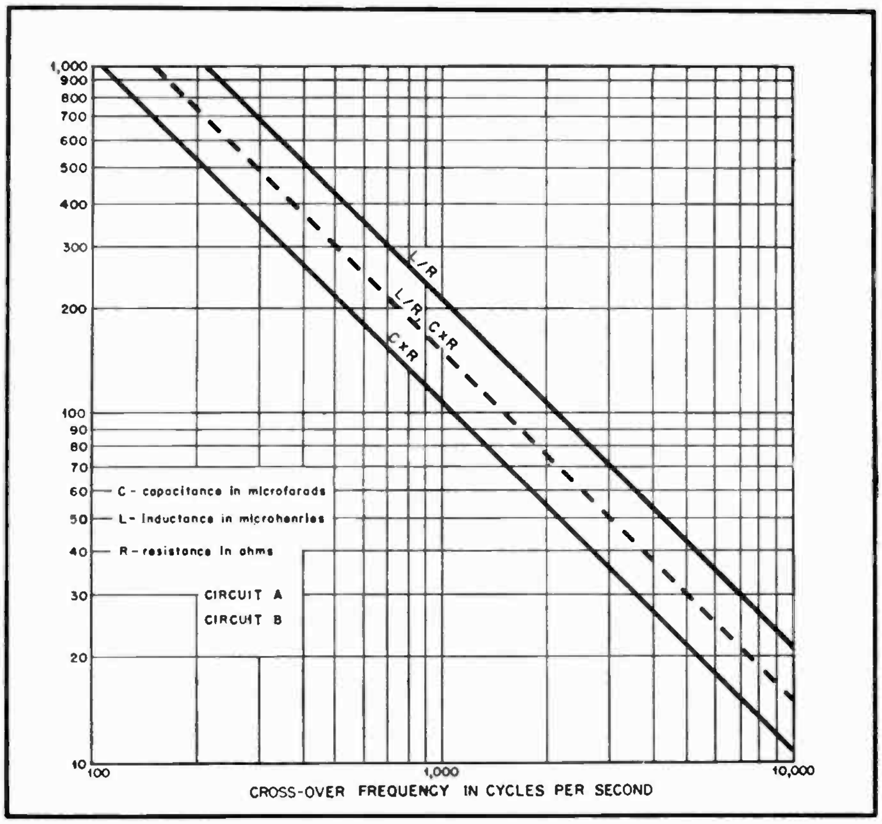

Fig. 3. Chart for determining values for L and C in networks of Fig. 1.

A Transistor Phonograph Preamplifier for Magnetic Pickups

BASIL T. BARBER*

Complete instructions on building a transistor preamplifier with miniscule power requirements, with an analysis of its performance and data on transistor circuits in general for audio use.

RECENT AVAILABILITY of improved types of transistors and their present successful application in the field of communications, computers and guided missiles, indicate that the transistor has at last ceased to be looked upon as a laboratory curiosity and is rivaling in performance many functions that the vacuum tube has monopolized for the past thirty years.

In the audio field, with the exception of a few "transistorized" hearing aids and AM tuners, progress has been slow, at least in comparison with other fields. This was due mainly to the inherent technical limitations of the new item and its scarcity and high cost, rather than the audio engineer's or high-fidelity addict's lack of pioneering spirit. Now that most of the original limitations of the transistor have been either eliminated or greatly reduced, the way should be open for an increased number of transistor applications in audio reproduction.

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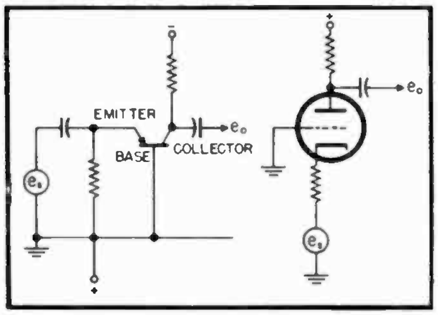

Fig. 1. Grounded-base circuit and its vacuum-tube equivalent.

Following is a brief classification of the three basic types of transistor circuits with their duals in the more familiar vacuum-tube circuit. A word of caution is warranted. Do not begin replacing grids with bases, plates with collectors, and cathodes with emitters! A basic understanding of transistor operation is essential to avoid frustration and disappointment in designing circuits around them. Adequate literature is already available, and although the theory of semiconductors is not likely to be found easy to digest, one does not have to wade through Shockley's "Electrons and Holes in Semiconductors" in order to design simple transistor circuits for audio applications.

Transistor circuitry can be, very broadly, put into three classes, depending on which element is grounded. Functionally, each class performs differently and its selection for a particular application will depend to a large extent on the specific requirements of the application as to gain, input and output impedance, noise level, power output, frequency response, etc.

Grounded-Base. Figure 1 presents the first class. Electronically it is similar to a grounded-grid vacuum-tube amplifier. Input impedances of as high as 1,000 ohms and output impedances of as high as 20,000 ohms are possible. These are comparatively low; it is necessary to use high-value input capacitors and matching transformers if transistors are employed in cascade. Gains per stage as high as 37 db are possible, with matched impedances. There is no 180-deg. phase reversal as in a tube, and two batteries are required, or a tapped one. The principal problem with grounded-base circuits is one of stability, especially if transistors with α, the current amplification factor, greater than 1 are employed. Junction transistors are superior to the point-contact type in this respect.

Grounded-Collector. In *Fig.* 2 we have the grounded-collector circuit, which resembles the vacuum-tube cathode-follower. The voltage gain is less than 1, but can be made as high as 0.95, a loss of less than 1 db. *Power* gains as high as 20 db have been recorded. There is no polarity reversal and external stabilization with bleeders may be necessary. The most interesting feature of the grounded-collector circuit is the relatively high input impedance which, although it can be as high as 1 megohm, depends on the load resistor R_L. Low values of R_L will bring the input impedance down. The output impedance can be made as low as 100 ohms. The techniques of self-bias can be employed as successfully as in vacuum-tube circuits. *Figure* 3 shows a common method of

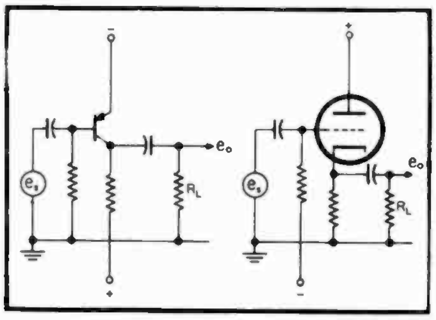

Fig. 2. The grounded-collector transistor circuit.

NETWORKS

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ing the relative outputs of the two speakers.

It is assumed that inductors will be wound especially for the circuit, since it is highly unlikely that the non-standard values of inductance required will be available commercially. The fact that L and C will resonate at the crossover frequency can be used to determine the required inductance. However, this method does not insure the relationship $L = R^2C$ (or $L = 2R^2C$), due to tolerances in R and C, and the voltage relationships in the circuit may deviate seriously from those predicted on a basis of the nominal values. A more satisfactory procedure would be to adjust the inductance to the actual resistance and capacitance values, absorbing the tolerances in these quantities in a slight shift of the crossover frequency. The proper inductance may easily be determined by use of the circuit shown in *Fig.* 2; if R_1 and R_2 are equal, the correct amount of inductance will provide a constant voltage gain from a to b, independent of frequency. Values of L and C, normalized with respect to R, are plotted against crossover frequency in *Fig.* 3.

If circuit A of *Fig.* 1 is used, care must be taken to maintain equal resistance, inductance, and capacitance in both branches of the circuit. R and C can be trimmed with the help of a simple bridge, but cut-and-try seems to be the only feasible way to insure balance of the inductances, at the same time satisfying the condition, $L = 2R^2C$. In this respect, the simplicity of circuit B proves to be a strong argument in favor of the latter circuit.

Either of the two circuits shown in *Fig.* 1 is readily adaptable to systems of three or more speakers. For a three-speaker system, a complete dividing network and its associated speakers may be substituted for one of the speakers indicated in *Fig.* 1—since the network with its speakers presents a constant resistive impedance, the original network will operate exactly as if working directly into a speaker.

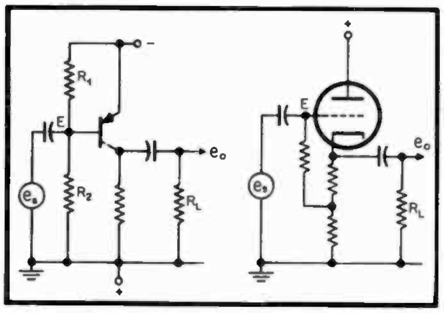


Fig. 3. Grounded-collector circuit can be self-biased in somewhat the same manner as a cathode-follower tube.

self-biasing. E is the bias obtained from voltage divider R_1–R_2.

Grounded Emitter. Figure 4 shows a popular type of transistor circuit for attaining good voltage amplification. Gains as high as 46 db are possible, and the circuit exhibits a 180-deg. phase reversal at its output, making possible the application of negative feedback around it without resorting to transformers. The input and output impedances are higher than the ones of the grounded-base class, but the problem of stability still exists, although it can be corrected with bleeders as before. In *Fig.* 5, self-bias is again employed, eliminating one battery.

Transistors and Audio

With the above brief resumé in mind, an evaluation of transistors can be made in terms of functions and characteristics directly applicable to the audio field. This evaluation is based on the latest information available and is by no means complete or conclusive as, judging from the present rate of improvement, only a short period of time may be required to modify any of the main transistor characterstics completely.

From a designer's point of view the following may well be the most important criteria on which a transistor may be judged for its applicability in audio work.

Impedance. In non-linear elements such as magnetic amplifiers, transistors, crystals, and semiconductors in general, the input as well as the output impedance is low in most cases, resulting in a serious problem of impedance matching, especially if these elements are to be employed in cascade. In telephony and general transmission work with coaxial or 600-ohm lines this becomes a desirable characteristic, but in audio it is likely to be a serious problem, although shielding headaches are minimized. Impedance-matching transformers can be used, but their expense and their introduction of phase shift, distortion, and limited bandwidth make their usefulness questionable. The input impedance of the transistor, usually less than 1,000 ohms, makes necessary coupling capacitors of high value, though they can have low voltage breakdown ratings.

Assume that we have a transistor circuit with an input of 500 ohms. The input coupling capacitor should have a value of not more than 50 ohms at 20 cps for 1 db (10 per cent) attenuation at that frequency, or about 160 µf!

This is a rather impressive value for a coupling capacitor.

Recently available minature tantalum capacitors are ideal for this application, although their present cost reminds one of the black market days of the last war.

Gain. The actual gain will depend on the degree of impedance matching and, depending on circuitry, can be from 17 to more than 40 db per stage, which comes close to the capabilities of most high-gain triodes and pentodes.

Noise. Noise may be the limiting factor of a transistor in audio. Remarkable improvements have been made during the last few months in this respect and the noise level has been reduced from 60 db (at 1 kc) to below 10 db in some types, which compares favorably with a vacuum tube. Junction types have lower noise level than the point-contact types. The noise level is, to some extent, a function of the collector voltage and a lower-power supply voltage will give less noise, but less gain also. Another serious disadvantage of the noise present is that its magnitude is inversely proportional to the frequency, rising at about 3 db per octave. In addition, the input impedance seems to play a role in the amount of noise present and a matched load does not necessarily give the best signal-to-noise ratio.

Frequency Response. Transistor circuits have been designed flat to as high as 10 mc, more than adequate even for that super-ultra-fidelity unit we have in the back of our minds. The average circuit will give a practically uniform frequency response to beyond the audio range. The low-frequency cutoff point will be determined by the value of the input capacitor. Point-contact transistors have better frequency response, but their high distortion and problematical stability exclude them at present from any audio applications.

Distortion. With junction transistors distortion can be kept down to about 2 per cent and with negative feedback values well below 1 per cent are possible.

Dynamic Range. Outputs as high as 3 volts, enough for preamplifiers and equalizers, are easily available. With a unit of low noise level, a dynamic range of 60 db or more can be attained with excellent linearity.

Power. Power requirements should, in all fairness, be considered in terms of the power level of the signals to be handled.

Present transistors can give as much as 0.1 watt maximum power output and this level can be increased to about 1 watt in a push-pull-parallel configuration. In designing preamplifiers and equalizers, however, the question of power should present no problem since only a few milliwatts is necessary. A point of interest is a comparison of efficiency between a transistor and a tube employed as preamplifiers. A transistor requires about 30 milliwatts, while a tube would need about 5 watts,[1] or almost 200 times as much power. At present, 80 per cent of the power used in a transistor amplifier is wasted in dropping resistors required to provide a constant-current supply from a constant-voltage source and for stabilization purposes. As transistors and batteries improve, even the present small power drain will be considerably reduced.

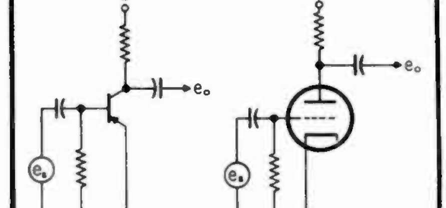


Fig. 4. The grounded-emitter circuit is the dual of the grounded-cathode amplifier.

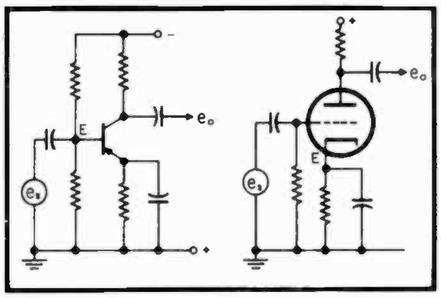


Fig. 5. The self-biased grounded-emitter circuit and the cathode-biased tube are not quite true duals.

Hum. As the transistor needs no heater supply, complete elimination of hum, a stumbling block that has felled many an audio enthusiast attempting to build his own equipment—is a reality. In addition, there is no time delay for warmup after the power switch is turned on.

Microphonics. A transistor can withstand a forced vibration from 5 cps to 5 kc with 100 g acceleration and shock in excess of 20,000 g, with no ill effects on its operation. Although it is hoped that such conditions are not likely to occur too often in a home audio system, these characteristics are desirable if for instance a transistor preamplifier is to be mounted directly under the motor-board of a record player.

Stability. Units having current gain α greater than 1 invariably present a serious problem of stability. With latest models of junction transistors and with the use of external stabilization, this problem has been almost eliminated.

Temperature. Temperature effects are likely to cause some trouble, especially in extreme climatic conditions. In normal surroundings temperature becomes no problem. As the temperature increases above normal, the transistor becomes more noisy and loses some of its gain, but it can be improved with negative feedback. A word of caution when working with transistors: The leads should not be subjected to prolonged contact with a soldering iron, as excessive heat will result in permanent destruction of the unit. The same general precautions should be taken as when working with germanium crystals.

Life. Half-life, determined on a statis-

[1] Filament approximately 2 watts and plate about 3 watts.

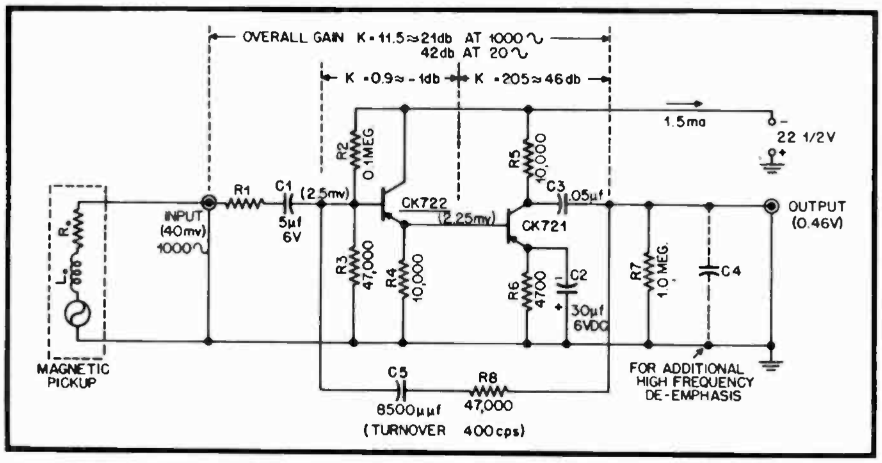


Fig. 6. The transistor preamplifier circuitry is simple and its response—including bass equalization—is excellent. The 22.5 volts may be furnished by the power amplifier bias supply, a small power supply, or a battery.

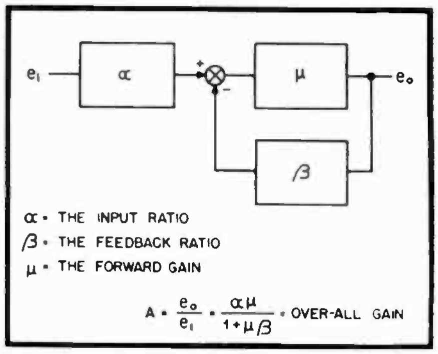


Fig. 8. This block diagram and the formulas show that grain of the transistor preamplifier is calculated in the same way as tube circuits.

tical basis, is about 70,000 hours or close to 10 years of continuous operation. Since the only permanent part of the average high-fidelity system is its chronic susceptibility to modifications (and improvements), the transistor is likely to outlive, if not the owner himself, at least a dozen or so circuits built around it.

Reliability. With the proper power supply, present units seem to exceed the vacuum tube in reliability. Uniformity in production has been a serious problem, but late units are manufactured within ±2 db uniformity in their characteristics, which is even better than the tube tolerances (±3 db).

Space. A transistor is about the size of a coffee bean.

Cost. Transistor cost is high at present, ranging from $5 to $15, but prices will doubtless decrease as demand increases and methods of production are improved. Availability ranges from stock to a few weeks delivery on most types.

The Transistor Phonograph Preamplifier

Taking the foregoing into consideration, it should be therefore possible to design a phonograph preamplifier which will compare favorably with its electronic counterpart. From *Figs.* 3 and 5, it is seen that a combination of a grounded collector and a grounded emitter in a direct-coupled circuit is possible, provided that the appropriate bias levels are supplied. Direct coupling eliminates the need for a large interstage coupling capacitor, and also provides a relatively high input impedance. Peter Sulzer was first to apply this.[2] Although this type of circuitry needs two transistors, one of which supplies no gain, the definite advantages seem to outnumber the cost of the extra transistor, especially with the elimination of a large-value capacitor and the likelihood of employing an input transformer.

Figure 6 shows the final circuit selected. A CK721 is employed as a voltage amplifier, instead of a CK722 giving about 40 per cent higher gain at about the same percentage increase in price. The input impedance, being about 20,000 ohms, makes it necessary to employ a large input capacitor to get down to 20 cps. It should have an impedance, at 20 cps, of not more than 1/10 of the input impedance of the circuit or 2,000 ohms, which means a 4-μf minimum value.

Figure 7 shows the frequency response of the schematic in *Fig.* 6. Noise level, about −39 db, is rather high for our application. A gain of 42 db compares very favorably with most electronic tubes, taking into consideration that the first transistor is actually employed as a cathode follower. Since the complete circuit has a 180-deg. phase reversal, frequency-selective feedback can be applied around it, modifying the frequency response to provide the low-frequency pre-emphasis necessary for any magnetic pickup. In addition, the feedback will increase the input impedance at the high frequencies, decrease the noise present, and improve the linearity and harmonic distortion.

With a total gain of 42 db, the best compromise may be to use half of it for the middle-frequency band beyond the low-frequency turnover, and the other half for the necessary low-frequency preemphasis. The AES curve is employed for developmental work.

From *Figs.* 6 and 8 we have,

$$\mu = K_1K_2 = (0.9)\ (205) = 184 \approx 42\ db$$

and therefore

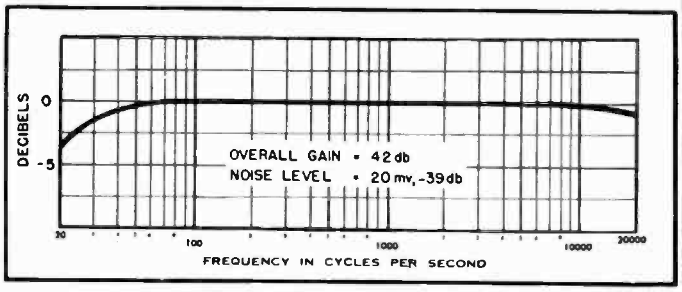
$$A(@1Kc) = \frac{42}{2} = 21\ db \approx 12.5 = \frac{\alpha\mu}{1+\mu\beta}$$

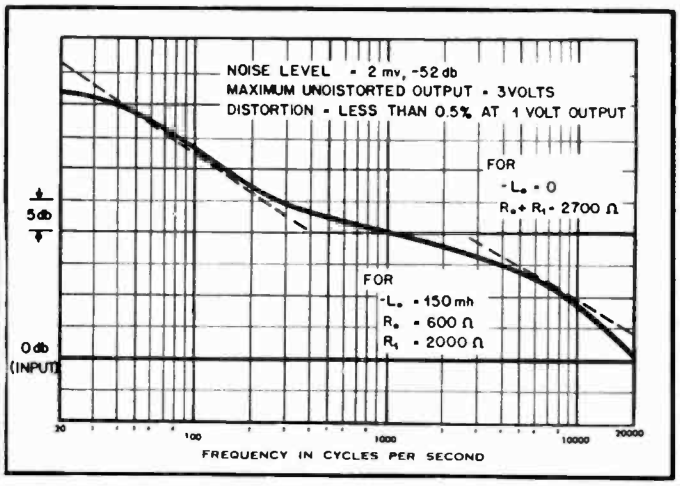
from which

$$(\alpha - 12.5\beta) = \frac{12.5}{\mu} \approx 0.07$$

[2] Sulzer, Peter, "Junction transistor circuit applications," *Electronics*, Aug. 1953, p. 170.

Fig. 7. Frequency response of the preamplifier without its negative feedback loop.
Fig. 9. With the feedback loop added the preamplifier response comes very close to the AES curve, which is indicated by the dashed line.





One pair of α and β giving an approximate solution is $\alpha = 0.88$ and $\beta = 3/50$ giving $R_o + R_i = 2700$ ohms and $R_s = 47{,}000$, assuming that C_i and C_s present negligible impedance (at 1000 cps) as compared to $R_o + R_i$ and R_s.

Since the AES curve's low-frequenc turnover is 400 cps we have,

$$C_s = \frac{1}{2\pi R_s f_o}$$

$$= \frac{1}{2\pi(47{,}000)\ (400)} \approx 8500\ \mu\mu f.$$

The pickup inductance L_o assumes considerable importance, since the inductive reactance of the pickup approaches the value of the input impedance at high frequencies. In calculating the high-frequency de-emphasis, therefore, the value of L_o has to be taken into consideration, and this, unfortunately, will limit the preamplifier to a specific cartridge. In addition, the value of L_o will effect the linearity of the ratios α and β. As α decreases with frequency, β increases, but their product does not remain constant, and therefore Equation

$A = \frac{\alpha\mu}{1 + \mu\beta}$ holds only approximately.

Note that these variations of α and β for a certain fixed frequency are not fixed, but depend on the value of L_o. If the impedance of L_o is small compared to the value of the resistance R_o, the gain variation will be less than 1 db, which may be considered acceptable.

Figure 9 shows the closed-loop frequency response of *Fig.* 6. With $L_o = 0$ and $R_o + R_i = 2{,}700$ ohms, the unit is flat to beyond 100 kc, while with a simulated Pickering cartridge ($R_o = 600$, $L_o = 0.15$ H and $R_i = 2{,}000$) the frequency response follows the AES standard curve closely. In actual practice R_i may be even lower, depending on the manufacturing tolerances of the pickup. With the application of negative feedback, the noise level has been reduced to 2 millivolts or about -52 db, which compares favorably with the average tube preamplifier.

The normal gain of 21 db is obviously inadequate for a low-output cartridge such as GE or the L-6 Audax. For the 10-millivolt class we need about 35 db at 1 kc, and with a 400-cps turnover we need another 22.5 db if we expect to go down to 30 cps, a total open-loop gain of at least 57.5 db without allowing any negative feedback for the low frequencies. Transistors giving gains of this order are not available yet. The Pickering line, on the other hand, giving a nominal output of as much as 70 millivolts, will work every satisfactorily with this circuit, giving an output of as much as 1 volt once the value of R_i is adjusted. Using a low-impedance, (professional-type) GE cartridge, the problem of input impedance matching is somewhat improved and it should be possible to cascade two junction transistors to get the necessary gain of 57.5 db for that pickup.

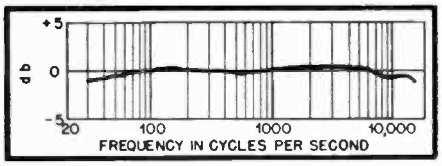


Fig. 10. This curve shows the result of an actual test with a Pickering cartridge and the London LP test record, which contains tones graded in level to conform to the usual LP music equalization.

Figure 10 shows the response of the preamplifier with a Pickering cartridge, Model 140, and a London LP frequency test record. Four pickups tested all gave substantially the same response. In listening tests the unit measures up very well, although we doubt that, at least at present, it is going to render obsolete its vacuum-tube counterpart. The complete absence of 60-cps hum and the elimination of any possible speaker mechanical feedback (microphonics) gives a new experience in the reproduction of sound, especially of the low-frequency band.

Power Supply

Although the complete preamplifier draws only 1.5 ma at 22.5 volts, this advantage is somewhat nullified by the fact that the voltage required is negative and is not found in the average present-day audio amplifier. This bias can be obtained, however, by several methods:

(1) Employing the same bias as the power amplifier, in the event fixed bias is used for the power tubes, since this bias will be in the same order of magnitude as the 22.5 volts originally used.

(2) The center tap of the high-voltage winding of the power supply transformer in most audio amplifiers is usually grounded, but if the center tap is returned to ground through a suitable bleeding resistor, a negative voltage is developed across the resistor which can be employed for power supply for the transistors. The value of this voltage will have to be subtracted from the total available B-plus, however, and the method may therefore prove unsatisfactory in some cases.

(3) A small power supply can be built with either a line-isolation transformer or two filament transformers connected back to back, using germanium crystals for rectifiers. Filtering is no problem because of the small current drain.

(4) The most obvious method will be to use a B-battery. Burgess type 5156 and the miniature hearing-aid type U15 have been found very satisfactory. By modifying the automatic shut-off mechanism of most record players, a d.p.s.t. switch can replace the present s.p.s.t. and the battery can be connected to the preamplifier through the extra pole of the switch. Since there is no warm-up time, the unit will go on immediately when the record player is started and will be disconnected automatically after the last record is played, resulting in a considerable extension of the life of the battery. For an average (!) record playing time of six hours a day, a U15 is expected to last about six months and with the new mercury batteries this may well be doubled.

Conclusion

In this article an attempt has been made to present some of the advantages and limitations of transistors, their basic circuit configurations, and their electronic equivalents and to show how these concepts can be applied in designing audio components that will compare favorably with their vacuum-tube counterparts. A complete preamplifier-equalizer employing a total of five transistors with a total drain of 4 ma at 22.5 volts is under development at present. Preliminary runs and listening tests give satisfactory performance and an article on the unit will be submitted for publication when performance tests and evaluations are completed.

Although an adventure into transistors and tantalum capacitors is likely, at present, to prove rather strenuous financially, it is certain that future improvements in higher gain, lower noise level and distortion, better stability and uniformity, together with lower cost, will prove the transistor to be an indispensable part of the preamplifiers and remote controls of the future.

BIBLIOGRAPHY

W. Shockley, *"Electronics and Holes in Semiconductors,"* D. van Nostrand, New York, 1950.

IRE Proceedings, Transistor Issue, Nov., 1952.

H. C. Montgomery, "Transistor Noise in Circuit Application," *Proc. I.R.E.* Nov., 1952, pp. 1461–71.

Ryder and Kircher, "Some Circuit Aspects on the Transistor," *B.S.T.J.* Vol. 28, No. 3, July, 1949, pp. 367–400.

R. L. Wallace and Raisbeck, "Duality as a Guide in Transistor Circuit Design," *B.S.T.J.*, Vol. 30, April, 1951, pp. 381–417.

D.C. Pack for Heaters and Bias

ALLAN M. FERRES*

A simple and practical device which will provide hum-free heater power for low-level amplifier stages, thus making one further step toward perfection in reproducing systems.

IN ORDER TO TAKE full advantage of the excellent fidelity now available in home radio-phonograph installations, the hum level of the system must be reduced to a value much lower than that required a few years ago. Moving toward improved realism, designers have incorporated into the systems speakers and amplifiers with better low-frequency response and, at the same time, signal sources and circuit which require greater voltage gain at these low frequencies. The magnetic cartridge, which is the accepted standard for high-fidelity record reproduction, and the increased use of magnetic tape in the home, combined with "loudness" and separate bass-boost controls, all contribute to the problem. A gain of 30 to 40 db at hum frequencies over the mid-frequency gain is often encountered in most equipment.

The hum reduction problem centers around the low-level stages where the signal level may be as low as 100 microvolts. In order to keep the hum from exceeding the acceptable maximum, usually considered to be 55 to 60 db below normal listening level, careful design is essential.

After all possible precautions are taken to reduce the hum picked up by the input source, the residual hum is produced by the tube operating at the lowest signal level. In good design, this is the first tube in the circuit, that fed by the input source. The hum produced in this stage may be considered to come from one or more of four sources: magnetic fields, electrostatic fields, plate-supply ripple, and heater-supply ripple. As the a.c. magnetic field is usually caused by the power transformer, proper spacing between it and the input tube will eliminate this source of trouble. The effect of electrostatic fields may be minimized by completely enclosing the tube and its associated parts in an aluminum or copper shield can. As a resistance-coupled stage, customarily used in preamplifiers, requires plate current of the order of one milliampere or less, the plate voltage ripple can be reduced easily by means of an R-C filter, and so may be discounted as a "problem" in hum reduction. Therefore, the remaining factor—heater-supply ripple—is the common cause of the residual hum.

When a.c. is used to heat the cathode of a tube, hum enters the signal circuits by heater-cathode leakage, the a.c. magnetic field surrounding the heater, and by capacitive coupling between the heater and the other tube elements and circuit components. Although careful selection of tubes and arrangement of parts can bring this hum down to a minimum, a point is finally reached where the only method of further reduction is the use of d.c. as the heater supply. (Although high-frequency heater current can be used, it is not considered in this discussion due to its complications.)

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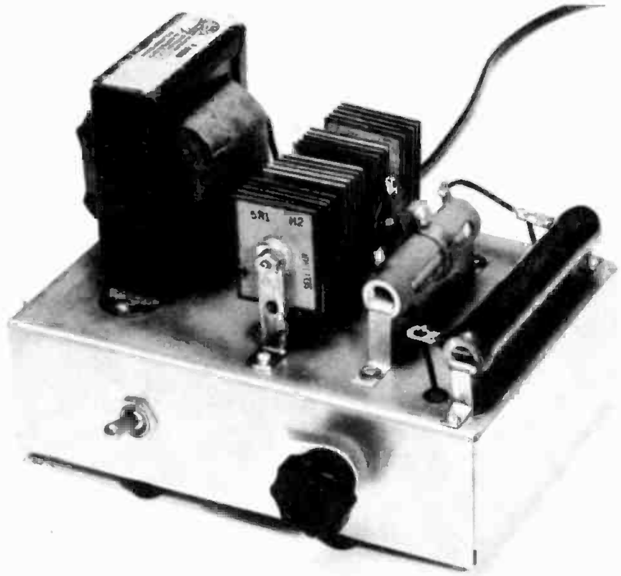

Fig. 1. External view of the author's d.c. heater pack.

Advantages of d.c. Supply

The use of a d.c. heater supply has many advantages. As power-supply a.c. does not have to enter the shield compartment of the low-level stages, shielding requirements of grid and plate leads are not as stringent and better high-frequency response is easier to obtain. Cathode followers and certain types of phase inverters and feedback circuits which require the cathode to be un-bypassed can be used at lower signal levels. This, at times, simplifies design problems. The type of high-frequency oscillator often used in FM receivers in which the cathode is above ground potential is a stubborn source of modulation hum until d.c. is used. More leeway is permitted in the choice of the type of tubes which can be used and the circuit will be less critical as to individual tube selection.

In any given amplifier, the improvement to be expected by changing the heater current from a.c. to d.c. is difficult to predict, as too many factors are involved. If the amplifier under discussion is still in the design stage, then d.c. should be used if the lowest hum level is required, depending only upon the cost involved. If the amplifier has already been built, then the improvement can be readily measured by disconnecting the heater leads from the transformer and substituting a battery of the proper voltage. (This, incidentally, is a good way of isolating the cause of hum in equipment when trouble-shooting.) It is possible that when the battery is used, the hum will increase. This is due to hum voltages from the heater having been out of phase with hum caused by some other source. It is usually due entirely to coincidence and should not be depended upon as a method of hum reduction, as the other source often produces a voltage which varies in amplitude and phase. The other hum source can now be tracked down and corrected.

If the d.c. heater supply used has a ripple of 1 per cent or less, it may be assumed to produce no more hum than the battery, and the supply should be designed with this in mind, unless experience has proven that in a particular circuit a greater ripple can be tolerated.

Methods of Obtaining d.c.

In general, there are three sources of d.c. which can be used. The first is to connect the heaters in series in the negative side of the plate power supply, and using a shunt or bleeder resistor, as required, so that the proper current will flow through the heaters. If a suitable power supply is available, this is the simplest and cheapest method. The voltage drop across the heaters is sometimes used as bias for the power output stage, eliminating the need of a relatively high-wattage cathode resistor.

The second method, often used in commercial equipment, is to connect the heaters in parallel and operate them from a low-voltage, high-current power sup-

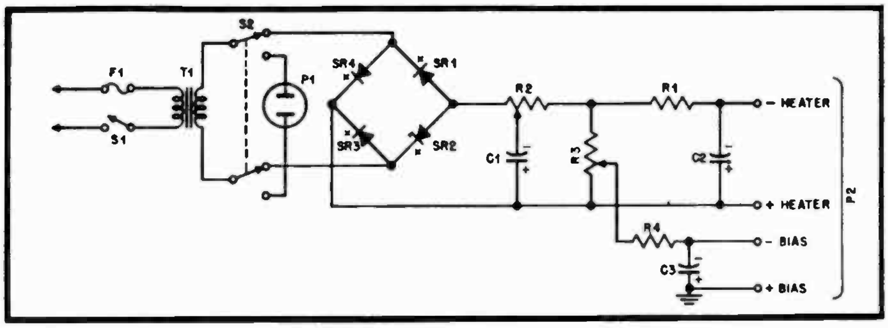

Fig. 2. Schematic of the d.c. power supply.

ply. Parallel operation of the heaters is convenient, but suitable rectifiers and transformers are not readily available from the usual parts distributor, and the cost of such parts tends to be high.

The third method, used in the supply described in detail here, employs an isolation transformer with a 117-volt secondary, a selenium-cell bridge rectifier, adjustable series resistor for voltage adjustment and filtering, and two 150-volt electrolytic capacitors. All parts are standard items, readily available. The part values are chosen so as to furnish 150 ma at any voltage from 12 to 72, as may be required, with a ripple of 1 per cent or less. Although there is no objection to operating any of the tubes in an amplifier on d.c., it is usually not necessary for hum reduction in stages where the signal voltage averages one volt or so. In practice, the phono, tape and microphone pre-amplifier and the tubes in the equalizing pre-amplifier are d.c.-operated to advantage. It may also be desirable to supply d.c. to the heater of the high-frequency oscillator of the FM tuner, if modulation hum is encountered. As almost all 6.3-volt tubes currently used in oscillators and low-level audio stages have 12.6-volt equivalents, no difficulty will be experienced with tube selection. The tube heaters should be connected in series with the first, or lowest-level tube, connected to the negative terminal of the power supply.

Another feature of this supply is its ability to furnish up to 90 volts or so of fixed bias for the more efficient operation of Class A or A_1 output tubes. The bias is adequately filtered and adjustable from zero to the maximum value. For experimental or test work, the isolation transformer can be used separately, being switched to a two-pole utility receptacle for that purpose.

Construction

The mounting of the parts is not critical, provided a normal amount of ventilation is furnished. The unit built by the author was mounted on a 5×7×2-in. chassis, the parts being arranged as shown in *Fig.* 1. The cardboard covered filter capacitors, R_3 and the fuse are mounted underneath the chassis. On the front can be seen the bias-adjusting pot and the a.c. on-off switch. On the back of the chassis are the line cord, the 4-terminal output socket, the two-pole receptacle, and the secondary switch for the transformer. The four rectifiers are mounted by passing a threaded rod 4 in. long through the center holes and supporting it at the ends with two 1½×½-in. angle brackets. Small rubber grommets protect the leads passing through the chassis. The values of R_1, R_2 and C_2 are selected from Table 1. *Figure* 2 is the complete schematic for the unit.

For increased tube life—and in some cases for lower tube hiss—it is recommended that 12 volts be applied to 12.6-volt tubes and 6 volts to 6.3-volt tubes. A fine adjustment of the heater voltage is made by varying the position of the tap of R_2. It should be noted that there is no direct connection between the heater supply and the chassis, which is grounded. This is done so as to permit a wider choice in the selection of a heater grounding point in the amplifier. The proper point to ground the heater string is determined by turning on the amplifier and then running a lead from the point where the audio input jacks connect to the amplifier chassis to the tube heater pin which produces the least hum. Usually this is either the more negative terminal of the first tube or between the first and second tubes.

TABLE 1

Values for R_1, R_2, and C_2

Output (volts)	R_1 50-W fixed	R_2 50-W adj.	C_2 150-v. Electrolytic
12	500	200	80 µf
18	500	200	80
24	500	100	40
30	500	100	80
36	500	100	80
42	250	250	40
48	250	250	40
54	250	150	40
60	250	150	40
66	200	100	80
72	200	100	40

After the ground connection to the heaters has been made, then the fixed bias is adjusted to the desired value by means of the 50,000-ohm pot.

The advantages of using d.c. on the heaters of low-level stages can be fully appreciated when it becomes possible to turn the gain up to maximum—with no signal input—without any output from the speaker, except a possible increase in tube hiss. When such a condition is reached, the system may be said to compare with professional installations, assuming that its other characteristics are equally ideal.

Parts List

C_1, C_3	40 µf, 150-volt, elect.
C_2	See text and Table 1
F_1	1-amp. fuse
P_1	2-pole female receptacle
P_2	4-hole socket
R_1, R_2	50-watt wirewound; see text and Table 1
R_3	50,000-ohm potentiometer
R_4	47,000 ohms, 1-watt
S_1	SPST toggle switch
S_2	DPDT toggle switch
$SR_{1,2,3,4}$	150-ma, 130-volt selenium rectifiers
T_1	Isolaton transformer, 117/117 volts (Merit P-3096 or equivalent)
	Fuse clip
	5×7×2 chassis
	Knob for R_3
	Line cord and plug

A Corner Horn For The Small Listening Room

WAYNE B. DENNY*

Large-aperture horns, desirable for adequate bass response, can be made to appear much smaller than they really are by effective use of a corner.

Most designers and users of higher-quality reproducing equipment agree that adequate bass response requires rather large speaker enclosures. Horn speakers in particular often require 20 or more cubic feet of space for efficient radiation of low-frequency sounds. This requirement poses a problem to those who must confine their listening to apartment-sized living rooms. Too often, the person who installs a better than average audio system finds that the system—particularly the speaker enclosure—"takes over" the room to the point where it dominates the entire listening area visually as well as aurally. This may not bother the high-fidelity addict but it is likely to engender a high noise level in the form

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Fig. 1. The finished horn seems very small physically in comparison to its effective size.

of vocal complaints from the distaff side.

Is it possible to build a speaker horn which is adequate in size but which does not completely dominate restricted living quarters? After some preliminary study it appeared possible to solve this problem satisfactorily and the unit to be described is offered as one solution. No new principles are involved but effective use is made of some well known ideas.

Inspection of *Figs.* 1 and 2 will serve to illustrate the fundamental features of the horn. The driver is back-loaded at low frequencies in the conventional manner. The horn expands toward the rear—that is, toward the corner of the room—and then expands toward the front at either side of the structure. The large portion of the horn is formed by the top panel, side panels, floor, and room walls. Room walls and floor are visible and it is this feature which makes the unit appear much smaller physically than its actual acoustic size. The entire volume beneath the top panel is of the order of 20 cubic feet! Yet it looks no larger than a moderately large corner table. Part of this deception is caused by the ratio of width to height. The height is just under 33 inches—about right to hold a table-model television receiver where it is most easily viewed. This is important in those rooms which provide but one unobstructed corner suitable for viewing and listening.

Major dimensions are shown in *Fig.* 2. As shown, the horn approaches exponential shape and provides for a theoretical low-frequency cutoff at about 60 cps. The mouth area—just over 1000 square inches—is large enough to subdue serious reflections and this aperture is augmented by walls and floor.

The structure is very solidly constructed of ¾-inch plywood. All joints are secured by wood screws and glue except the top which is, of course, removeable. Weatherstripping material is used to ensure a tight seal between the top and the walls of the room. Internally, 1 x 2-inch reinforcing strips are used along all corner joints. These strips are not shown in the diagram of *Fig.* 2 but their use is illustrated in *Fig.* 3.

Construction is relatively simple and need not deter the average amateur carpenter or cabinetmaker. The only difficulty—a minor one—is caused by the fact that several panels meet at angles other than 90 degrees. A little care is necessary to insure a solid and tight-fitting joint. All panels can be obtained from a single sheet (4 x 8) of plywood as shown in *Fig.* 4. It is suggested that all outside panels be cut as shown so that the grain of the wood is uniform in direction. The smaller panels are cut from the residue and are preferably cut oversize and planed to fit.

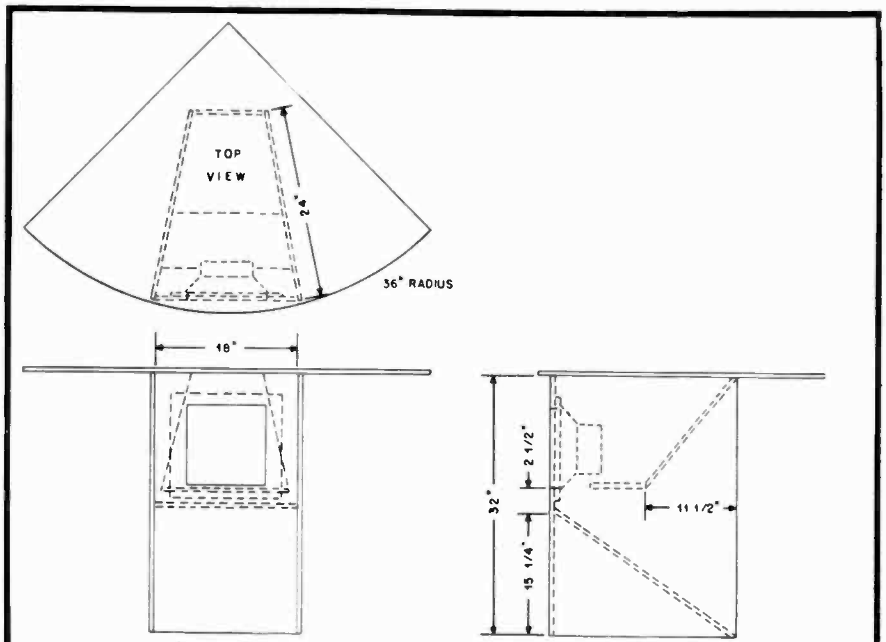

Fig. 2. These three views show the general makeup and major dimensions of the enclosure. Dimension details can be changed to suit other speakers.

Fig. 3. This view of the enclosure turned around from its normal position, was taken from above and behind. It is useful in showing many details, though somewhat deceptive from a photographic standpoint. To avoid misunderstanding, remember that the side walls are rectangular, the top parallel to the floor.

At prices prevalent in the Midwest the entire structure can be built of fir plywood for about $25. The use of birch or similar outside ply will double the cost.

Performance Characteristics

A number of speaker enclosures have appeared in the literature during the past two years which employed the General Electric S1201-D speaker as drivers. A few of these descriptions have included impedance curves. For purposes of comparison, a similar driver was installed in the corner horn described here and impedance curves were prepared with the aid of the circuit of *Fig.* 5. The results are shown in *Fig.* 6. Comparison of the solid-line graph with the corresponding curves for other enclosures shows that the present unit has unusually small fluctuations of impedance with respect to frequency. Although no claim is made that such curves are completely indicative of speaker performance, the inference is that increased air loading makes the response less variable with frequency. Analysis of the data with and and without fiberglas absorbing material in the space behind the cone indicates that impedance fluctuations are smaller with the fiberglas installed. Percentage-wise, the fluctuations are small in either case within the frequency range from 30 to 400 cps.

A rule of thumb procedure for checking the damping of a loudspeaker consists of starting and stopping a small electric current through the voice coil. After the circuit is *closed* the speaker is subject to heavy electrical damping in addition to mechanical and acoustic damping. After the circuit is *broken* an open circuit exists and the damping is exclusively mechanical and acoustic. The difference in sound, if any, provides some

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A Tuned-Pipe Enclosure for Bass Enhancement

RAYMOND H. BATES*

A simple and easily-built enclosure will give excellent sound quality from any good 12-inch loudspeaker, with low-frequency reproduction equivalent to many larger cabinets.

According to literature presented in the September 1952 issue of Audio Engineering—I refer specifically to the article written by Mr. John E. Karlson entitled "A New Approach in Loudspeaker Enclosures"—a long closed-end pipe, having a long notch at the open end, will give practically continuous radiation over the frequency range required for good audio reproduction. The pipe length must, of course, be considerably larger than the width and depth in order to achieve the proper effects. With these fundamental characteristics of the closed-end pipe in mind, I have developed a fairly simple yet effective corner enclosure for 12-in. speakers, shown in *Fig*. 1.

The configuration was generated by considering first a long triangular shaped pipe 6 ft. long with the speaker mounted at one end and the other end having an exponentially tapered slot, as shown in *Fig*. 2. Since this is obviously a back-loading device, and we are interested primarily in enhancing the low-frequency response, the slot or notch at the open end can be made reasonably short, considerably less than the 2/3 indicated for broad coverage. Further, if we now fold the 6-ft. pipe, as shown in *Fig*. 3, we get a package that is of practical size. This configuration permits direct radiation of the middle and high frequencies while the lows are radiated directly and are also augmented by the back radiation through the short exponential slot. That was borne out by impedance measurements, and was verified by extended comparative listening tests. Note in *Fig*. 4 that the free air resonance of 65 cps for the Electro-Voice SP12B is broken up and smoothed out with the curve essentially flat down to 30 cps. The non-resonant character of this enclosure has been demonstrated by extended listening tests using such records as the Cook organ records as well as the Capitol test record. There is

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CORNER HORN

(*from preceding page*)

indication of the coupling between the cone and the air. Using a single dry-cell as a current source, this test was applied repeatedly to the corner horn enclosure and no audible difference between *make* and *break* was found. It was inferred, therefore, that the damping was adequate and that transient response would be satisfactory. The latter conclusion was substantiated by listening tests. One simple test is to listen to a *good* recording of percussion instruments. Bartok's *Sonata for Two Pianos and Percussion* (Victor) is satisfactory for this purpose.

No finite horn, no matter how carefully designed and constructed, is completely free from resonance phenomena. The horn which has been described exhibits a slight resonance and this is shown by the shape of the impedance curves of *Fig*. 6. There are two humps in the impedance curve where the impedance rises about 30 per cent above the nominal voice-coil impedance. The two humps are nearly symmetrical about the free-cone resonant frequency. This suggests that the resonance frequencies of horn and driver are the same, a desirable condition. There was no deliberate attempt to bring about this desirable state of affairs, the curves just came out that way.

If the horn is used in conjunction with drivers for low-frequency reproduction only, it may be expected that the resonant frequency of the driver may be somewhat lower than for the one shown. If so, a slight readjustment of the internal panels may be desirable. It is for this reason some of the internal dimensions have been omitted. For example, slight changes in construction will permit the use of a 15-inch driver. Under such circumstances the throat area is easily increased, with a consequent decrease in horn length. Fortunately, the design of the enclosure is such that changes of this type are easily made after construction is nearly complete.[1]

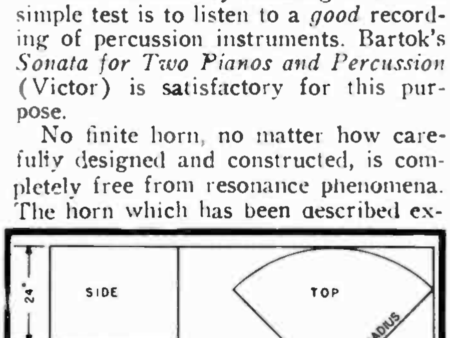


Fig. 4. How to cut all the main panels from a single 4×8-foot sheet of plywood.

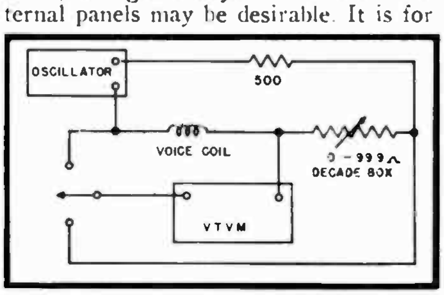


Fig. 5. Circuit used for impedance measurements. At each frequency the decade resistance is adjusted for similar voltmeter readings in both switch positions. The resistance is then equal to the absolute value of voice-coil impedance.

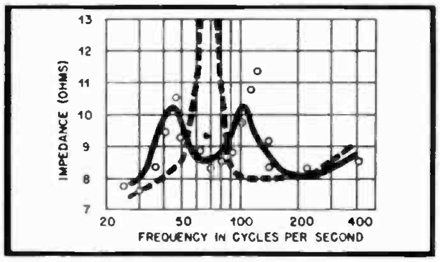


Fig. 6. The solid line shows the impedance curve of the GE speaker in the enclosure, with fiberglas completely filling the space behind the driver. The dots show irregularities which occur when the fiberglas is absent. The dashed line shows impedance peak of the driver without an enclosure.

[1] Apparently not all General Electric Model S1201-D cones have the same resonant frequency. If a driver of different resonant frequency is used it should not be expected that the impedance curves will be identical to the ones shown in *Fig*. 6. However, if the fluctuations in impedance with frequency are not severe, good results may be anticipated.

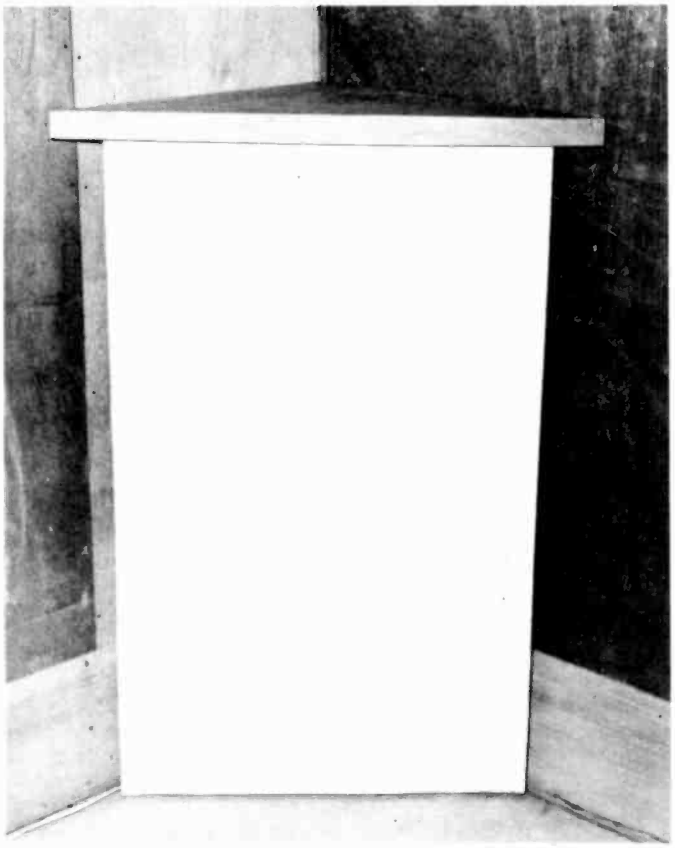


Fig. 1. The author's cabinet in finished form. Light appearance of front is due to the use of natural-colored monk's cloth for the grille.

no audible boomy or boxy sound.

For those who are interested in constructing this enclosure the diagram shown in *Fig.* 5 will serve as a guide. The usual precautions of using wood screws and glueing all joints for an air-tight seal apply. The slot dimensions are indicated on the drawing.

Figure 6 indicates the various pieces needed to assemble the enclosure. In assembly the two sides should be nailed together first after wood glue has been applied to the contacting surfaces. Next the top and bottom pieces should be assembled, again using nails and wood glue, for a neat joint. Now the two lower left pieces of *Fig.* 6 should be assembled using nails and wood glue, and then this sub-assembly should be glued and screwed to the two sides already assembled. See *Fig.* 5 for proper location of this sub-assembly. The front panel should next be fastened as indicated in *Fig.* 5, using wood screws and glue generously to insure an air tight seal at all contacting surfaces. This whole package should now be allowed to set for whatever time is indicated on the can of wood glue you may have used. I used Casco and let the enclosure set overnight, and should resemble *Fig. 7.*

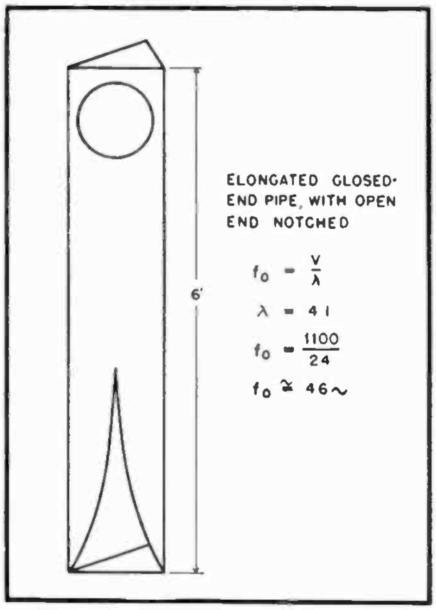


Fig. 2. Design of cabinet was developed from this long closed-end pipe with notched front.

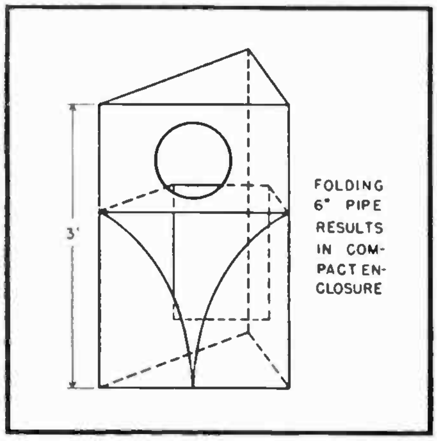


Fig. 3. Appearance of cabinet after folding from the original "pipe" of Fig. 2.

The basic enclosure is now ready for final finishing. If you like the simple modern appearance that mine has, as shown in *Fig.* 8, you may now screw a foot piece to the bottom, cut the same size as the bottom piece, and then glue a ¼-in. sheet of foam rubber thereon for acoustic insulation from the floor. The enclosure should now be set in whatever corner you have chosen for it (preferably one on the longer axis of the listening room) so that the sides of the enclosure do not touch the moulding along the base of the corner walls of the room so you can measure the top finishing piece. This piece should be sufficiently large to fit snugly against the two corner walls and still jut out over the front grill of the enclosure approximately 1 inch for good appearances. In my case this turned out to give dimensions of 20 x 20 x 28 in. Mount this piece to the top by inserting wood screws through from the inside of the enclosure. Now glue a thin strip of foam rubber along the edges that contact the wall in order to obviate any vibrations being set up at the points of contact. Next the Kimsul pad can be fastened to the top with glue and a few furniture tacks. You are now ready to mount the speaker. A 15 x 15 in. speaker mounting board should be cut from ⅝-in. plywood stock, with a 10½-in. circular hole cut in the center for mounting the 12-in. loudspeaker. Speaker lead-in wires should be connected through two ⅛-in. holes conveniently placed in one side of the enclosure near the top. The speaker board may now be mounted over the 13-in. hole in the front panel using eight wood screws equally spaced about the periphery of the mounting board as shown in *Fig.* 9. Finally the front grill frame, 38¼ x 23 in., should be fabricated from ¾-in. square pine stock. This frame should be covered with lumite in your choice of color. The completed front cover may now be fastened to the enclosure front by means of four cabinet door catches. These should be mounted on the sides, two near the top and two near the bottom. This

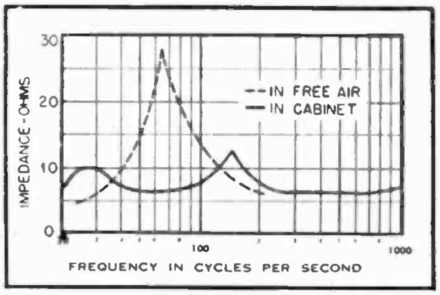
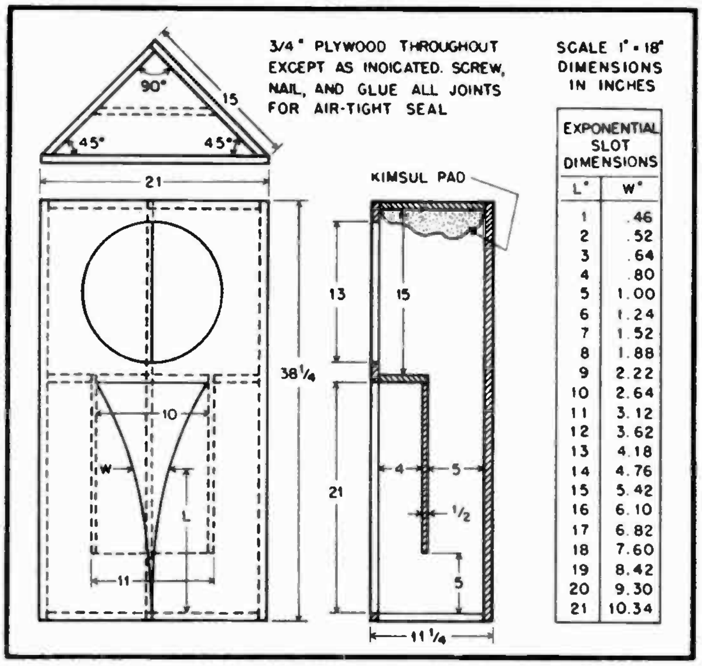
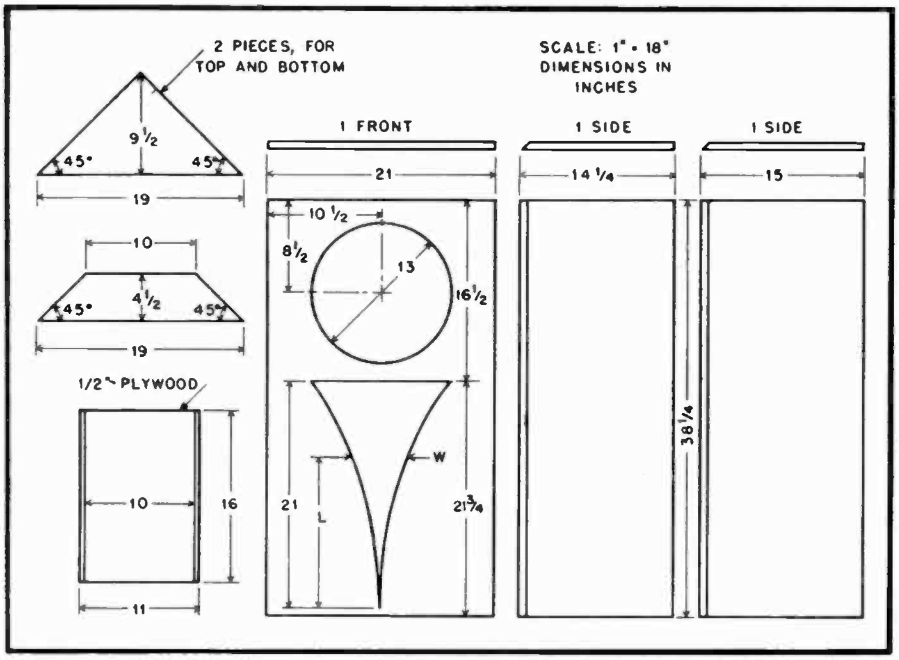


Fig. 4. Measured impedance curve of E-V SP-12B loudspeaker in the author's cabinet. Note reduction of natural resonant peak and extension of low-frequency range.

Fig. 5. (right). Sectional drawings of the simplified corner enclosure. Fig. 6 (below). Details for cutting plywood pieces for the cabinet. 15×15 in. speaker mounting panel.

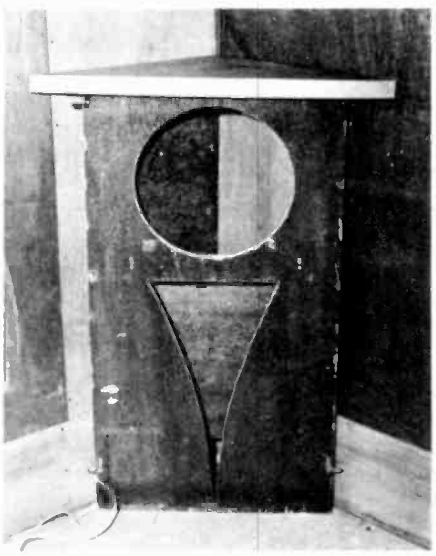


EXPONENTIAL SLOT DIMENSIONS	
L"	W"
1	.46
2	.52
3	.64
4	.80
5	1.00
6	1.24
7	1.52
8	1.88
9	2.22
10	2.64
11	3.12
12	3.62
13	4.18
14	4.76
15	5.42
16	6.10
17	6.82
18	7.60
19	8.42
20	9.30
21	10.34



expedient allows for ready access to the speaker in case you want to change speakers etc. In my case, the top was finished by sanding and then applying clear varnish to bring out the natural grain of the mahogany. The over-all effect is very modern. The simplicity of this basic enclosure permits of flexibility in final exterior finish so that if you do not like the modern approach you may apply some other type finish.

Fig. 9. Semi-finished cabinet with loudspeaker in place on its mounting panel. Note brackets on edges of front panel for mounting the trimming frame.



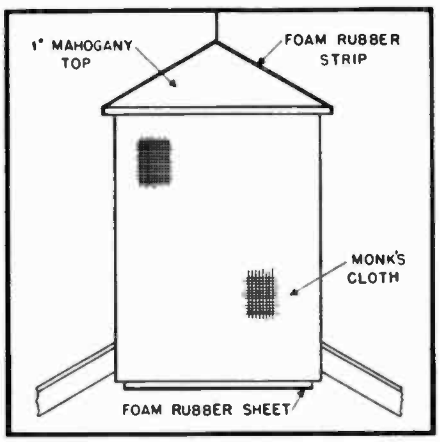


Fig. 8 (above). Sketch of finished appearance.

Fig. 7 (left). Cabinet in stage of construction before mounting loudspeaker panel.

Evolution of "The Horn"

EDWARD V. KETCHAM, JR.*

The conception, design, and construction of a speaker enclosure consisting of a closet-contained low-frequency horn with a midfrequency horn and two high-frequency sound producers.

"THE HORN" is a home-built loudspeaker system. It is large in size and magnificent in performance. It was conceived as a result of reading about Albert Kahn's huge home audio installation publicized as a "Residence Entertainment Center" by Electro-Voice, Inc. Mr. Kahn is president of this organization. The "Center" features a bass section comprised of a large horn driven by two 18-in. woofers, and occupies one entire end wall of a sizable room.

More specifically, "The Horn," truly an amateur's project, was first considered as a half-size edition of Mr. Kahn's installation, turned on end and backed into a rectangular hall closet. As completed (*Fig.* 1) it became a combination of Klipsch speaker cavity and horn throat principles, feeding through an exponential horn, plus design features and driver components of the Electro-Voice Patrician. The entire assembly was designed and built to fit existing closet space and exhaust into one end of the living room listening area. Technicians may find faults in its layout, but it performs superbly.

General procedure was as follows. Details of the "Residence Entertainment Center," the Patrician, and some basic rules of horn design were obtained. Paul Klipsch's papers on folded horn principles and the Jensen monograph on horns were studied. The basic idea of a low-frequency horn with its mouth divided in two by a compartment containing associated apparatus was drawn to rough scale. It became apparent that it would be advisable to face the low-frequency woofer speaker *away* from the listening area, and fold its associated horn around two right-angle turns to bring the latter's mouth out in the desired direction. This design feature would provide the longest path and most gradual taper within the limits of available space.

Concentrating on this low-frequency horn, an approximate path length was established. After several false starts a list of cross-sectional areas was compiled. The list itemized the recommended area of (a) the mask opening in front of the 18-in. woofer, (b) a pre-throat chamber, (c) an eight-inch-long throat, and (d) calculated areas at six-inch intervals along the length of the horn. Figures from (d) established an exponential curve or expansion rate to the recommended mouth area. Working dimensions were figured from (d) by assuming widths as large as closet space allowed and dividing these into the listed values, the quotients denoting height or depth dimensions. The drawing of *Fig.* 2 details the results of these calculations.

Lest your interest sag at this point, the end product is a low-frequency horn from which pours full, clean, unforced low-frequency sounds—a tuba player at an interview trying for his lowest note, Mr. Cook's organ records, a bass viol, drums—without resonant peaks or "behind-the-door" muffled effects. First indications of its performance became evident upon completion of the basic structure and the installation of the 18-in. woofer in its tightly sealed cavity. A test was made on the concrete floor of the garage workshop using a Cook test record and 10-watt amplifier.

The first try ran at moderate volume. Waiting for the test record to run its course provided anxious moments as treble tones weakly wandered forth. One thought was paramount. "When will it take hold? *Will* it take hold?" At about 500 cps the test tone became solidly effective and then progressed lower and lower. Positive response was audible down to 30 cps and apparent response below that; however, the pickup used and this layman's ears preclude accurate judgment in this rock-bottom region. Theoretically "The Horn" functions to 25 cps and the speaker resonates at 27 cps.

A repeat test at high volume produced sufficient unadulterated low-frequency

Fig. 1. The finished enclosure is mounted in a closet with the divided horn exhausting into the listening area. Between the two sections is the TV receiver. The upper section will be covered with a grill similar to that over the lower section.

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sound power to be a potential cause of physical damage in a finished room. Vibrations created a sense of being surrounded by something untouchable. Small objects moved of their own accord. Others vibrated audibly. Small wonder that a chance visitor arriving upon the scene, replete with the odor of drying paint, accused the writer of nefarious activities relating to the production of illegal beverages!

Construction

The photos of *Figs.* 3 and 4 illustrate construction details. Upper and lower mouth sections, throat and back are built separately, then assembled. The speaker cavity came last. Joints are glued throughout. After testing at loud volume strong bracing was added to the back of the speaker cavity to reduce the possibility of developing extraneous vibrations. It is believed this can be better accomplished by building the cavity irregular in shape while maintaining its cubic content.

Having successfully provided for low-frequency range, the next stage was the building of a 200- to 600-cps horn for a 12-in. woofer, as prescribed by the Patrician details. This was patterned directly from the furnished data except for two variations. Available width being less than specified, the height was increased to maintain prescribed mouth area. The second change consisted of arranging the horizontal and vertical axes off center so as to aim the unit toward the exact center of the listening area at ear height.

Then the upper frequencies. Purchased Partician components were arranged to point at the listening area center point. From 600 to 3500 cps a horn-type driver unit operates through a multicellular horn with 600-cps cutoff. Two tweeters operate from 3500 cps up. One, a horn-type driver operates through a multicellular horn with 1500-cps cutoff. The other is an 8-in. cone speaker modified for extreme high-frequency operation. A four-way dividing network channels source material signals to the proper drivers.

With the assembly of all components completed, further tests were in order. After preliminaries with the test record, the writer proceeded to launch units of the New York Central Railroad upon a new route via Cook's "Sounds of Our Times" by way of an open garage door. The open mouthed admiration of a four-year-old neighbor was wondrous to behold.

A word now from the voice of experience! No doubt you know of the fellow who built a boat in his basement and was unable to get it out. Similarly let it be understood that a big audio horn once built . . . has to be installed. In planning, one must consider not only available area for the installation but also difficulties in reaching that area. "The Horn," while restricted in width to that of a hall closet, is ceiling high and cannot pass through doorways in an upright position. A five-foot closet depth and corresponding front-to-back dimension, however, make transportation possible face down. This does not entirely conclude the problem as the over-all diagonal measurement from bottom front to top back momentarily is all important when raising the assembly to an upright position. In the case of "The Horn" one inch to spare proved sufficient.

Finishing was handled in a novel manner, as the photos reveal. The wall was first torn away exposing the closet installation area. The unit was backed into position. Wedges were driven between the unit and surrounding walls. Lag bolts through the sides into adjacent studs made everything "solid as a house." With "The Horn" finally at its destination, panelling was installed across the entire end of the room on top of existing plaster. The panel seams were spaced so that the wall opening at the closet measured exactly four boards wide, as can be seen on *Fig.* 5. An up-and-down sliding panel of like material and similar width was constructed on concealed slides so that it can be raised out of the way through the ceiling. A counterweight simplifies this maneuver. In passing, the writer admits partial defeat at this stage as the panel was divided and hinged across it center making a visible seam. The only alternate was to "raise the roof." The next project will be to provide push button control of this panel with an electric motor mechanism.

A sliding drawer or cradle permits the TV chassis to be serviced; a TV front panel and a plastic grill cloth provide finished appearance. Source material comes from amplifier, FM tuner, and record player in a fireside wood closet at the far end of the room.

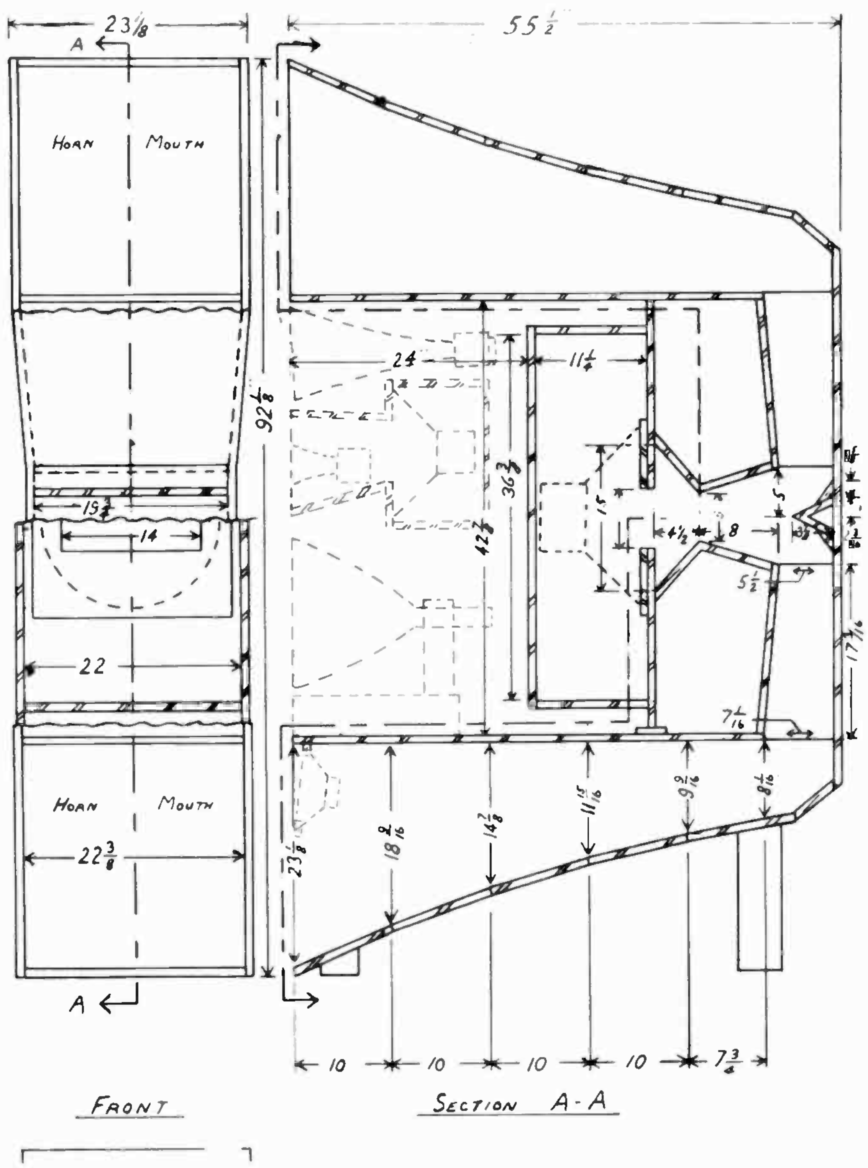
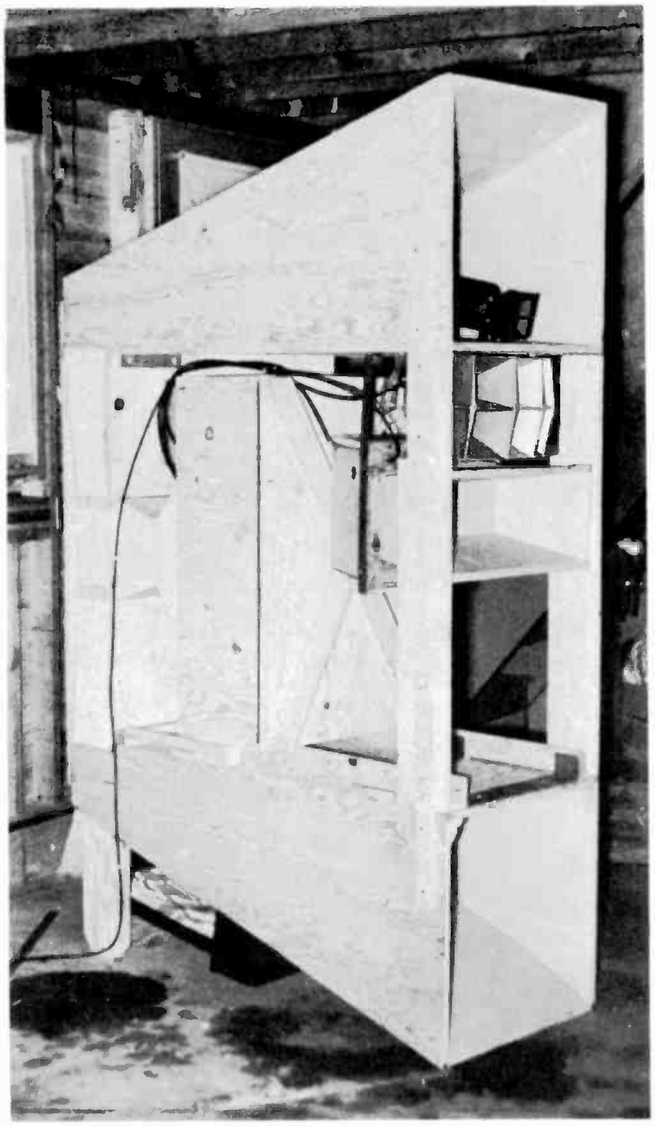


Fig. 2. This drawing gives all the dimisions of the author's version of "The Horn." Some alterations may have to be made by other constructors to fit existing spaces.

Other Speakers

In experimenting with "The Horn" by changing relative volume levels of the different drivers, the writer has formed an opinion that may be of general interest. It is that most approaches to extended low-frequency reproduction



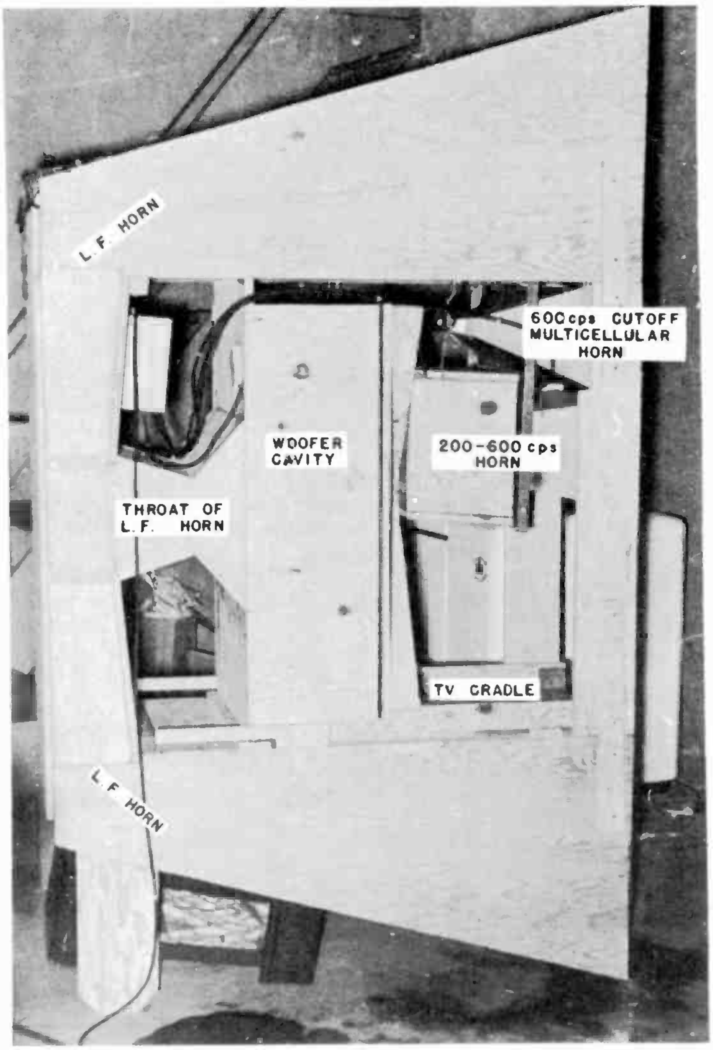


Fig. 3 (above). Side view of "The Horn" shows some idea of the construction. Fig. 4 (left). This three-quarter view shows placement of the 200-600-cps and lower-frequency horns.

in the home indirectly cause poor speaker response in what may be termed upper bass. To substantiate, the bass-boost curve of home amplifiers peaks between 50 and 100 cps. Enclosure designs are available that are remarkably efficient in this range, and when used with a low crossover network and level controls usually make amplifier bass boost undesirable. Under such circumstances the direct-radiator cone speaker employed for midrange reproduction lacks bass boost that it can use to advantage. Said boost not being present, the overall response curve of the speaker system droops in the upper bass range. This droop may lead some to think their low-frequency units perform better than is actually the case, due to the relative difference in output volume between the low and midrange speakers.

The remedy is to employ a separate upper-bass speaker. Its efficiency must compare favorably with the extreme low-frequency unit. Such a device is the 200–600-cps horn with 12-inch driver described above as a component of "The Horn." It plays a large part in producing the wide-range, even response that is the outstanding characteristic of this multiple speaker system.

Building and installing "The Horn" was an adventure. Test record tryouts and lengthy listening sessions have revealed wide-range efficiency most rewarding for the effort expended in its creation. It utilizes equipment found in the most elaborate factory assembled systems and several enthusiastic critics consider that its performance ranks with the best professional demonstrations, yet through home construction many dollars were saved, and the system was built in. The writer formerly used a high-priced coaxial speaker mounted at the location now occupied by "The Horn." The closet served as an infinite baffle and the same source material equipment was in use. There is little comparison. Direct-radiator speakers are not in the same league with several properly-horn-loaded drivers.

Yes, it was a satisfying adventure. Look around, friend! Have you a workshop? A strategic closet? Ever knocked down a wall? You have? Well?

Fig. 5. This photo shows how the entire installation is covered by the vertically sliding panels when it is not in use.

A Corner-Mounting Infinite Baffle

M. V. KIEBERT, JR.*

A presentation of some of the problems involved in loudspeaker enclosures for high-quality reproduction, and the author's solution with a suitable design to house an LC-1A speaker mechanism.

PAST EXPERIENCE has all too often made us painfully aware that what was often called a "loudspeaker" was just exactly that and not the "reproducer" that it should have been.

In order to attempt to satisfy a very critical listener who happens to be a rather exacting engineer by profession and an out-of-hours musician by hobby, but who knows what he likes when it comes to musical reproduction, a survey was made of the over-all reproduction problem. Seven years of test, exploration, and development followed. Not only was it necessary that it must not offend the eye, it was also required that the assembly should be capable of being moved from place to place as the engineering business dictates, a brick or stone enclosure thus being ruled out.

In casting about for a suitable device it is immediately apparent that there were but two basic design paths to follow; the horn-loaded unit, and the direct-radiator system.

The problem of power output was next considered. Measurements of the acoustic power levels of the various instruments were reviewed; a sound-level meter was used to survey typical levels encountered at choice locations at several Philadelphia Philharmonic and Lewisohn Stadium concerts.

Noise-level surveys at these concerts were also made and compared with typical home listening levels in order to assess more accurately a scale factor which would provide equal desired-signal-plus-background to background levels and hence comparable dynamic ranges. Consideration was given to the effect of monaural listening as compared to binaural listening and to the aural discrimination intrinsically present when sight is used to supplement sound.

With monaural listening it appeared that an acoustic level of one to two acoustic watts could do a satisfactory job, but that an additional 10 to 20 db of dynamic range would be desirable. The measured dynamic ranges of direct (live concert of the Philadelphia Symphony) performances were surprisingly low (presumably because of the averaging due to reverberation in the hall). Twenty to twenty-five db appeared to be a maximum.

This study provided initial clues as to the required optimum design trends. Early experience with theatre horns and their 20 to 25 per cent conversion efficiency, indicated that an amplifier with a clean 4.5 watts of electrical output would do a good job covering 600 to 1500 people. Typical direct radiator systems have conversion efficiencies of 3 to 16 per cent so the power amplifier must have between six and thirty electrical watts output to do the same job in a satisfactory manner.

From our point of view the low-efficiency, power-demanding, direct-radiator system is not good design. The lower efficiency units generally are that way either because of inadequate magnetic circuits, or as a result of diaphragm break-up—both being cases to be avoided if IM is to be kept to a low valve. The higher more efficient direct-radiator units such as the Altec 604C or the RCA LC-1A permit a good design compromise by only requiring a nominal six- to twelve-watt power amplifier which is relatively easy of attainment with low-voltage, low-cost power supplies and rather nominally rated and operated power output tubes.

Fig. 1. A simple yet neat appearance characterizes the author's speaker enclosure which provides for a 15-in. single-unit speaker mechanism.

Frequency Range

The next item for consideration was frequency range and the sound distribution patterns desired for average installations.

Many engineers and hobbyists rather indiscriminately add high-frequency and low-frequency units into a system and frequently do achieve some measure of frequency balance insofar as level as a

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function of frequency is concerned. However from the writer's point of view this procedure would be analogous to having an artist paint a background in one picture, the theme or object in another, and the high lights in a third. Then by hanging all three pictures side by side we should get the integrated Whole. Leonardo Da Vinci didn't do it —so we're not trying to do it either.

Considerable experience in the design of directional antennas (which may be likened to acoustic radiating systems) emphasized the distorted spatial distribution patterns that always occur when space separation exists between two sources of radiation even when these are in phase. Add some phase variation and the spatial pattern becomes all the more non-uniform.

From the foregoing it appeared that horns and multiple-unit systems were not too desirable and in fact critical listening tests by musicians and non-musicians always seemed to favor the single-source direct-radiator system when this was given A-B comparison tests with multiple-unit horn-type systems. During these tests the observers did not know which system was in operation. The almost universal response was that one system seemed "smoother" than the other.

One axiom in enclosure design is this: if when you touch the enclosure you can feel the low notes, it's not good!

G. A. Briggs of Wharfedale got around this difficulty by using a brick enclosure. John V. L. Hogan minimized this difficulty in an early (around 1936) WQXR receiver by heavy cross-bracing of the enclosure. The greater portability of the latter arrangement appealed to us and according this design procedure was followed.

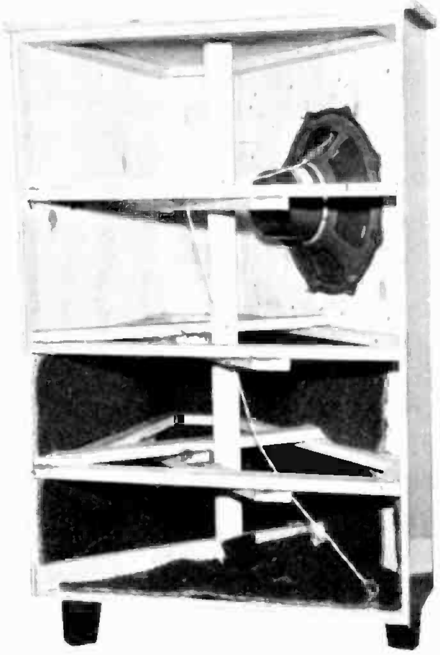

Fig. 3. Internal view with one back panel removed to show cross bracing and unsymmetrical mounting of the speaker. Also note start of Ozite lining in lower part of cabinet.

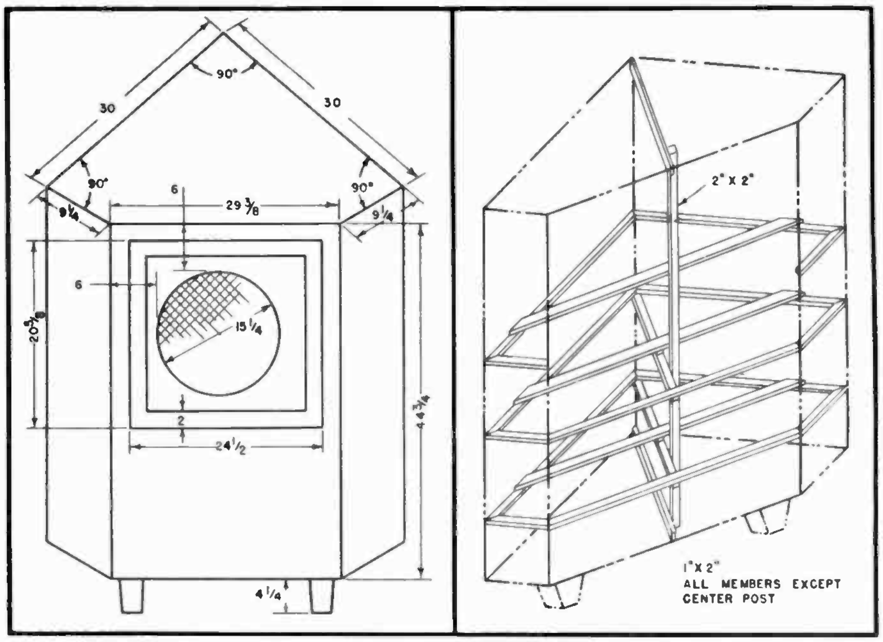

Fig. 2. Constructional details for the corner-mounted infinite baffle.

For our requirements a corner speaker held several desirable advantages. The corner design improves the radiation characteristics, as is well known; the triangular design permitted very rigid construction; the design easily permits asymmetrical speaker mounting and consequent avoidance of spurious peaks, and the system presented a minimum decoration problem insofar as integration into a living space was concerned.

With a low-frequency cone resonance of between 35 and 40 cps. and a desire to minimize the rate of roll-off below this resonance as well as to ensure smooth low-frequency curve immediately above it, a volume of approximately 15 cu. ft. was decided on, and a completely enclosed structure was considered essential. *Fig.* 1 is front view of the enclosure.

Construction

In order to fit snugly against the wall, three legs 4¼-in. high were used in order to raise the cabinet above the average baseboard. This is not too desirable because of the small resonance peak occasioned by the cavity formed between the enclosure and the floor. For this reason the legs are removable and we ultimately visualize installation of the enclosure against the ceiling.

The enclosure is made of ⅞-in. plywood with construction details as shown in *Fig.* 2. Assembly is with wood screws and casein glue with particular care taken to insure a rigid and air-tight assembly.

All peripheral and cross brace stiffeners were made of 1 × 2-in. fir. The top-to-bottom member was 2 × 2-in. fir tied into the three sets of interior cross braces by cemented-in blocks aided by wood screws.

The speaker mechanism chosen—the RCA LC-1A, more commonly known as the Olson speaker—is mounted asymmetrically in order to break up all reinforcing reflection paths which might otherwise degrade the response curve. The unit is mounted near the top of the cabinet in order to provide the optimum spatial distribution with the sound source as close to ear level as possible.

The interior is lined with ozite cemented on with rubber cement. Additional absorption was utilized by draping strips of ozite across the internal cross-bracing members as shown in *Figs.* 3 and 4.

The performance of this speaker mechanism in this particular cabinet is considered adequately rewarding for the work involved in designing and constructing the enclosure, and it is felt that the complete system is as good as the writer can make it—until the urge to redesign shall again arise to start the entire cycle over again.

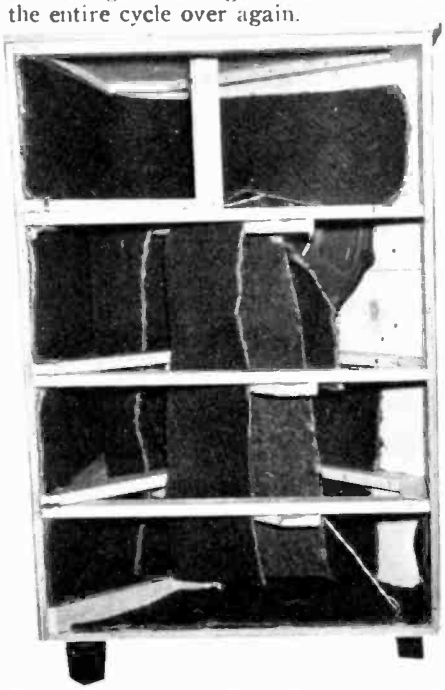

Fig. 4. Scrap strips of Ozite lining are draped over internal bracing to further dampen the cabinet.

Coupled Loudspeakers

CHARLES W. HARRISON, JR.*

A discussion of the principles of multicone speaker operation and a description of a composite corner-mounting assembly composed of four acoustic baffles of trapezoidal cross section.

THE EFFECTIVENESS of a sound source depends in an important way on the phase relationship between the normal velocity of the radiating surface and the force reaction (or sound pressure) of the medium on the surface. The principal value of a horn is that it will permit sound to be generated by the vibration of a small diaphragm, but radiated from an aperture large enough to keep pressure in phase with particle velocity down to relatively low frequencies. Horns of exponential or catenoidal shape are conformable to practical application.

It is customary, in calculating the throat impedance of a horn of finite length (assuming it to be baffled so that radiation is confined to 2π steradians), to replace the mouth of the horn by a massless diaphragm of suitable shape working in an infinite baffle. This diaphragm has no effect on the operation of the horn, but permits one (for mathematical purposes) to terminate the horn in an impedance equal to the radiation impedance of a piston operated in an infinite baffle. From the basic horn equation, in terms of the velocity potential, one obtains an expression for the pressure and particle velocity at any point within the horn. These expressions, together with the piston functions, permit determination of the throat impedance. The important thing is that the external sound field set up by the horn can be duplicated by a vibrating piston of comparable radiating area. The basic problem is that of obtaining an aperture large enough to keep the sound pressure and air-particle velocity in time phase down to low frequencies. For practical purposes the large piston radiator (a direct-radiator loudspeaker mechanism of suitable design) may be replaced by a system of properly phased tightly-coupled smaller drivers. However, it is to be remembered that in the useful frequency range of a horn, the input resistance at the throat is high and this affords additional damping over that available in the diaphragm suspension system of the horn driver. This is important at low frequencies if nonlinear effects are to be minimized that are generated when large diaphragm excursions take place. For comparable performance the multiple loudspeaker must employ more highly damped drivers than are used with a horn. Such drivers are inherently inefficient, but efficiency is of no real practical importance in the design of a high-quality loudspeaker for home use.

The question might now be asked as to why one would want to construct a loudspeaker array giving a performance similar to that afforded by an exponential or catenoidal horn. The answer is merely that the physical size of the comparable horn must be many times that of the coupled loudspeaker system. Unfortunately mouth reflections in horns give rise to air column resonances, and unless the mouth is comparable in dimension to the longest wave-length to be reproduced, tremendous variations in sound transmission must be tolerated. A second design criterion for finite horns is that the cut-off frequency of the corresponding "infinite" horn should be at least an octave below the lowest frequency in the desired transmission range if reasonably smooth response is to be obtained. When these requirements are translated into a horn capable of "properly" reproducing notes in the vicinity of 35 cps a speaker of gigantic dimensions is obtained. Even the theater woofers which utilize exponential horns approximately four feet long on axis are nothing more than directional baffles below 60 cps. This fact can be verified readily by reviewing the theory of the exponential horn found in any text book on acoustics. The only solution, if one insists on using a low-frequency horn, appears to be that of building one in the yard out of brick or concrete. Such a speaker should never be built out of plywood, for serious spurious responses are sure to be obtained from the "ringing" of the horn walls. Yet in spite of the large size of a *properly baffled* horn, its use in the theater is justified because efficiency and power handling ability for frequencies above about 50 cps are important factors. The multiple loudspeaker using well-damped drivers unquestionably affords the best method of getting good bass response in the home where space is restricted.

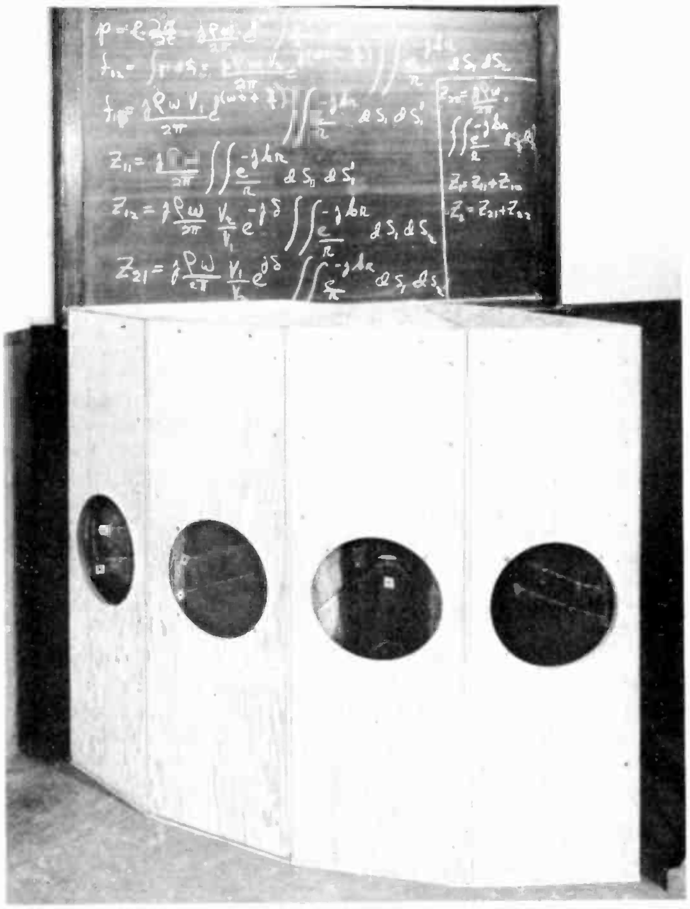

Fig. 1. Photograph of acoustic baffles employed in coupled loudspeaker tests.

* *Commander, U. S. Navy, 611 Massachusetts Ave., Norfolk 8, Virginia.*

The writer constructed a multiple loudspeaker consisting of identical wedge shaped acoustic baffles of such angular dimensions that four of them fit snugly into a 90-deg. corner. As shown in *Fig.* 1, each front panel is just wide enough to accommodate one direct-radiator loudspeaker mechanism mounted at its center. This insures tight acoustic coupling between adjacent drivers when the speaker enclosures are assembled in the corner of a room for operation. Notice that the radiating elements of the speaker resemble the mouth of a sectoral horn. In the following sections the reasons behind this design are set forth.

Radiation Resistance of Coupled Loudspeakers

It is well known that the radiation resistance (in mechanical ohms) of an isolated sound source whose dimensions are small compared to the wavelength of the radiated sound is proportional to the square of the effective radiating area. Furthermore, the value of this resistance is independent of the physical shape of the sound source. Applying this principle to the multiple loudspeaker under discussion, one concludes that when the multiple loudspeaker is isolated from all surroundings (no longer operated in a corner) that the radiation resistance of the four-speaker array at low frequencies is about 16 times the radiation resistance of one of the drivers comprising the array in the absence of acoustic coupling. It is clear, therefore, that the radiation resistance of *each* driver when operated in multiple is increased by a factor of 4 over that obtainable from the same driver operated as an isolated speaker. It is to be emphasized that the physical orientation of the drivers in a baffle which is small compared to the wavelength of the radiated sound is not important insofar as the radiation resistance of the array is concerned. This fact permits location of the drivers in a suitable spatial relationship to insure a desirable sound pressure pattern. If the multiple loudspeaker is now located in the corner of three large mutually perpendicular rigid planes, the radiation is confined to a solid angle of $\pi/2$ steradians, and this "horn" loading further enhances the radiation resistance of the array. But such a situation is physically unrealizable in an enclosed space (excluding an anechoic chamber) and the idea of sound transmission in a solid angle of $\pi/2$ steradians must give way to a rigorous analysis of sound transmission phenomena in rooms.

Possibly a more elegant way of demonstrating qualitatively that the radiation resistance of a multiple loudspeaker is proportional to the square of the effective diaphragm area at low frequencies is to comment briefly on the theory of the circular piston radiator mounted in an infinite baffle. This problem is, discussed in a straightforward manner in a recently published excellent book in the field of acoustics.[1] One finds the reaction force on one elemental area of the piston due to the motion of another elemental area. By a process of summation (integration) one finds the force on a given elemental area due to the motion of all other elements. Inclusion of all of these elemental areas on which the force has been computed by a second integration (taking due account of the number of times a given element is included in the calculation) gives the total force acting on the diaphragm. This force is equal and opposite to the force exerted on the fluid medium by the piston. The latter force, divided by the cone velocity, gives the mechanical radiation impedance of the speaker. As might be expected, the radiation resistance (in mechanical ohms) turns out to be proportional to the square of the effective diaphragm area at low frequencies.

The analysis has been outlined because it brings out two essential facts pertinent to this discussion:

(a) The diaphragm of any loudspeaker may be considered to be made up of a number of suitably shaped smaller diaphragms.
(b) The interaction or coupling between these smaller diaphragms has been properly taken into account in the analysis.

Statements (a) and (b), together with the theory of the piston radiator outlined here. permit one to draw correctly the conclusion that the radiation resistance of a tightly coupled multiple speaker system installed in an infinite baffle is proportional to the square of the effective diaphragm area at low frequencies. Naively expressed, a person might say that the radiation resistance of a multiple loudspeaker is greater than that of one of its drivers mounted in an enclosure because the drivers aid each other in compressing the air in the vicinity of the cones, causing each driver to "think" that it is working into a more dense fluid medium. A more elaborate and quantitative analysis of coupled loudspeakers, based on the principle of symmetrical phase components, has been published.[2]

It now seems pertinent to discuss briefly another aspect of statement (a) that the diaphragm of any loudspeaker may be considered to be made up of a number of suitably shaped smaller diaphragms. In a recent article[3] it is strongly implied that two low-frequency speakers are highly satisfactory, but four such speakers are taboo because "they present a problem of phasing." The writer is inclined to divide (hypothetically) the two diaphragms that it is "permissible" to use into quadrants. Eight sector-shaped diaphragms are obtained. Is it to be supposed that an intolerable phasing problem has now been encountered? The writer thinks not. To throw more light on this kind of fallacious reasoning consider the following numerical illustration: Suppose that a phase-conscious observer (if one can be found) is located some distance from a piston radiator mounted in an infinite baffle, and that the line from his ear to the center of the diaphragm never makes an angle less than 30 deg. with respect to the baffle plane. Further assume that the observer does not wish complete destructive interference to obtain at any frequency in the range from 60 to 10,000 cps. Surely the "phase problem" is at its worst at null points in the angular dispersion pattern. Reflections from all objects are ignored, and it is assumed that the velocity of sound is 344 meters per second. Using these data one finds that if the "phasing problem" is to be avoided—even when using a single piston radiator, the diameter of the loudspeaker must not exceed 1.9 inches. It is obvious that one may ignore the radiation potentialities of this loudspeaker at 60 cps. If the general problem of phase is really of significance (except as discussed later in this paper) one might very well conclude that

(a) Multiple microphone pickup of any program is to be avoided if the outputs of these microphones are to be electronically mixed and transmitted over a single channel.
(b) Efforts to obtain a diffuse sound field, i.e., a sound field of random phase, in a studio or auditorium by the use of splays and asymmetrically placed patches of absorbent material, should be abandoned.

The writer has attempted to point out in this section that a multicone loudspeaker is capable of excellent bass response (by virtue of the fact that at low frequencies the mechanical radiation resistance is proportional to the square of the effective diaphragm area), and that an array of speakers presents precisely the same problem with regard to phase as any direct-radiator loudspeaker of small diameter, or any other vibrating body. No horn is immune from the "phase problem" for the mouth of the horn may be closed by a massless diaphragm with no deleterious effect on the performance of the horn. Before concluding this topic the writer would like to suggest a problem of considerable theoretical interest concerning the coupled loudspeaker system under discussion. Suppose one were interested in calculating the radiation impedance on the mechanical side of one of the drivers forming the four-element array, assuming the array to be operated in the corner of three perfectly rigid mutually perpendicular semi-infinite planes under free field conditions. One equivalent mathematical model consists of two suitably oriented symmetrically disposed speakers driving an infinite wedge. The sides of the wedge are linearly tapered,

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[1] L. E. Kinsler and A. R. Frey, *"Fundamentals of Acoustics"*, John Wiley & Sons, Inc., New York, 1950, pp. 187–195. No audio engineer should be without this book.

[2] S. J. Klapman: "Interaction impedance of a system of circular pistons," *J. Acous. Soc. Am.*, Vol. 11, Jan. 1940, pp. 289–295.

[3] W. C. Shrader, "Audio in the home," AUDIO ENGINEERING, July 1952, page 30.

and if extended would meet at an angle of 22.5 degrees. The height is twice the height of an enclosure. The top and bottom (which are missing) have the cross-section of an isosceles trapezoid. The basic problem is to find an expression for the acoustic pressure in the vicinity of a driver satisfying the wave equation and all boundary conditions. This is an advanced problem in mathematical physics.

Operation of a Loudspeaker in the Corner of a Rectangular Room

Probably the simplest boundary-value problem in applied science is the determination of the acoustic pressure distribution in a rectangular enclosure having six perfectly rigid sides under free oscillatory conditions. The wave equation in terms of the velocity potential is separated in cartesian coordinates. The acoustic pressure is equal to the density of air multiplied by the partial derivative of the velocity potential with respect to time. The air particle velocity is equal to the negative gradient of the velocity potential. The boundary conditions are that the normal component of particle velocity at all bounding surfaces of the room must vanish. One immediately obtains an expression which shows that the acoustic pressure is always a maximum in the corners of the room regardless of the room dimensions and the order of the mode of free oscillation. However, it is to be noticed that this analysis says nothing about how the normal modes of oscillation are established, and in such a room there would be no decay of sound, the vibrations continuing forever. It turns out that when the walls of an enclosure are sound absorptive, a perturbation of the simple theory just advanced enables one to compute the pressure distribution in the room for an individual mode during the transient decay of the sound field (no sound energy supplied). The pressure distribution for each mode is still a maximum in the corners of the enclosure.

Now when a "cavity resonator" bounded by sound-absorptive walls is driven by a simple source of sound, one proceeds as follows to determine the sound pressure distribution in the room:[4]

(a) The form of the wave equation used must permit the injection of sound into the fluid medium by the simple source.

(b) The flux of air from the source can be represented by a source function, and this function can in turn be expanded in characteristic functions at the source location (a point anywhere in the room).

(c) The steady state sound pressure at any point in the room can be similarly expanded in series.

By using certain information obtained from the problem of the transient decay of sound in an absorptive room, and (a) through (c), one can evaluate unknown coefficients. A little mathematical manipulation then enables one to arrive at an expression for the sound pressure at any point in the room. The next step is to move the sound source analytically into a corner. It will be found that the pressure at any point in the room has been maximized. However, and this is the point of this entire discussion, *a max. max. value of sound pressure is obtained when the sound source is located in a corner, and the sound pressure is computed at the same point.*

The force of the fluid medium on the diaphragm of the point source is the vector sum of the pressures at this point due to all possible modes multiplied by the surface area of the vibrator. As mentioned before, the negative of this force, divided by the cone velocity, gives the radiation impedance in mechanical ohms. The real part of this impedance is the radiation resistance. The larger the acoustic pressure reacting on the cone the larger is this resistance. It is apparent, therefore, that *for maximum cone loading under steady state conditions the speaker should be designed for corner operation.* For practical purposes cone loading is increased only for sustained low frequency notes. For transient signals it would appear that the use of a corner for speaker location would be no more beneficial than locating the speaker anywhere else in the room.

There is one important advantage of locating a speaker in the corner of a room in addition to the increase in cone loading obtained on sustained low frequency tones. Since all of the normal modes of a room have pressure maxima in the corners, a speaker so located will excite all possible modes. These modes are few and far between (with respect to frequency) at very low frequencies, and it is considered important to excite all of them in the interest of "uniformity" of transmission.

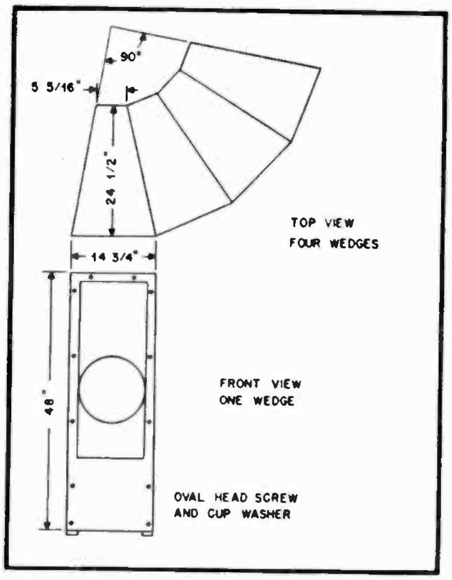

Fig. 2. Scaled drawing of multiple loudspeaker.

Under steady-state conditions, a point source of sound gives rise to a standing-wave pattern having nodes and antinodes at fixed locations within a room. Two or more such sources separated a small distance and operated simultaneously tend to fill in nulls in the sound pressure pattern. The four-driver loudspeaker array may be considered a point or simple source of sound only at low frequencies. When the frequency is increased somewhat the point source must be replaced by the appropriate distributed sound source. The large aperture of this speaker makes it rather unlikely that objectionable pressure minima will exist anywhere within the room.

Sound Pressure Pattern

Although the intensity of sound in a room depends greatly on the properties of the enclosed space, most of the experts agree that while the ear tolerates a certain degree of non-uniformity, a loudspeaker having a smooth response under free field conditions will generally be more acceptable under all listening conditions.[5] Consider the loudspeaker array shown in *Fig.* 1, to be oriented in the corner formed by the intersection of three mutually perpendicular perfectly rigid semi-infinite planes. Assume also that the adjacent walls of the wedge shaped enclosures are perfectly rigid (and thus non-absorptive). The equivalent mathematical model of this system is a perfectly symmetrical circular array composed of 16 wedges not displaced in length. Each wedge is twice as high as a wedge comprising the physical loudspeaker system. Two drivers are mounted in each wedge, one-fourth the way down from each end in the front panels. The mathematical model dispenses entirely with the three semi-infinite planes. It is easy to see intuitively that at low frequencies the acoustic pressure pattern in the azimuth plane[6] is essentially uniform since there is an angle of only 22.5 deg. between adjacent speaker axes. To maintain this uniform pattern throughout the frequency range of interest requires the use of drivers having a beam width of more than 45 deg. at the highest frequency in the range to be reproduced. The pattern of the array in the azimuth plane can be computed easily by graphical methods. One simply adds up vectorially, i.e., in proper phase relationship, the sound pressures contributed by each driver in the angular direction of a given point. This process is repeated at

[4] A good reference is P. M. Morse and R. H. Bolt "Sound waves in rooms", *Review of Modern Physics,* Vol. 16, No. 2 April 1944, pp. 69–150.

[5] H. F. Hopkins and C. R. Keith, "New loudspeaker system" *J. Soc. Mot. Pict. Eng.,* 51, pp. 385–398, Oct. 1948.

[6] The azimuth plane is defined as a plane midway between and parallel to the two planes that individually contain the centers of 16 speakers. This plane is at right angles to the axis of the circular array. The vertical plane includes the array axis and the centers of four drivers—two in one wedge, and two in the diametrically opposite wedge.

other points until the array pattern has been constructed. Obviously one must be furnished the primary pressure distribution patterns of the drivers taken under free field conditions.

It is not at all difficult to compute the pattern in the vertical plane, again knowing the primary patterns of the drivers used. More beaming takes place in the vertical plane. The writer did not use two drivers in each wedge (one above another) because the sound pressure pattern in the vertical plane would become too narrow even at relatively low frequencies, and the perfect symmetry of the array would be destroyed, as will be explained.

The multiple loudspeaker described in this article can be used with or without a tweeter. If four 12-inch drivers, such as the Western Electric type 728B or 754A are used, an excellent loudspeaker is obtained without the use of a high-frequency unit. For the initial tests the writer employed four WE 728B drivers in the wedges, and a WE 594A loudspeaking telephone with 31A horn for treble. The cross-over frequency is 800 cps. One important asset possessed by this dual loudspeaker is that the sound pressure pattern in the azimuth plane for both the tweeter and woofer join smoothly at cross-over. Persons employing, for example, folded horns for bass like to forget about the problems arising from phase differences near the crossover frequency, abrupt discontinuities in the distribution pattern at crossover, and the phase difficulties associated with the sharp bends even at fairly low frequencies. None of these difficulties plagues the speaker described in this article.

Another feature of the coupled loudspeaker is its excellent mid-range response, i.e., response in the range of frequencies from say 300 to 800 cps. Apparently a folded horn works well only in the region from 30 to 300 cps. This necessitates crossing over at about 300 cps. But this is undesirable because it has been found an advantage, from the listening point of view, to have any effects due to out-of-phase conditions between the low- and high-frequency speakers come above the region of maximum energy transmission.[5] Evidently the use of a three-way speaker system, employing a folded horn for bass, does not obviate this difficulty.

On the Choice of Acoustic Baffle Shapes for Multiple Loudspeakers

Acoustic baffles having isosceles trapezoidal cross section have several highly desirable attributes. As pointed out in the last section, if four identical baffles are used, the angle between adjacent speaker axes is only 22.5 deg., and if only one driver is installed in each baffle excellent angular dispersion of sound is assured. The choice of wedge-shaped baffles is important for other reasons. For one thing this array is symmetrical. If the four wedges are assembled in a corner formed by the floor and two rigid walls, the non-parallel sides of the enclosures cannot vibrate because there are no force differentials acting on them. The front panels are just wide enough to accommodate the drivers. These can be adequately braced. The back panel is only several inches in width. If it is properly secured to the non-parallel sides its vibration can be ignored. The top and bottom of each enclosure is small in area. The bottom certainly will not vibrate because of the load on it, and the top can be "held down" by a heavy tweeter.

The sides of a rectangular enclosure for several drivers will necessarily vibrate because the array lacks symmetry. This can be minimized only by construction of an enclosure of extreme rigidity.

Because of the symmetry of the wedge enclosures one can be certain that if identical drivers are installed in the four baffles that what one diaphragm does will be necessarily duplicated by the others at all frequencies. It does not follow that when four or more drivers (possibly of different sizes) are installed in a common rectangular enclosure that all cones will go in and out together at all frequencies. The mechanical constants of the diaphragm suspension system, the mass of the diaphragm and driving voice coil, the stiffness of the enclosures, and the coefficient of acoustic coupling between drivers may be of such value as to cause one diaphragm to move in irrespective of the fact that the polarity of the voltage applied to the driver terminals would normally cause the diaphragm to move out. This is not a new concept. In antenna array design it is not uncommon to measure a negative resistance at the terminals of one of the dipoles comprising the array. To achieve stability, i.e., minimize diaphragm slip, all multiple-loudspeaker enclosures should be compartmentalized and the drivers employed should be highly damped. The fact that highly damped identical drivers are used in the writer's multiple loudspeaker and that none of them is operated in a common enclosure insures that to within a few degrees all diaphragms do go in and out together. Observe that the use of wedge shaped enclosures is not optional. The symmetry afforded by this construction requires the acoustic forces acting on each diaphragm to be the same. These forces are not equal, for example, even when four drivers are operated in a compartmentalized rectangular enclosure. This fact may be verified immediately by applying the principle of acoustic images. It is a serious matter if any out-of-phase condition exists between diaphragms, because the chief purpose of a multiple loudspeaker is thereby lost; namely, that of achieving a good low-frequency performance. The phase relationship of the vibrating diaphragms can be checked by stroboscopic means, or by measuring the sound pressure pattern of the speaker in an anechoic chamber.

At first glance it might appear that a baffle of trapezoidal cross section lined with sound absorptive material would be more effective than the corresponding baffle of rectangular shape in damping out standing waves within the enclosure. Unfortunately, normal modes exist for trapezoidal shaped baffles as they do for rectangular, spherical, and other-shaped enclosures. The solution to this problem is to fill the wedge with small rectangular parallelepiped shaped blocks of fiberglas using strong perforated curtain material as a retaining wall to prevent the fiberglas from pressing on the driver diaphragm. The normal modes will now be damped out by the viscous drag of air particles oscillating in the small pores of the fiberglas and other absorbing material present. In addition the effective volume of the enclosure will be increased by the use of fiberglas because it can be demonstrated that a sound wave is propagated more slowly in fiberglas than in air. For a given frequency the wavelength of sound radiated from the back side of the diaphragm into the fiberglas filled enclosure is shortened; accordingly the inside dimensions of the baffle, in terms of the wavelength, are increased. The absorption of sound in a fiberglas filled enclosure is approximately an isothermal rather than adiabatic process. At low frequencies the effective volume of the enclosure is about 1.4 times the actual volume. At higher frequencies essentially free field conditions obtain on the back side of the diaphragm because no acoustic waves are reflected. Thus the enclosure may be regarded as one of infinite size. The impedance seen by the driver, looking into the enclosure, is the characteristic resistance of air. It is worth noting that at low frequencies very little power is radiated from the backside of the cone into the enclosure where it must be absorbed. This is because the baffle is small in terms of the wavelength, and the acoustic pressure is very nearly in time phase quadrature with respect to the particle velocity. Thus at low frequencies an enclosure may be represented as a lumped compliance (purely reactive element) in the equivalent acoustical circuit; a fact that is well known.

Construction of Multiple Loudspeakers

Baffles having the cross section of an isosceles triangle will serve equally as well as those having the cross section of an isosceles trapezoid. The angles of each baffle are fixed by the requirement that four baffles must fit side by side in a corner. This fixes the angle between adjacent speaker axes at 22.5 deg. If three enclosures are employed this angle is 30 deg. which is probably excessive from the point of view of obtaining a good azimuth pattern at relatively high frequencies. If this three-speaker array is used as a woofer only, the 30-deg. angle between speaker axes is probably not too objectionable.

Driver specifications should be followed with regard to the required volume of each enclosure. Where the speaker system is used without a tweeter, the writer has found satisfaction with Western Electric Type 728B or 754A drivers. A smaller version of this multiple speaker could be made up using Western Electric Type 755A 8-inch drivers. Such a speaker would be about optimum for home use. Because of the 755's extended high-frequency response, no tweeters would be needed. Where the multiple speaker is used as a woofer only, experience has shown the Bozak B-199 to be a sound choice due to its low resonant frequency and correct damping.

The enclosures must be completely lined with Kimsul or other sound absorptive material. With certain drivers it is an advantage to fill the baffles completely with blocks of fiberglas 1 × 2 × 4 inches in dimensions. Type PF-314 fiberglas is recommended. The blocks of fiberglas will adhere to the Kimsul, thus preventing packing. Alternatively, to prevent packing the enclosure may be compartmentalized using cloth retaining walls. This treatment smooths out the speaker response, i.e., increases the speaker damping at low frequencies, and increases the effective volume of the baffle. It is recommended that fiberglas treatment be kept well clear of the diaphragm of the driver. The enclosures should be essentially air tight and no ports are permitted. The design should be such that the front panel is just wide enough to accommodate the driver mounted in the center. This insures close coupling of the drivers at low frequencies. The baffles will have to be approximately 4 ft. high in order to obtain the requisite volume. When they are set side by side and then jammed into a corner, excellent baffle effect is obtained. The polarity of each driver should be checked, using a small dry cell, before it is installed in its baffle. Naturally, the speakers are to be connected electrically so that all diaphragms operate in the same phase.

The series or series-parallel connection is recommended. It is worth mentioning that if four identical drivers having a nominal input impedance of 8 ohms are connected in series the nominal input impedance of the array is increased from 32 ohms to a somewhat higher value by virtue of acoustic coupling between drivers. This effect is not very pronounced when highly damped inefficient drivers are employed in a multiple loudspeaker, but is to be kept in mind when matching a highly efficient horn loaded driver to an amplifier. Readers are advised to conduct listening tests as an aid in deciding the best driver connections and the proper amplifier load taps. If incorrectly damped drivers are used, hangover effects are likely to be observed. This difficulty may be minimized or eliminated entirely by making a large felt washer for each driver with center hole of sufficient size to slip over the magnet frame. The periphery of the washer is secured to the front panel with carpet tacks. The air set in motion by the back side of the diaphragm must pass through the pores of the felt. Driver damping becomes, therefore, a function of the porosity and thickness of the felt used.

The baffles for the writer's multiple loudspeaker were made out of ¾-in. plywood. The *outside* dimensions of each wedge are as follows:

Height—48 in.
Width of front panel—14¾ in.
Width of back panel—5-5/16 in.
Slant depth—24½ in.

These figures do not include the dimensions of the front panel frame.

A scaled drawing of the multiple loudspeaker is shown in *Fig. 2*.

The front panel of each wedge is removable, the other five sides being permanently secured. In the interest of obtaining a finished appearance it is desirable to construct frames to fit over the front panels as suggested by the drawing. A brass wire screen measuring about 13 × 36 in. should be stapled to the front panel to prevent accidental damage to the speaker cone, and then a suitable grill cloth tacked in place. If the frames have been properly recessed they will fit snugly over the edges of the grill cloth and wire. The frames may be secured to the front panels using small oval headed brass screws and brass cup washers that have been blackened by chemical treatment. Observe that different screws are used to secure the front panel to the enclosure and the frame to the front panel.

If wedges of trapezoidal cross section are constructed an air column will exist between the speaker and corner of the room. For enclosures 4 ft. high, resonance of this air column will occur at about 70 cps, and again at about 211 cps, etc. It should be filled with sound absorptive material. In making this elementary calculation no end correction for the pipe was applied.

Acknowledgment

The writer is indebted to Mr. H. F. Hopkins of the Bell Telephone Laboratories for making available certain information concerning multiple-speaker experiments carried out at the Murray Hill Laboratory.

The significant comments on the use of fiberglas in an acoustic baffle to improve speaker performance are due to Professor Jordan J. Baruch of the Massachusetts Institute of Technology.

Messrs. Theodore John Schultz, John Bouyoucos and A. A. Janszen of the Acoustics Laboratory, Harvard University, reviewed the manuscript and made useful suggestions for improvement.

The Concrete Monster

JAMES FERGUSON*

It could only happen in California—an exponential horn built into the side of a house! It's a neat job and must sound fine. We envy his originality—as well as his apparent disregard for domestic relations.

One of the most important links in the audio chain, and often the weakest, is the speaker system, especially the speaker enclosure or baffle. Radio and audio magazines devote frequent articles to them. Many preface the test by saying that the exponential horn is the best and most efficient method of transferring acoustical energy from the cone or driver to the free air in a room. Its high efficiency reduces distortion by not requiring so high a setting of the volume control for a given sound level, and smaller excursions of the voice coil keep it in the linear portion of its travel.

The articles say that the mid-range and high frequencies offer no space problems for horns, but when it comes to the low bass the horn's mammoth size (a length of some 16 feet and mouth diameter of 10 feet) necessary to reproduce properly organ pedal notes and other bass down to 30 cps or lower simply rule it out. The little woman would not tolerate such a monster in the living room, even if the hi-fi bug would. So the articles offer you something less bulky which sounds less well at the bass end. When you compromise on size, you often compromise on quality.

After considering this knotty problem for some time and building several enclosures which promised "big bass" performance in moderate size spaces, I still was not satisfied. So I decided to build an exponential horn outside the house, and merely let it poke its mouth into the room. And for the benefit of the many perfectionists who read this magazine, the following is a brief outline of the system I have constructed.

I used three speakers, all exponential-horn-loaded, and two dividing networks crossing over at 200 and 3,500 cps. The "listening-room" is a combination living-

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and dining-room (*Fig.* 1), 31 feet long and 18 feet wide, narrowing to 12 feet in the dining section. Looking from the living-section into the dining-section, the cabinet containing the mid-range horn and four-in-a-row high-frequency horns, *Fig.* 2, is located in the right corner under a window.

This cabinet is shaped like a cube 2 feet square and 27 inches high, with the projecting corner sliced off diagonally so that its face is 21 inches wide. The lower 20 inches is occupied by the mid-range horn and back-of-cone air chamber which surrounds it. It has a throat area of 50 square inches which doubles every 6 inches of horn length, being 400 square inches at the mouth. It is driven by a 12-inch Wharfedale speaker which has a porous cloth cone suspension. The four-cell high-frequency horn (each horn with 4-inch-square mouth and 10-inch length) is located in the upper part of the cabinet with room for planters on each side. It is powered by a Stephens P-15 high-frequency driver.

The sound from the big bass horn enters the room in the upper left corner of *Fig.* 1 through an opening 4 feet wide and 5 feet high, *Fig.* 3, with its top at the ceiling. The angle of ceiling and walls forms the last 5 or 6 feet of the horn's length, and its effective mouth diameter at this point is approximately 10 feet. The large opening has in no way detracted from the room's appearance inasmuch as the grill cloth covers not only the opening, but the rest of the wall above a dado and over to the corner window. The opening fills the space to the left of the mirror shown in the picture. An uninformed visitor would never suspect the presence of the horn's mouth from inside the room.

The distance of a few feet that separates the bass and mid-range treble horn gives a spatial effect to orchestras which is very pleasing, if not third-dimensional.

The Horn

Now for the big horn and air chamber (which is not built of wood, but of steel and concrete) illustrated in *Figs.* 4 and 5. The speaker, a 15-inch Stephens 103LX "woofer" with low-resonance cone, is mounted underground in a 55-gallon steel drum—the type that has a removable head—and is sealed with a gasket and ring clamp. The speaker is mounted on a ¾-in. plywood ring which is fastened to the underside of the metal drumhead. In addition, a second wood ring of smaller diameter was used to fill in the space between the mounting board and the head, due to the curvature of the head.

A 10-inch circular hole was cut in the metal head, and ½ inch was turned up for a flange to receive the first section of the horn. That made the opening 11 inches in diameter and the hole in the wooden rings was tapered from this size to 13 inches next to the speaker. The air-tight cavity of the drum serves to balance the impedance of the back side of the cone to the horn-loaded front side. It also prevents the sound of the back-wave from being radiated to the neighbors.

Incidentally, the sound from the entire horn outside the house is hardly audible from a few feet distance, even when it is delivering considerable volume in the house.

The drum was partially filled with large stones to reduce its volume to approximately 4½ cubic feet. The stones break up the space into several small cavities of varying sizes and shapes, producing a distributed resonance, rather than one large peak, as would be the case with one large cylindrical cavity.

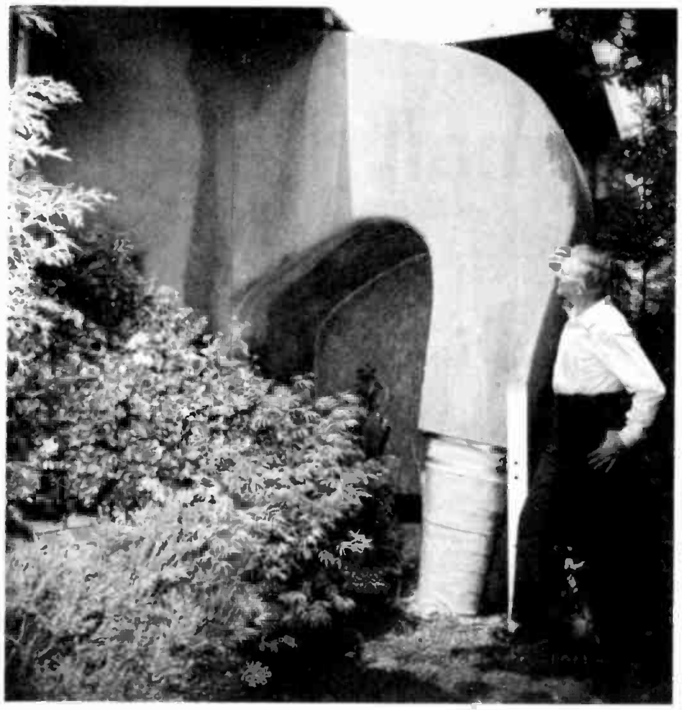
The horn starts from the drumhead at ground level and emerges into the opening in the wall, making about a 95-degree turn, and having an actual mean length of about 11 feet (though its effective length, including the mouth formed by the walls and ceiling, is over 16 feet). The first 3 feet, forming the throat, was made of 22-gauge galvanized steel in two sections. It was coated on the outside with asbestos-filled pitch or asphalt, and wrapped with a solid layer of binder twine. Then it was given another coat of pitch and wound spirally with a 6 inch strip of water-proof crepe paper as shown in *Fig.* 4. This provides excellent damping of spurious vibrations in the metal walls. The metal portion of the horn telescopes into the next, concrete, part for removal of the speaker.

The concrete horn proper was made as follows, built up as shown in *Fig.* 6. Quarter-inch round iron bars were bent to form the corners, then square frames made of the same material were placed along the length of the frames at about 2-foot intervals and tied together with wire where they intersected. Not shown are some extra rods which were added in the broad parts of the frame; these rods were spaced about 12 inches apart in each direction. Also not shown are some diagonal bracings of ⅜-inch rods. The frame was next covered with heavy-gauge galvanized metal lath, which was secured to the frame with "hairpins" of 18-gauge galvanized wire, spaced about 10 inches apart and twisted.

Fig. 2. Closeup of the mid- and high-frequency speaker section in the corner cabinet.

Fig. 1. The author's dining room, which is part of the living room. Midrange and treble speakers are in the small cabinet beneath the window at the right, while the horn opens into the wall at upper left.

Fig. 3. Mouth of the low-frequency horn opens into the listening area through a grille-cloth-covered port approximately 48 by 60 in.



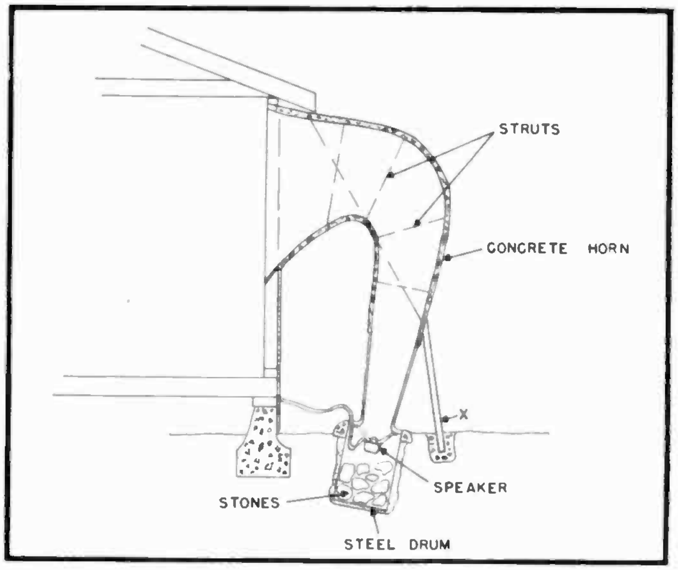


Fig. 4 (left). The author with his externally located exponential horn. Wonder what the neighbors think it is. Fig. 5 (right). Cross section of horn structure with relation to the dining room wall. "X" indicates the $2 \times 2 \times 1/4$ angle irons embedded in concrete blocks to support the structure.

Next, the opening was made in the wall by removing the stucco from the outside and lath and plaster from the inside, exposing two studs passing through the opening. One was removed for the length of the opening for easy access during construction, and the other "airstream" shaped and left in place.

(I must confess here that I was reluctant to tell my wife of my intention to cut a barn-door size hole in her dining-room wall, so I maneuvered her into an over-night visit to her daughter in a nearby city. When she returned home, the deed was done and it was too late for her to object. It seems that many wives don't share their husbands' enthusiasm for hi-fi.)

The frame, or cage, was supported in place and a coat of cement mortar (composed of one part plastic cement and three parts sand—to which was added a water-proofing and hardening compound, such as Anti-hydro, Sealcrete, etc.) was trowelled onto the metal lath. After this had set for 48 hours, more coats were trowelled on the inside and outside until a thickness of 2 inches or more had been built up. Each coat was scratched or roughened so the next would adhere firmly, and the last outside coat was "floated" with a sponge-rubber float. The inside was trowelled smooth. A metal ring was used at the small end of the frame, which acted as a screed[1] for plastering, and to make a neat ending for the concrete portion of the horn.

The horn has an expansion rate which doubles in cross-sectional area every 27 inches of horn length, beginning with the 11-inch circular opening in the drumhead. The sheet-metal section, *Fig. 7*, makes a transition from round at the small end to square (with rounded corners) at the beginning of the concrete part. After the concrete had thoroughly dried out, the inside was coated with a sealer and then with gloss lacquer to make the surface more reflective.

The opening into the room was covered with closely woven white nylon cloth about 1 inch back of the grill cloth (or at the stud line), and which was used to prevent a "dark look" at the opening. It doesn't seem to make any difference in the sound. An open-mesh drapery material was used for the grill cloth.

I drive the speaker system with a home-built ultra-linear amplifier, using a Fisher audio control, with phono, tape recorder, and an AM-FM tuner as sound sources. TV sound comes from a 630-type television set.

And how does it sound? Well, in previous articles on speakers and enclosures, the superlatives have been pretty well used up, but I've been listening quite a while and this is the best I have heard yet on the low end. The highs are good, too. ●

[1] Screed: A strip used to gauge the thickness of the plastering.

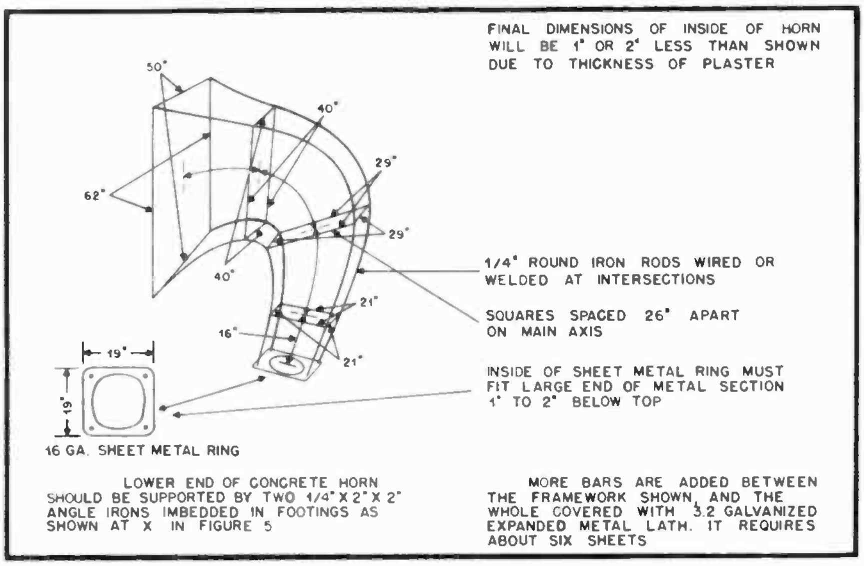


Fig. 6. Basic framework of 1/4-in. rods used as reinforcement for the horn. Framework is covered with metal lath before cementing.

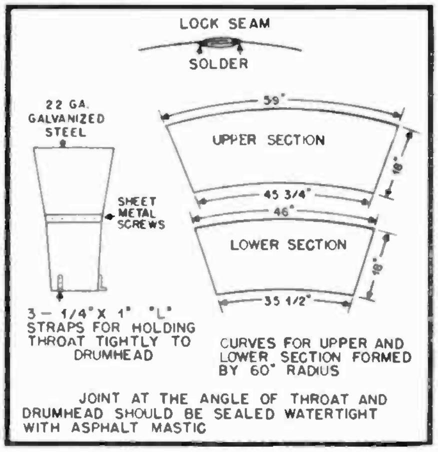


Fig. 7. Patterns for the construction of the telescoping throat end of the exponential horn.

Real Theater Sound in a Small Package

THOMAS R. HUGHES*

Part 1. The author, after years of analysis and desire, has reshuffled old facts and ideas to make a fresh approach.

To those who have followed the articles on loudspeakers and enclosures in radio magazines it must seem that the subject has long been exhausted. There are as many different suggested ways of enclosing speakers and as many extravagant claims for the set-ups as there are pages in the dictionary but we know that in many cases our ears and memory are just deceiving us.

The ears, after a good night's rest, are too ready to tolerate anything in the musical sound that is stimulating and different. But the very divergence of design and choice in speaker systems is proof that no satisfactory standard has been achieved.

The speaker system we have developed solves, for the first time, all of the speaker problems connected with high-quality music reproduction in the average size living room. This has been accomplished by a departure from the rut designers have been in for several years—that of trying to adapt theater-type speakers to living room use.

In order that you will not assume this is just another speaker enclosure that happened to charm the ears of its master, we ask you to bear with us in a thorough discussion of the principles involved, before we finally give you the details of construction and accomplishment.

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Our objective is to reproduce (as accurately as is physically possible) fine music, of all types, in the average size home. The problems which have to be solved for most of us are: the high cost of available systems; their large bulk and difficulty of fitting into the decorative scheme; the problem of having to operate recommended systems at too high a volume level in order to achieve the effect called "presence"; the problem of obtaining life-like definition of instruments and effects when the musical score explores a polyphony of superimposed parts and melodies; and lastly the problem of extra amplifier capacity required for the available high-quality systems.

Engineers conquered the problems of adequately reproducing fine music in the theater many years ago, but there has been a lack of realistic approach to the needs of the normal home listener. Having played in both symphony and dance orchestras in younger years and being an engineer by profession, the writer has been struggling with these problems for many of his older years. Having dozens of music-loving acquaintances who would be in the market for the proper article, if it were available, we determined to develop a stock item that could be moved into any room without disrupting it. Since the problems are so interwoven, we will not take up their solution in any regular sequence but will refer to them at relevant points.

Fig. 1. The completed three-horn corner reproducer in a typical living-room setting.

A Fresh Approach

The solution of all our problems starts with the proper generation of the lower frequencies and here is where we part company with the designers and authorities. We are not so interested in a flat response down to 40 or 50 cps because musical scores rarely call for fundamentals so low, nor do orchestras or ears provide a flat response. What we demand is the clear generation of the full harmonic structure of the cello, with its rugged vigor, or that of the bassoon, with its alluring intensity, and the jungle vibrance of the marimba bass notes.

Someone has said that most speaker designers listen to music through their slide rules. However, the effects we are interested in can not be distinguished by instruments in a laboratory. Furthermore, they are difficult to achieve in the home with normal speakers. So we will give you our specifications for simple solution.

In spite of the fact that many folded-horn corner-speaker enclosures are being built for home use with 15-in. woofers, the sales points for a large woofer are only applicable when it is used in a box or baffle enclosure. And it should never be used for symphonies in an average size living room.

Where it is used in a bass-reflex cabinet or any other box resonator, the prime objective is to have the natural lower cone resonance fall below the cutoff point of the program material. This can only be accomplished with a large cone of careful design. But, with horn loading of our woofer, we are not much concerned with resonant points because a horn levels them out to a great extent and still maintains acoustical coupling below these points.

On the other hand, there are many factors against the use of the large woofer in a home installation. The most important one is that, when it is worked at its intended amplitude, it produces outraged cries from family and neighbors. You don't get the robust timbre of bass instruments or singers from a large speaker that is just coasting along at comfortable room level.

Another factor against the large woofer is its cost. If you are determined to hear and feel the impact of that bass drum beat and you open up the input to your big cone, you will have to have a heavy field magnet for proper attack and decay at full amplitude. That's when the cost starts mounting but that isn't the main factor in the cost of a really fine woofer of theater proportions. Unless many details are considered in proper shaping, strengthening, and suspending of the large cone, you will not get a clean attack and decay—even with the heavy field magnet.

These are problems you have to accept in producing adequate volume for a theater or auditorium. but why should the home owner have to be saddled with them? By the use of a small, light, but stiff cone, we have avoided all these difficulties to a great degree.

Therefore, we will start with a woofer which has a small hard cone—not over an 8-in. nominal size with a 6-in. effective working diameter. Phenolic impregnated paper works very well for the material of the cone. The angle its surface forms with the axis is smaller than in the normal speaker because we are not concerned with direction of sound projection. In other words, it is a steep cone.

This is contrary to all the literature on the subject, but we want it to work as a piston right up to the 1000-cps crossover, and only a small cone will do that. When a larger cone is used, it ceases to work as a piston at much lower frequencies, and trouble is likely to be encountered with spurious vibrations in the walls of the cone, producing subharmonics and the familiar buzzing on sustained low notes.

To drive the cone, we use a voice coil with a relatively large diameter for an 8-in. speaker—not less than 1¼ in. and preferably 1½ in. or larger. This places the thrust more nearly in line with the outer suspension ring and, with the steeper sides of the cone, prevents generation of subharmonics. The coil is round copper wire, as we are more concerned at low frequencies with low resistance to signal currents and proper heat dissipation than we are with mass.

Another important feature of our speaker is its heavy magnetic field. The first woofer we used consisted of the field pot from an old RCA 104 loudspeaker, with a new cone made of phenolic-impregnated paper. This was an electromagnetic speaker which required around 25 watts of field excitation.

Both the spider and the outer surrounding suspension ring must be freely compliant to avoid distortion from mechanical nonlinearity. However, the outer surround must be sufficiently tough to endure the rigorous punishment it receives, since the stiff cone does not absorb any of the flexing as an ordinary felted paper cone would. A ring of imitation leather similar to plastic materials used in upholstering makes an excellent surround.

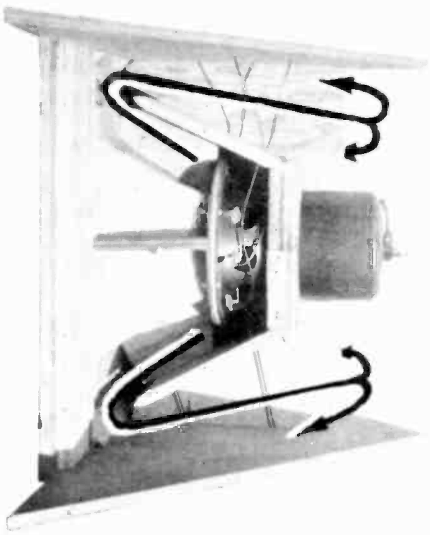


Fig. 2. View of the structure described by the author in early stage of construction. The back wave from the cone passes to the front of the cabinet, back around a partition to the rear, thence out through the side passages shown in Fig. 1, as indicated by the arrows.

Harnessing the New Creature

The loading of this driver is accomplished with two horns. The outer mounting ring of the speaker frame bolts to the sheet metal horn, pictured along with the speaker. This horn has a 4½-in. square throat and an exponential flare that doubles in area every 1¾ in. of axial length and ends in a mouth about 13 in. square. From the formulas for exponential flare and size of mouth opening, it is seen that this gives a low-frequency cutoff at around 450 cps.

Since the woofer is designed to work as a more-or-less rigid piston throughout its range, its response falls off rapidly for signals at much above 1000 cps, but the dividing network is designed for 1000-cps crossover. Thus the working range from this center horn is approximately 450 to 1000 cps.

The back of the woofer is loaded with a folded horn which depends on the walls and floor at the corner of a room for extension of its mouth and flare. Designers know from long experience that frequencies above 450 cps will not follow the tortuous path of a folded horn, so this horn handles the frequencies from 450 cps down to the lower extremity of its response.

Speaker expense is held to a minimum by the absolute simplicity and foolproof design of the woofer. It sounds too simple and easy to construct but it really works. And, for use in a small house, any good low-priced horn tweeter can be matched to it.

We are not concerned with direction of sound projection in a normal living room for it bounces off the walls at us from all directions. And, when the speaker system is placed in a room corner, there is no use in talking about 120-deg. spread of a multihorn tweeter, because the walls only allow 90-deg. spread.

Advantages of Horn Loading

The greatest advantage of a horn over all forms of boxes and resonators is the much smoother response curve with its freedom from sharp peaks and dips as different frequencies are reached. The only difficulty of reproduction or transmission by horns is in the natural nonlinearity of the air column when the greater amplitudes of the low frequencies are impressed upon it.

The nonlinearity of air coupling applies equally to any form of box or resonator. At the same time there is trouble in boxes and resonators from both the mechanical resonance points of speakers and the acoustical resonance points of enclosed air, as well as that of standing and reflected waves. Standing waves are set up by the meeting of equal waves reflected from opposite sides of an enclosure, such as a bass-reflex cabinet.

Standing waves are only produced when the wave length is shorter than the internal dimensions of the enclosure and when the sides of the enclosure are parallel. Thus, it is easy to produce standing waves in a living room with two bare walls facing each other and the speaker axis parallel and centrally spaced with these walls. Standing waves produce overemphasis or de-emphasis of certain harmonics so that a clear rounded voice or tone structure is not heard.

Reflected waves at high frequencies (the wavelength shorter than the internal dimensions of the box) cause distortion within any enclosure because they are out of phase with the direct-radiated frequencies. Thus the box must be lined with loose hair felt or other acoustical material which will effectively trap such unwanted high frequencies within the box. The larger the enclosure the more out of phase these paths become and the more important their trapping becomes. This is especially true in a labyrinth passage for a single wide-range speaker.

But we are not concerned with either standing or reflected waves in our system because we don't generate any short wavelengths within our large horns. The shortest wavelength that can be generated at the crossover region is slightly under one foot, and could cause none of these troubles in the small passages of the folded horn. It is the combination of low crossover frequency, horns, and corner position in the room that removes these problems.

Before you can appreciate the happy set of circumstances which makes this design so ideal for music reproduction the process of sound transmission through horns at different frequencies must be understood.

Sound waves are longitudinal in motion and tend to disperse in all directions in the form of a spherical wave. If originating between two buildings, for example, the wave readily bends or diffracts around corners and disperses in all directions. This is true mainly for the longer wavelengths, however, as they tend to cause actual compression and dilation of the air with enough energy to overcome the friction of changing direction.

The formula for surface area of a sphere is $4\pi r^2$ while that for the portion of a spherical wave subtended by the walls and floor around our speaker location is $\pi r^2/2$. Thus the sound generated in the corner is eight times as effective as it would be if generated outdoors on the top of a tower. By similar comparison it is apparent that the same sound source would be many times more effective if its wave propagation were confined to the almost parallel sides of a long horn.

The function of a horn, then, is to start the sound wave out as a minute sector of a spherical wave and gradually allow the sector to increase in surface area till it is finally turned loose in the atmosphere. The lower the note the stronger the walls have to be to contain it and the more readily it spreads and diffracts around the edge of the horn mouth when released. By using the corner of the room as an extension of the horn, the wave is never allowed to diffract around the edge and, consequently, the volume of those low notes is conserved for the listener to enjoy, in balance with the high notes.

It is not so simple as it sounds, though. Tests have proven that the greatest movement of the air particles takes place along the walls of a large horn when low notes are being transmitted. This is known as "annular movement" and is comparable to "skin effect" when high-frequency currents travel an electrical conductor. This effect diminishes as notes of higher frequency are transmitted. If we start low notes down the sides of a large horn and have higher notes taking a more direct course down the axis of the horn, it is easy to see that there is going to be considerable turmoil.

The solution to this problem is to hold the large horns to a relatively narrow range of transmitted frequencies. This is what we have done by having two horns to handle the frequencies below 1000 cps. The tweeter, for use at a higher crossover, can use a simple horn, with considerable saving in price.

But, not only do you have distortion between a wide range of frequencies in a large horn—you have distortion between single low notes and the outside atmosphere. Naturally, if you send a burst of annular ripples down the side walls and then allow them to suddenly diffract around the mouth of the horn, outside air is going to try to rush inward along the axis and cause both reflected waves toward the speaker and distorting eddy currents with the outcoming sound.

Natural Distortion from Instruments

Next time you are near some brass instruments, if you listen carefully you will notice that the lowest notes from the bass horns have a crackling sound. Like a series of sharp pops—similar to the crackling noise from exhaust pipes of large truck engines as they accelerate. To a lesser degree, you can distinguish the same crackling in vigorous low notes of the baritone horn and follow up through the alto horn, trombone, and low notes of the trumpet. You don't get completely away from the crackling effect till you listen to the smaller "toy" trumpet used in much of Handel's music.

Examine a mute for use in one of these instruments and you will see that it is designed to close out the outside air from the center but permit the annular passage of the fundamental notes along the walls of the horn. Thus it suppresses the upper harmonics of the fundamental notes and kills the eddy currents set up in the mouth by outside air.

These brass horns can not take advantage of the walls of the room to extend them, so they have exaggerated flares or "bells" at their mouths, to soften the sudden release of annular wave motion as it reaches the end of their restraining walls and to achieve better acoustical coupling. Thus musicians live every day with this harsh distortion of the brasses, and the expression "brassy" has an understood significance. The enthralling rustic vibrance of a "brass choir" would be lost without this effect.

All of these features of the horn work to advantage in this system. The fundamental ranges of the brass horns fall below 400 cps and therefore are in the range of the folded horn. The annular effect is what makes it possible for us to bundle the whole works up in a small cabinet and bring the bass notes out of narrow slits along the wall.

It is these very bass notes that will readily swing around the folded passages of our horn while the higher notes are snuffed out by loss of energy in acoustical resistance. Thus we have an acoustical filtering and less worry about inter-frequency distortion. At the same time, we have the effect of a large horn mouth without the distortion from eddy movements, because the center of it is taken up by the main body of our cabinet.

You may say that since we are essentially using the corner of the room beyond our cabinet as the mouth of our horn, we have theoretically reproduced the same features we said caused inter-frequency distortion in a large horn. For we bring the higher frequencies of our center horn out into the same mouth with the annular movement of bass notes. But there is little comparison here—as a plan view will show—for the angle of spread in a 90-deg. corner of a room is so great that their mixing is hardly of consequence. In fact, this is where the nonlinearity of the surrounding air works to our advantage. Remember that the low notes are trying to expand sidewise to form a sphere and have no tendency to move inward to mix with notes from the center horn, while the bulk of our cabinet in the center prevents the eddy movements in the same way the mute does in a trombone or other brass horn.

In the room with our speaker system one has the feeling of sitting in the orchestra with the instruments playing all around him. Within a few feet of it you are, in effect, sitting in the mouth of the horn. If one wants to hear a symphony with the feeling of being in the twentieth row of a concert hall, he can place the speaker in a room next to the living room and let the notes blend before coming through a door into the living room.

Part 2.

The first woofer had such a large field coil that the cabinet had to have considerable depth from the front, or face, back into the corner of the room. This meant that the face was wider also; as you can see, the base of an equilateral triangle increases with the altitude. The second assembly was built around a woofer with a permanent magnet and was scaled down somewhat in dimensions because the smaller magnet could be fitted into the corner more snugly.

The woofer may be built up by removing the frame from the permanent magnet structure of an old 12- or 15-in. speaker and replacing it with a sturdy 8-in. frame. However, a high-quality field-coil magnet can be used, if available, provided it has adequate wattage so as to provide a large magnetic flux—somewhere around 10,000 lines per sq. cm. It is necessary that the structure selected have a deep voice-coil slot, and that the pole pieces are long axially. The voice-coil diameter should be at least 1¼ in., and 1½ in. or more would be even better.

Suitable speaker frames—including the magnet structure—can usually be obtained for a relatively low cost from a shop which specializes in reconing speakers. While many of the modern high-flux-density speakers would provide an ideal magnet for this purpose, it would be futile to suggest disassembling one of them for the special type of woofer we desire. It was stated previously that such speakers were relatively expensive, and that this design was economical.

Having secured a suitable magnet structure, it might also be possible to obtain from the same source a frame

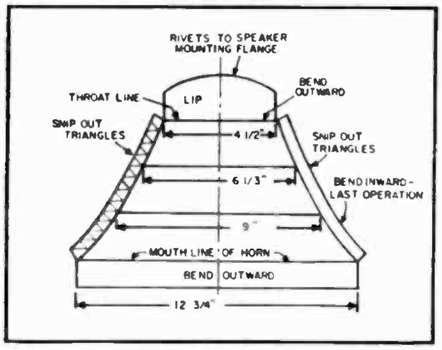


Fig. 3. Detail of one side of the metal horn covering the range from 450 to 1000 cps.

or "basket" for an 8-in. speaker. Remember that this basket should be as deep as possible—the one used on some of the earliest dynamic speakers is the type to obtain. Assuming that you succeed in obtaining a permanent magnet unit, be sure to cover the voice-coil opening immediately with Scotch tape—either the cellophane type or that known as masking tape—to preclude the slot being filled up with metal particles. Once these particles become lodged in the voice-coil slot, it is almost impossible to remove them—although one of the most effective ways to "try" to get them out is to fold a piece of this same tape over the end of a thin strip of metal, which can then be used to "wipe" the slot. A large percentage of the filings can be removed in this manner, although some are almost certain to remain.

After you have mounted the smaller frame on the magnet the rest is up to the speaker reconing expert. He must design a steep cone of stiff parchment, such as phenolic impregnated paper, and a voice coil to handle the wattage required for a small woofer. To match the average horn tweeter and utilize smaller capacitors in the dividing network, the voice-coil impedance should be around 16 ohms at 400 cps. This isn't critical, however; a value of 8 ohms is still usable.

The most critical item is the suspension of this cone. It will have a much greater excursion than any normal 8-in. speaker, so both the center spider and outer surrounding ring should be of some plasticized cloth fabric or soft leather rather than the usual molded mat of paper. These suspensions must be free in movement (not stiff) but able to stand the punishment of the greater flexing. The effective working diameter of the cone proper—its outside diameter—will probably be around 6 in.

The Intermediate Horn

Bolted to the mounting ring of the 8-in. woofer is the direct-radiating intermediate horn. Through this horn passes the major portion of the fundamental notes and the drum and cymbal crashes, etc. For the low notes of the outside horn, the shape of the walls is not so important but for this intermediate horn we must carry the flare out in a smooth exponential curve.

To meet all requirements, a throat opening of 4½ in. square works out best. Take a sheet of paper and draw an axis across the center. Then plot points on it starting with the 4½-in. throat, so that you will have a side elevation of the horn standing on its mouth, as in *Fig.* 3. Measure off three divisions of 1⅞ in. each along the axis and strike off a chord at each division, perpendicular to the axis. The area of the throat opening is 20.25 sq. in. By doubling this area and extracting the square root, it is found that the span along the side at the first chord will be 6-1/3 in. Plot half of this distance along the chord to each side of the axis. At the next chord the dimension will be 9 in. and at the mouth it will be 12¾ in.

Next, trace a smooth curve through the four points plotted on each side of the axis. This is best done with a "French Curve;" you may not find a curve that falls exactly on all four points but retrace the different sectors to get it as close as possible. Then use a flexible rule to measure the length of this curve to determine how long to cut the sheet metal side from throat to mouth.

To make these sides, get some 22 or 24 ga. furniture steel. 20 ga. is a little stiff for shaping but would give greater rigidity of the horn walls. On the steel, draw the axis again and then lay off the distance along the axis that was measured around the curve. Divide this distance into three equal divisions for the chords and lay off the former distances across the chords for the points of the curves.

The curves will be the same for all four sides of the horn, but two of the sides must have a ½-in. strip added along the edge as shown in *Fig.* 3. At the mouth of these two sides leave an extension of an additional 1 in. to bend over. At the throat of all four sides, leave an extended lip as shown. Cut out the two pairs of sides as drawn, and then snip out little wedge-shaped pieces from the ½-in. extended edge left along the curves of two of the sides.

The flange shown in *Fig.* 4 must be made from furniture steel of at least 20 ga. or heavier. Cut out a circular piece the same diameter as the outside edge of the 8 in. speaker frame. Scribe a circle on it which will just clear the inner edge of the speaker gasket where it bolts on to this flange. Then mark off the 4½ × 4½-in. square in the center of this circle and saw across the diagonals with a hacksaw, as shown. Drill

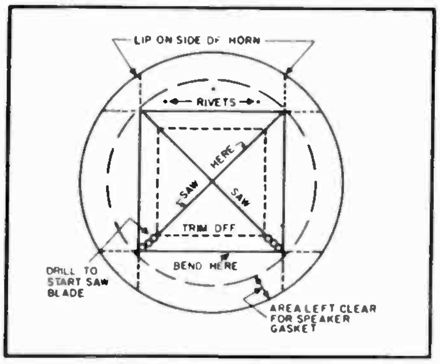


Fig. 4. Detail of the speaker mounting ring to be attached to the horn of Fig. 3.

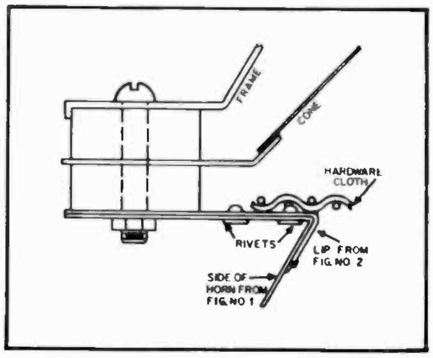


Fig. 5. Detail of the mounting of the speaker on the flange and horn.

two rows of ¼-in. holes at the corners for starting a saw blade in the cuts.

Clamp the circular edge in a vise so that a 4½-in. side of the square is flush with the edges of the vise jaws. If the vise does not have jaws wide enough, the width can be extended by two small pieces of angle iron. Bend the triangular piece, between two diagonal saw cuts, inward to a 90-deg. angle against the vise jaw. Then trim off the triangle to leave a ½-in. lip as shown. Bend over the next triangle, being careful to choose the side that will let the tin snips clear the first lip just cut. Continue around for all four triangular segments so that a 4½ × 4½-in. mouth opening with ½-in. lips results.

The lip extensions, at the throat line of the horn sides, should be cut to match the portion of the flange they mate with, around the mouth. These lip pieces may be bent over at approximately 90 deg. and the extensions on the bottoms of two sides may be bent over in the same direction, to an angle of approximately 45 deg. Then the two sides without serrated edges may be clamped against the flange, facing each other, with the ½-in. lip of the throat extending inside the horn and their upper lip pieces against the flange on the side away from the speaker gasket, as in *Fig.* 5.

While these pieces are clamped along the edges, drill small holes and rivet small tinner's rivets or nail stubs, so that they fall within the circle you drew to clear the speaker gasket. After riveting the two sides on, they can be carefully shaped over a large pipe or other cylindrical surface, for the horn flare curvature. Next, form the horn flare in the two remaining sides and then bend over the saw-teeth edges to the proper angle with pliers.

These saw-teeth are to reach along the outside of the other two sides and allow the soldering iron point to apply heat effectively and to flow the solder between mating edges. After you are sure every thing fits properly, these remaining two sides may be riveted to the flange at the speaker end. Some mechanics like to tin the mating surfaces before assembling. Use a little "cut" muriatic acid on the edges and solder with half-and-half solder and a large soldering iron. Solder an inch or two at a time while clamping the edges in snug position with "C" clamps or other suitable means.

After all corners are well soldered,

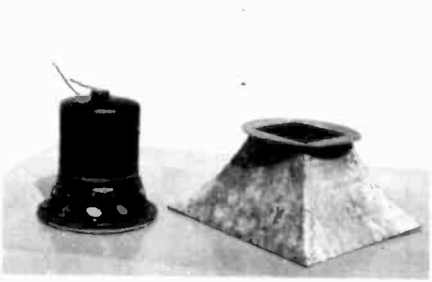


Fig. 6. Completed horn, shown with the early electrodynamic driver unit used in the first model.

place the throat opening over a flat extension of the vise or an anvil and use a hammer lightly to shape the inner ½-in. lip back against the inner surface of the horn for soldering. For riveting in tight places or for this shaping procedure, a stiff bar or angle iron can be clamped to extend out from a strong vise. This lip should be soldered to the horn wall to kill any chance for spurious vibration between the horn and flange sections. *Figure* 6 shows the finished appearance.

When you are through soldering there must be no visible air leaks up the corners of the horn or where the flange is attached. The speaker mounting ring must have a continuous gasket to mate with this horn flange, for completing this air-tight effect. Before the speaker is bolted to the horn, solder a piece of light hardware cloth across the throat of the horn flange, leaving no free edges to vibrate.

Design of the Main Case

The surest way to cut and assemble the sections of the wood cabinet without mistakes is to make a plan drawing of its assembly to scale, similar to *Fig.* 7 or 8. This is especially true if the plywood is of different thicknesses. In the cabinets we constructed we used ½-in. plywood throughout, but the front of the cabinet, sides, top, and bottom could be made of thicker plywood if it is on hand. All pieces must be at least ½-in. thick, however.

Figure 9 shows the metal horn attached to the front by wood screws but this was our first assembly. In later models, the fin-like separator pieces project at right angles from the center of each side of the sheet metal horn, to act as struts between partitions, and they are slotted across one side to let the 3/16-in. truss rods pass through them. These pieces must be shaped to fairly well match the surfaces they are to push against, and to obtain a snug fit so that partitions will not be sprung out of position as truss rods are tightened.

A truss rod passes from the side of the sheet metal horn to the outside of the cabinet in each of the four directions, at the center of the sound passages. After partitions for folded horn sound passages have been cut and shaped to fit together correctly, these fin-like separators must be lined up with the direction of sound and tacked in place with small brads. As the horn partitions are assembled, a 3/16-in. rod (such as steel welding rod) can be threaded through the group to help line up the separators while they are tacked in place.

Finally each truss rod may be cut to length and threaded on each end. Screw a nut on one end with a thread or so sticking through; clamp the rod in the vise with the nut resting on the jaws; and rivet the end with a ball peen hammer. This end is used at the outside of the cabinet with a large washer to keep the nut from seating into the wood with the vibrations. The other nut is screwed on the end which sticks into the sheet metal horn, with a good lock washer to keep it from working loose. Allow plenty of thread on this end for tightening.

In installing an 8-in. speaker frame on the larger magnet an additional 16-ga. metal flange can be included in the assembly, as shown in *Fig. 7*, to form a tie or hub for the partitions of the sound chamber. This member seals off the passage of sound to the rear and gives a more direct support for the heavy magnet. With our first woofer we used a wooden collar, fabricated by glueing two layers of ⅝-in. plywood together, as shown in *Fig.* 8. One layer was given ¼-in. slots to let a 3/16-in. tie rod pass through on each side of the speaker magnet.

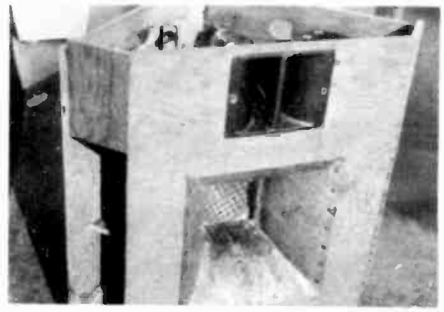


Fig. 9. Method of mounting horns to front of cabinet structure.

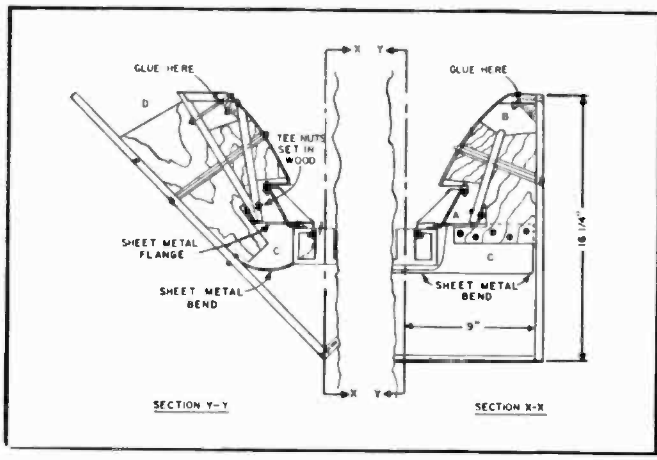
Where the metal horn fits through the front panel and where the partitions were fit to the wooden collar around the speaker, a thin layer of plastic (non-hardening) putty was applied between surfaces. This was to seal from passage of air and to prevent any spurious vibrations at joints.

No. 8 wood screws, of different lengths, were used in our first cabinet. Lengths were chosen so that 1 in. or more of thread was seated in the holding member and they were spaced 1½ in. apart in short runs and 2½ or 3 in. along the main sides. A small hole was drilled to guide each screw and prevent splitting of edges of panels. Then the hole through the held piece was enlarged for the shank and countersunk. A bit of floor wax was used to ease each screw home in its long run.

Arranging Inner Passages

In computing the sound passage area past the speaker mounting flange, the cross sectional area between partitions, at A in *Fig. 7*, is figured and the circular area of the speaker flange deducted. It will be seen that the sound waves have to turn a 90-deg. angle at the space marked A before departing from this chamber back of the mounting ring. Thus frequencies of the intermediate range are filtered out by acoustical resistance before they can cause much distortion in the folded horn passages.

Continuing along the inner sound



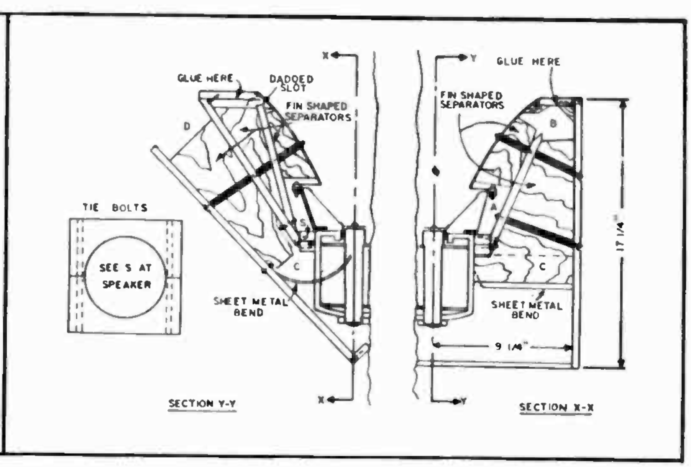


Fig. 7 (left). Horizontal and vertical sections through the center of the speaker unit for the second model, using a PM dynamic driver. Fig. 8 (right). Sections through first model, using the driver shown in Fig. 6.

passages, the cross sectional area at each change of direction is figured for doubling every 12 in. in axial length. Since there is no continuous 12-in. run in any part of it, this can only be a rough approximation, by proportions. Remember to allow for the same degree of expansion as the sound bends around the end of partitions because it may travel four or five inches axially in these bends. Leaving the sound chamber at A, the low notes pass toward the front of the cabine. and are bent back at B to pass rearward along the top and bottom panels to C.

Many corner speaker cabinets depend on the walls of the room to form the final folded passage, but this design uses complete side panels with the truss rods terminating in them. Pieces of ¼-in. hair felt were glued along the outside of these panels to prevent their making vibrating contact with walls of the room. Rubber feet were installed on the bottom. Thus, there is practically no sound transmission through the room walls at normal operating levels.

Curved inserts of sheet metal were bolted to the side walls and center separator to help the sound around the last bend at C and prevent possible distortion in this large chamber which surrounds the magnet. This sheet metal may be any gauge from 28 to 22, for easy shaping. Use lock washers on all bolts, which are 8-32 machine screws in most locations.

Before tightening the truss rod nuts in the metal horn, on the final assembly, twist a turn of small wire under each nut and allow the two ends to reach out the front like hairpins. Then take an old gauze or lace curtain and wad it up and pack it loosely in the horn to fill it almost to the mouth. Place a piece of light hardware cloth over this curtain and secure in place with these hair pins. If the hardware cloth is not light and pliable, its ends may receive vibrations from touching the metal horn.

The purpose of this curtain wad is twofold. Naturally, the direct radiated sound is much more effective than the lower notes which have to traverse the folded horn. So this material acts as a mute to cut down the intensity of the emerging notes and provide a distortion-free coupling with the outside air. Thus a fair balance can be obtained between the two horns and further experimenting may be resorted to in order to suit the needs of the particular speaker and horn assembly.

Don't be too concerned about the looks of the sheet metal horn or other features around that part of the cabinet because it will have to be covered with some cloth or other finishing. Picture molding was used to form open louvers over the horn on our first cabinet. There are many ways to dress up this cabinet to fit in with the surrounding environment. Do not run any wide molding up the edge of the horn opening against the wall at D, *Figs.* 7 and 8. Any molding which extends into this opening from the wall would curtail the annular movement of the lowest notes along this outer wall.

Like most experimenters, we have never gotten around to finishing our cabinet because we are too busy on other projects. Louvers should not be used over the tweeter and any cloth used over its opening should be of fairly loose weave and hard, wiry thread, preferably plastic.

Both the tweeter and the center horn must be mounted flush with the front surface of the panel (disregarding louvers which extend beyond). The reason for this is, of course, to avoid any discontinuities in the sound path where the horn joins the open air.

Sounds like a lot of work, doesn't it? It has taken years of struggling to achieve this ideal and the real listening satisfaction for a true music fanatic. Let's put it this way: A sextet of saxophones of different ranges can imitate the brass choir, but the comparison in texture is about the same as comparing an ordinary 12- or 15-in. resonator-boxed speaker with our system—except that the saxophones are a little muddier than the box speaker.

Part 3.

As previously stated, the tweeter may be any good make of horn tweeter, with a horn that cuts off well below 1000 cps. Some manufacturers have versions that are inexpensive ($17 to $35) and suitable for our purpose. As stated earlier, we are not so concerned with the angle of spread of sound from the tweeter, where it is to be used in the corner of a living room. The rated impedance is most desirable at 12 or 15 ohms but it may be 8 ohms. In any case, it is simpler if it has the same impedance as the woofer used.

Within the woofer's operating range, its voice coil impedance does not range much above its d.c. resistance value, while the impedance of the tweeter varies over a larger range, in relation to the frequencies it is generating. So we can design the components of the dividing network for the impedance of the woofer and forget about the tweeter, as long as it is not over 100 per cent from that of the woofer, and we can match impedances by using add tional resistors, if desired.

Many good articles in the literature have covered the details of designing and constructing dividing networks. These articles give charts for estimating the number of turns of wire to wind on each coil and the dimensions of the coil spools, etc. So we will not take up space with such details here.

For winding coils, remnants of No. 16, 17, or 18 magnet wire can be obtained from motor winding shops and can be spliced together with a thin copper sleeve, sweated over butted ends. Thin "spaghetti" can be slid over the sleeve, after soldering, for insulation. The voltage is so low that insulation is not a critical item in these coils.

Surplus capacitors can generally be used by paralleling different sizes to get the right capacitance. These can be the oil-filled paper type with working voltages at 25 or 50 volts, if obtainable. If the higher voltage types have to be used they will be much larger in size and higher in cost. There are some electrolytics for sale in "bathtub" cases, so beware of them. The regular oil-filled paper types have no polarity marking of terminals but the electrolytics either have a positive and negative mark or the terminals have different colored insulators. And, of course, the electrolytic is much smaller.

We used the series-filter system for our dividing network, since authorities seem to agree that it is the most desirable. It requires coils and capacitors of different sizes for the two legs or circuits. The coils may be bolted to the sides of the speaker cabinet—in the upper compartment allotted to the tweeter. Place them several inches apart on adjoining sides, so that their axes are at right angles, and use brass or aluminum bolts.

Preliminary Listening Tests

Connect the two coils and two capacitors together in proper relation with joints soldered in a temporary manner for testing. Then place the tweeter about 8 or 10 feet away from the main cabinet for the first hour or two of listening. By this means you can tell what you are getting from the different components. It is much easier to judge, if you have an audio oscillator, but a frequency test record can be played on your record player to check the response at crossover between the two speakers. Using an output meter across the speaker terminals or a microphone placed in front of the speakers is of little avail because of the vast difference in the character of the two speakers and differing spread of sound dispersion.

If this is your first experience of trying out two speakers and a dividing network, you may be highly confused at first. Things do not happen like you expected and you wonder if you are a complete flop. But just be patient and make no decisions until you have grown accustomed to listening to them for several hours. If you are getting response from each speaker you know that there is nothing open-circuited and it is going to take time to appraise the balance.

If this is your first experience of listening critically to two horn-loaded speakers, you not only have the surprises of where the sound is going to hit you from but your ear will have to get used to richer notes packed with harmonics you were missing before. So, even the playing of your most familiar records can be disconcerting. Thus we repeat: your first decisions are going to waver back and forth and you had better not make any.

In the end, the thing you will have to decide (maybe with the help of others) is whether you have a satisfactory balance in volume levels between the two drivers. We mentioned, in an earlier issue, that a University 4408 tweeter so closely balances the output of our horn-loaded woofer that they are just connected directly to each leg of the dividing network. If one of your drivers overbalances the other it can be fed from a potentiometer connected across its leg of the network, as described in articles on dividing networks. This may be a 10-watt wirewound resistor with a sliding clamp contact for takeoff adjustment. Its resistance may be from 100 to 200 per cent of the voice coil impedance rating of that driver in ohms. It's a good idea to measure the d.c. resistance of voice coil and resistor with an ohmmeter, as an approximate check.

Means of Comparing Speakers

You are going to find that it is no easy matter to decide whether you should be satisfied with the balance or not. You play an album of records and then you change it one way. Later on you play some other records and decide it still isn't right. But remember that in recording you have differing characteristics in microphones and their placing, not to mention difference in recording equipment and operators.

Since your main concern is with the region that the crossover falls in, it is a big help to compare your balance with the output from any single cone speaker of fair quality. This is just one of the reasons you are going to need the control hookup we use. Another good use for this hookup is to prove to the family and friends that you aren't going slowly off your nut, when you try to explain the purpose of this new system and what is different about it.

When a real music lover comes to visit you and listens to a good over-all system for a while, he will soon make comments or ask questions about the means of obtaining this realism. As the reader doubtless knows by now, it is fruitless to start an explanation of speaker systems. But it is a simple matter to demonstrate all aspects by flipping the switches in our control hookup.

All that is needed to complete the hookup of these controls and the dividing network (as shown in *Fig.* 10) are four DPDT toggle switches, a phone plug jack, and an L or T pad. The pad may be 12 or 16 ohms, assuming the impedance of the drivers is in that region. All of the equipment shown in *Fig.* 10 is installed in the upper compartment with the tweeter and we have switch handles protruding into one of the side openings of the bass horn, next to the wall, where curious hands are not likely to discover them. The jack also opens into this passage as does the handle of the T pad.

Switches 1 and 4 are used most so they were placed on the ends of the line of switches, so their handles are easily found by feel. Switch 2 is thrown to the position which inserts the pad only when another speaker is to be plugged into the jack. Most of the time the pad is out of the circuit. But the pad must be used to attenuate the response through the dividing network, when comparing our horn-loaded speakers with ordinary box-mounted cones.

To check the balance between woofer and tweeter (after the connections of *Fig.* 10 have been made and an attenuating potentiometer for adjusting the tweeter-woofer balance has been inserted) a normal cone speaker, around 10 or 12 inch diameter, is plugged in. Then adjust the pad while operating switch 1 back and forth until the volume is the same in either position. Now play several orchestra or choral records through the new system and at points where you doubt your judgment switch to the other speaker and see if there is any great difference.

We want to emphasize that the balancing is to be done on the passages around the crossover region. Do not try to judge the system on solo flute or violin music, for example. The comparison speaker will be of little use two octaves away from the crossover.

Confidence in Appraisal

Listening at all the different horn mouths of the new speaker system will show you that all kinds of thin squeeky noises come from the tweeter and choked squawks come from the lower horn in some alternate passages. These things happen when the dividing network cuts the broad band of harmonics of a note (in the crossover range) into shreds and you were listening to the narrowest of the shreds. Several feet back, this is not apparent, but if you really want to hear it you can flip either switch 3 or switch 4 and sample it readily. Then, if you think it shouldn't sound that way, plug in the comparison speaker. Then when you flip the switch, the sound will come out of the comparison speaker and you can see how it sounds there. If you have to use the T pad to attenuate the tweeter or woofer response for comparison, it will be necessary to flip switch 2 back and forth with the other switch. By this means you insert the T pad with its attenuation, for the horn-loaded speaker, and drop it out for a straight through run to the comparison speaker, with its poorer efficiency.

This arrangement provides a versatile sampling hookup. The quality or response of over-all systems can be compared or either treble or bass response can be sorted out for comparison with that of another speaker. If you are trying to explain to someone that a cello or Ezio Pinza excite a broad band of harmonics, you can cut out either driver and show that there is still a considerable complement of harmonics issuing from the other. On the other hand, you can show that a lyric soprano has no lower harmonics while a mezzo-soprano has to have woofer response for rounded notes. It is uncanny what can be done with it.

Les Paul started the novelty of recording several sound tracks on one tape with different modes of playing the same steel guitar. With his record of "Little Rock Getaway" you can completely erase the highest or the lowest sound tracks with switches 3 or 4, because there are no stray harmonics of these tracks on the other side of the crossover. We can take certain records of Lily Pons and erase Pons and leave the orchestra still playing along fairly effectively.

When we are trying to point out to untrained ears how the efficient response of the little woofer will etch the sharp intensity into cello or English horn passages, their attention is likely to be distracted by the sharp excursions of some flute or violin. So we can cut off the tweeter and focus their attention on the feature we are trying to demonstrate.

Then there is the fellow who thinks his pet speaker is just as good as yours and he has infinite faith in his hearing and memory. Tell him to bring his own speaker and his pet records over and you can really give him a comparison. When he hears his own records he cannot claim any fakery in the line up, if his speaker is shown up.

Hold That Distortion

All of the foregoing discussions of the virtues of this speaker system are predicated on the assumption that high-quality music signal is being fed into the dividing network. There is the old argument between techincal experts—whether one should improve the speaker system or the other sound equipment first. We believe that the speaker is the end to start on because you can do a lot of see-sawing around in the feed-

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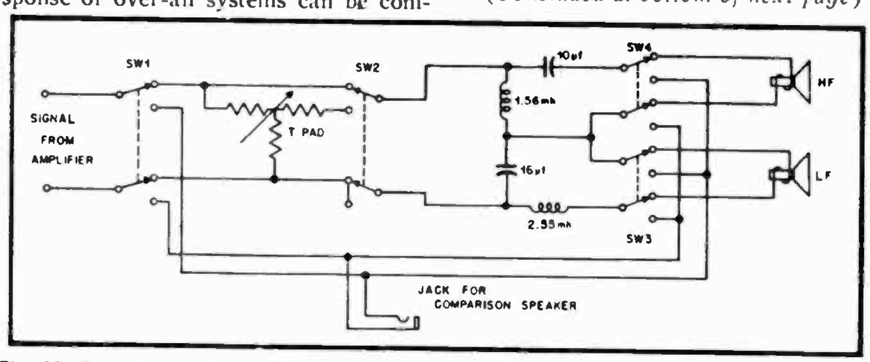

Fig. 10. Schematic of the dividing network to show the four switches used in making comparisons between the over-all system and other speakers, and for checking performance of the individual components.

How to Brace a Speaker Cabinet—

Vibration Reduction in Loudspeaker Enclosures

G. B. HOUCK*

Presenting the reasons for bracing a bass-reflex loudspeaker cabinet, and showing how to do it with the assurance of improved performance when the job is completed.

Benjamin Franklin once observed that "Empty barrels make the loudest noise." In this case he was referring obliquely to the common phenomenon of uninformed vociferation, not describing the performance of a loudspeaker enclosure. G. A. Briggs, in the second chapter of "Sound Reproduction," was commenting on the latter when he wrote: "The indications are that the effect of cabinet resonance has been underestimated in the past." He observed that the tone-quality of reproduced sound was greatly improved when the loudspeaker cabinet was constructed of materials having a high density. In a paper presented before the IRE PGA[1], Frank McIntosh pointed out that "boomy" sounds are caused by acoustic radiation due to decaying vibration of the panels in a poorly braced cabinet.

Briggs offers one solution to this problem—make the panels massive and they won't vibrate. The principle involved is that of relative momenta. Consider the effect of a moving mass of air striking a panel. Referring to *Fig.* 1, if a unit volume of air having a mass M_1, and an instantaneous maximum velocity V_1 strikes a unit volume of panel having

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[1] IRE Professional Group on Audio, No. 10. September, 1952.

TABLE 1

Physical Properties of Common Construction Materials

Material	Mass lbs./ft.³	Thick. In.	M_2 lbs./ft.²	M_2/M_1	V_1/V_2
Dry packed sand	105	1	8.75	114	10.7
Brick	125	3	31.2	406	20.2
Concrete	150	3	37.5	488	22.1
Plaster	—	1	8.0	104	10.2
White Pine	26	⅞	1.9	24.7	4.9
White Oak	46	⅞	3.36	43.8	6.6
2-⅞ wood panels with 1" sand between			12.55	163.5	12.8

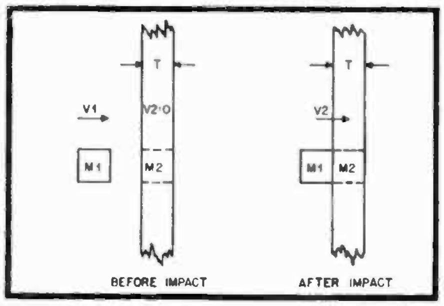

Fig. 1. Kinetic energy effect of Unit Volume of air M_1, impinging on Unit Area of panel, M_2.

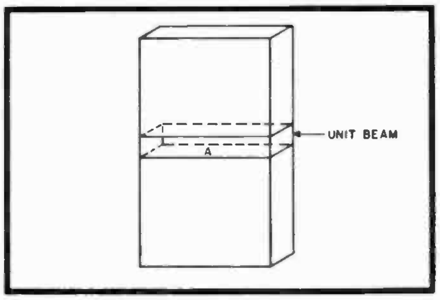

Fig. 2. Typical panel as used in the discussion.

THEATRE SOUND

(from preceding page)

ing equipment without being fully conscious of results unless you have a sensitive speaker system.

It is about like the old tale of the man who found a horseshoe and bought a horse to put it on. Our new speaker system will mercilessly display the defects in records and equipment. The first thing you have to do to soothe it is get rid of record surface noise with a magnetic pickup and a new changer—one of the better modern changers, with only a few grams needle force and a ball-bearing arm swivel, so it is free from the different forms of distortion contributed by drag of arm on groove walls.

If you have not done this already, you have no idea how different the records can sound. Fifty per cent of the surface noise and distortion from inexpensive pickups is from vertical vibrations and is mostly eliminated by change to a modern magnetic or variable reluctance pickup. The reason being that they produce practically no response from vertical vibrations. The other fifty per cent still remains from lateral distortion of stylus movement caused by a heavy arm with stiff swivel joints, working against the whip of eccentric or warped records.

After the man got a horse he had to get a stable and a riding habit. So a preamp stage ahead of the amplifier will be necessary, unless it already has one. Then we start in on equalization—because the speaker system shows that up. There are almost as many ideas for attacking this problem there are ideas for speaker enclosures but we have not experimented with these to the same extent. Success in the matter is not difficult, however.

The amplifier is the oldest member of the team, having started its career before the loudspeaker. So there are no secrets about amplifiers and plenty of satisfactory performers can be had at moderate prices. But, here again, modern tubes, circuits, and output transformers will probably make a better showing in reproduction of the exquisite detail in later recordings, as displayed by the new speaker system.

In the writer's opinion, simplicity is the watchword—from the diamond stylus right through all stages to the dividing network. Simplest tubes with self-biasing, no interstage transformers, simplest power supply with oversize components, and no special circuits in the regular line up.

It is hoped that this description may have been sufficiently full of detail that any interested hobbyist who wants speaker performance which, in the writer's opinion, is well above the average, will be able to duplicate the enclosure type with a minimum of effort. The pleasure given by a speaker of this type is usually well worth the trouble taken to build one—a fact which is attested by a number of the writer's acquaintances who are using the same type of speaker housing.

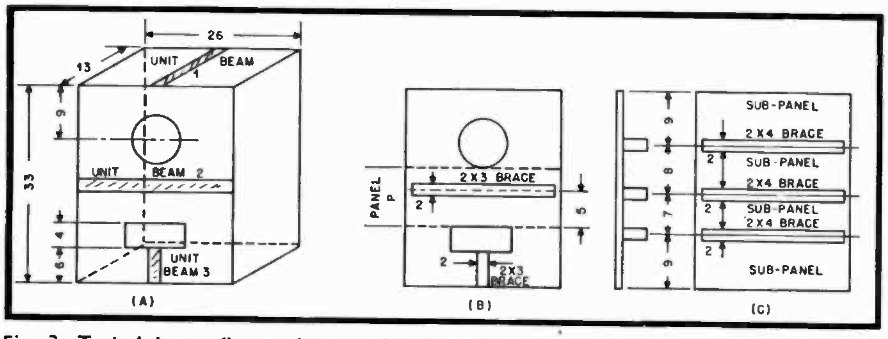

Fig. 3. Typical bass-reflex enclosure, with dimensions as used in the sample problem. All stiffening beams are mounted so that their maximum cross-section dimension is perpendicular to the plane of the panel.

a mass M_2, initially at rest, both masses will have a resulting velocity V_2. This relationship may be written:

$$M_1V_1^2 - M_2O^2 = (M_1 + M_2)V_2^2 \quad (1)$$

Note that for the optimum condition, V_2 approaching zero (panel does not move) it is necessary to have the ratio of M_2/M_1 as large as possible. Since M_1, the mass of air is fixed at roughly 1/13 lb./cu. ft., M_2 remains the only variable. By varying M_2, it is possible to obtain the ratios shown in Table I for several different materials. It appears obvious from a glance at this table that a small improvement in V_2 (hence a reduction in vibration) may be had only at the expense of a large increase in weight. For example, a panel made of concrete would tend to vibrate (other things such at the modulus of elasticity being equal) 1/4.5 times as much as one made of wood, but would weigh 20 times as much. Thus a typical bass reflex cabinet weighing 50 pounds constructed of wood, would weigh 1000 made of concrete—probably too much for the average living room floor to support. Even if this were permissible, such an enclosure would be virtually immovable and would present the baffling (no pun intended) problem mentioned in a recent editorial.[2]

Fortunately for the cabinet designer there is another solution to the problem of reducing panel vibration. Instead of relying on weight alone to accomplish the desired results, he can make the panels stiff, and join them rigidly together.

Stiffness of Panels

Several factors determine the stiffness of a panel. The chief factor of course is the geometry of the panel. In most cases it is very difficult to analyze the behavior of a vibrating plate, especially if one attempts to relate various design parameters to a resulting acoustic output. It is entirely beyond the scope of this discussion to examine these theoretical considerations in minute detail. Furthermore, it can be shown that a much simpler method of analysis provides the essential information necessary to make very substantial improvements in cabinet construction.

For all practical purposes it is reasonably sufficient to consider a panel as made up of an infinite number of small beams arranged side by side as shown in *Fig. 2.* Notice that in this type of analysis, the beams are represented as extending across the shorter dimension of the panel. Assuming the edges of the panel are supported, it is logical to suppose that the beam exhibiting the most severe deflection when subject to a load, will be one near the center of the panel such as beam A. Now, without attempting to determine an exact coefficient for the stiffness factor, it can be shown that the maximum deflection of beam A is dependent on a few easily determined variables. Actually, since the beams are integral parts of a homogeneous plate, the deflection will be somewhat less than that of a single unattached beam.

The equation for the deflection of a rigidly supported, uniformly loaded beam may be written as

$$Z = \frac{W L^3}{384 E I} \quad (2)$$

in which Z equals the maximum deflection, W equals the load on the beam (maximum instantaneous value), and I equals the moment of inertia of the cross-section of the beam. If this equation is compared with that for a non-rigidly supported beam, in which case

$$Z = \frac{5 W L^3}{384 E I} \quad (3)$$

it will be observed that the deflection is five times the magnitude of the former. Of course in actual practice these extremes are almost never encountered; no panel however loosely secured would exhibit a vibration five times as severe as one firmly attached to an immovable support. Nevertheless, this simple comparison emphasizes the need for rigid support of the panels.

The maximum deflection of such a beam may also be reduced by decreasing its length as far as is practicable. On the other hand, little benefit is derived from attempting to vary the value of E: reference to appropriate tables reveals that the modulus of elasticity of commonly used lumber varies from about 1.00×10^6 to 1.6×10^6.

The moment of inertia of these beams equals $bd^3/12$, where b is the width of the beam and d is its thickness. (See Table II) It is interesting to note here that doubling the thickness of a panel (and hence the thickness of a unit-width beam) reduces the deflection by a factor of eight; tripling the thickness reduces deflection by a factor of 27; and so on. Before demonstrating a typical solution to a cabinet design problem, it will be found helpful to introduce one further beam equation into the discussion. In the case where a panel contains an opening (speaker mounting hole or reflex port), the beams which have one termination at an opening are classified as cantilevers. The deflection for this type of beam is written as:

$$Z = \frac{W L^3}{8 E I} \quad (4)$$

For convenience, the terms in the equations which remain more-or-less constant in any one design, are W and E, and they may be lumped into one constant, K. The equators are then rewritten as:

Simple beam—no end supports

$$Z = \frac{5 K L^3}{384 I} \quad (5)$$

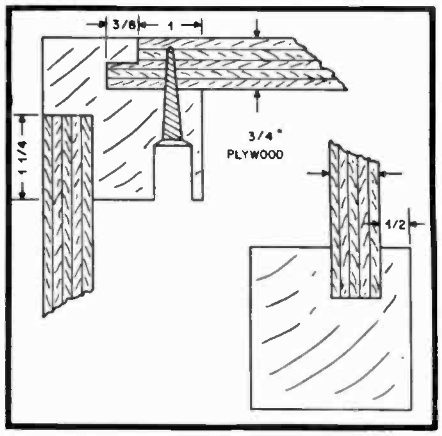

Fig. 4. Examples of cornerpost construction.

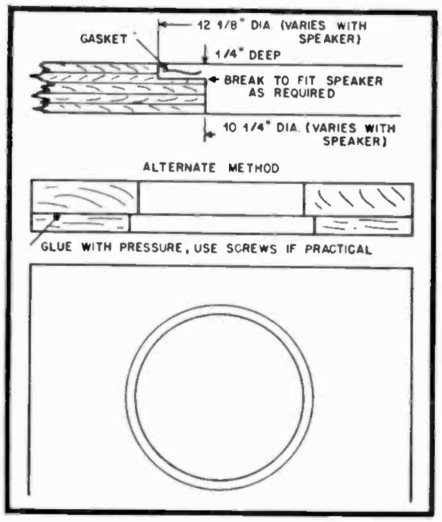

Fig. 5. Loudspeaker mounting details.

[2] Editor's Report, Audio Engineering, March, 1953.

TABLE 2

Moment of Inertia about the Center of Gravity for Common Lumber Sizes (max. Value)

Nominal Size In.	Dressed or Finished Size In.	Moment of Inertia In.[4]
1 × 1	7/8 × 7/8	.049
1 × 2	7/8 × 1 5/8	.31
1 × 3	7/8 × 2 5/8	1.32
2 × 3	1 5/8 × 2 5/8	2.45
1 × 4	7/8 × 3 5/8	3.44
2 × 4	1 5/8 × 3 5/8	6.45

Simple beam—fixed end supports

$$Z = \frac{K\ L^3}{384\ I} \qquad (6)$$

Cantilever

$$Z = \frac{K\ L^3}{8\ I} \qquad (7)$$

Figure 3 and the sample problem show the suggested method of analysis as applied to a typical cabinet. Cabinets of different shapes will present stiffness problems somewhat different from the sample illustrated. In this case, the analysis is used to determine the size, shape, and number of braces required to improve the performance of a cabinet originally constructed without full consideration of the factors just discussed. Naturally the need for extensive bracing is reduced if the enclosure is properly designed from the start.

Construction

There is more to the problem of designing a rigid speaker cabinet than mere choice of adequate panel thickness and arrangement of bracing, however. As was observed from the discussion of the beam equations, above, the method of supporting the panels is highly important. Since most panels are made of plywood, it is necessary to insure that all joints are constructed to achieve the maximum possible rigidity. A simple but effective method for accomplishing this is to use corner posts to which the panels are securely anchored. A few typical joints of this type are illustrated in *Fig.* 4. It will be found that such methods also contribute to the over-all appearance of the finished structure.

At first glance, the prospective builder may be somewhat dismayed by the suggestion that special tools are needed to prepare the joints for assembly. This type of work is best done using a joiner or router, although the patient craftsman may use a plane with very good results. In many instances the whole problem can be greatly simplified by having a local mill cut each piece to size and prepare the joining surfaces from drawings furnished by the designer. The task, then, is merely one of assembling the finished parts.

During this assembly operation it is best to screw and glue all joints together so that the finished cabinet will be tight and solid. The front panel of the enclosure can also be permanently anchored if the loudspeaker is mounted as shown in *Fig.* 5. It is fairly well recognized that speaker performance is greatly improved by mounting the speaker in this manner. As illustrated, an alternate method using two superimposed panels obviates the necessity for special routing. In either case captive nuts attached to the back of the panel receive the speaker mounting bolts from the front. Many loudspeaker manufacturers will, upon request, supply suitable gaskets for this type of mounting.

Conclusion

It has been the purpose of this discussion to analyze the problem of spurious acoustic output caused by excessive panel vibrations in a loudspeaker enclosure, and to suggest an approach which will result in a definite reduction in the effect and a vast improvement in the over-all performance. As might be expected, the vibrations dealt with are those which occur at low frequencies. The various equations do not take resonance into account but explain the phenomenon below resonance where dynamic and static deflections are nearly the same. If one remembers that the natural resonant frequency of a solid varies inversely with its weight and directly with its stiffness, it is obvious to conclude that constructing panels as rigidly as possible helps to raise the resonant frequency to a region where damping is more easily accomplished by the use of padding and sound absorbing materials. This is easily demonstrated with the test set up illustrated in *Fig.* 6. Connected in this manner, the oscilloscope indicates the power factor of the load. At resonance, the power factor is unity and a straight line appears on the scope. Power factors at non-resonant frequencies show up as loops of various widths. If the oscillator is adjusted until a panel resonance is detected, it may be observed that the application of hand pressure to the vibrating panel will cause the straight line to open up into a loop, and by returning the oscillator it will be found that the resonant frequency occurs at a higher frequency. During this test, of course, one must not pick an oscillator frequency which corresponds with the natural acoustical resonance of the cabinet. In this case applying pressure to the panels will have little effect.

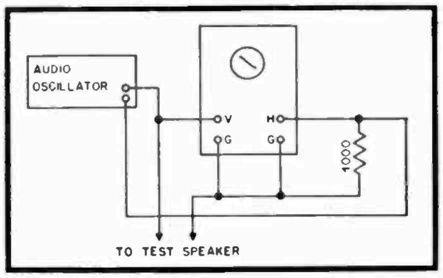

Fig. 6. Test circuit used for checking impedance and resonant frequency of speaker in enclosure.

SAMPLE PROBLEM

Consider the bass reflex cabinet shown at (A) in *Figure* 3. Assume that it is constructed of 1-in. plywood (7/8-in. dressed), and that all corners are joined in such a manner that the panels are rigidly supported.

The top panel will exhibit the smallest maximum deflections because it is the smallest panel. The deflection of unit beam 1 is given by equation (6).

$$Z_{max} = \frac{K\ L^3}{384\ I}$$

In this case *I*, the moment of inertia, from Table II is .049 and

$$Z_{max} = \frac{K\ 13^3}{384 \times .049} = 117\ K$$

Since unit beams chosen from the bottom or sides would have identical dimensions, the maximum deflection of these panels would be the same as the top panel.

The maximum deflection of the front panel is determined by considering unit beam 2. In this case

$$Z_{max} = \frac{K\ 26^3}{384 \times .049} = 936\ K$$

The deflection of the back panel is similar.

Because of its shape, the area of the front panel below the port (6 × 26 in.) exhibits a deflection which is characteristic of unit beam 3, which is a cantilever. The deflection is given by

$$Z_{max} = \frac{K\ L^3}{8\ I} = \frac{K\ 6^3}{8 \times .049} = 551\ K$$

In determining the bracing required to reduce front and back panel vibration to a value consistent with the top panel, consider the load as the area defined by panel "P" in (B). Its width is roughly 10 inches. Since a unit beam has a width of 1 inch, and the deflection is 936/117 or roughly 8 times as severe as for unit beam 1, the required stiffness of the brace beam is 8 × 10 or 80 times that of unit beam 2. From Table II it may be seen that the best brace would be a 1 × 4. The stiffness factor equals 3.49/.049, or 71, which is close enough to the desired value. Actually it may be more desirable to use a 2 × 3 as shown at (B) with a stiffness factor of 245/.049, or 50, in some instances where a brace 1 × 4 would adversely affect the acoustics of the cabinet. Since such effects are generally not serious near the back of the cabinet, it is recommended that 2 × 4's be used. Here the stiffness factor is 6.45/.049, or 132. If three braces are used as shown at (C), each will support approximately one fourth of the total load. This makes the stiffness factor equal to $8 \times \frac{33K}{4}$ or 66 K. If the above calculations are applied successively to the braced panel shown at (C), two deflections are obtained.

Since both sub-panels and braces vibrate in unison, the final deflection is obtained by adding the two and the sum will be found to equal 117 K.

Because of the short length of unit beam 3, sub-panel Q (A of *Fig.* 3) requires less bracing than sub-panel P. Just to be on the safe side however, a 2 × 3 should be installed as shown.

The reader will note that all the braces used add up to a total volume of ½ cu. ft. which is about 1/12 of the cabinet volume.

Feedback and Loudspeaker Damping

JOHN A. MULVEY*

The author proposes a solution to the problem of obtaining a feedback signal which is equivalent to the movement-generated e.m.f. of a speaker voice coil.

The relatively recent articles on positive current feedback used to improve loudspeaker damping have, to this reader, been most engrossing. In spite of studies made of the articles and arguments on this subject he has, until quite recently, felt much in the dark when it came to feeling able to claim much of an opinion of his own on the relative merits of such a system. It seemed that such a subject should be resolved into simpler contradictions than any presented so far in order to make one feel he has made a wise choice of one opinion or another.

This need has led to this attempt to reduce the subject into simpler terms by digging into the fundamental aspects of the subject. The author feels that the analysis which is here presented can show some things not apparent in foregoing discussions, and finally may offer amends to those who think they disagree, as well as to make some proposals that the engineers more actively engaged may find promising.

According to Lenz's law, the voltage induced in a coil due to its passage through a magnetic field is always of a polarity such as to produce a current which would oppose the motion. In other words, an e.m.f. due to coil motion in a field tends to stop the motion. This presumes, however, a closed circuit at the coil ends, for this e.m.f. can only act if it can act to cause or influence current through the coil. When such a coil is a voice-coil there is always such a closed circuit existing. The path is around through the windings of the output transformer secondary. This closed circuit can nearly constitute a short-circuit for the coil, permitting much current to flow with just a small movement-generated coil e.m.f. The more current which flows as a result of this e.m.f. the better will be the dynamic braking of the movement, that is, the quicker will the movement cease. But even with a short-circuited coil moving in a magnetic field the cuurrent through it is definitely limited. It is limited by the resistance of the coil itself. A method to overcome this limitation and further enhance the dynamic braking action is most desirable. Such a method, to be successful, would have to permit a greater current flow than a short-circuited arrangement would allow. At first any such method might seem impossible for it would seem that nothing could allow more current than a short circuit. But on second thought it should be realized that a second source of voltage in series with the voice-coil could be provided to increase the current above what simply a short-circuited voice-coil would allow. This suggests a circuit employing as a second source of voltage one greater but proportional to the movement-generated e.m.f. and in phase with it having also a lower source impedance, as in *Fig.* 1. So far we have spoken only of movement-generated voltages. In the case of a loudspeaker, where it is voice-coil current which causes the motion in the first place, movement-generated e.m.f. is still present and acts to oppose the voice-coil current. In this way it influences the actual current and always acts to stop the motion the same as if some physical force was causing the motion. However, generally under these circumstances the movement-generated e.m.f. itself causes no current to flow. It only acts to influence the voice-coil current which is moving the voice-coil, by subtracting from the voltage causing it. This is true all the time except during "hangover" periods. During these periods the physical forces of inertia or suspension compliance cause the movement-generated e.m.f. to exceed the driving voltage and so cause its own reverse current to flow. Notice here that this is a "reverse current," one of opposite polarity to that originally causing the motion. This reverse current then will act on its own and damp the motion and so limit the hangover effect. Since positive feedback—whether voltage feedback or current feedback—tends to increase the output signal, at this time if positive current feedback were introduced it would increase the current limiting the hangover effect so as to squelch it quicker.

From this the main point may be apparent but it is simply that positive current feedback does not oppose voice-coil movement until the movement-generated e.m.f. exceeds the driving voltage and produces a reverse current. At this time there is a phase reversal which, in effect, changes positive feedback to negative feedback. If this point is not made clear there is yet much room for controversy. It should be noted that the comparison was said to be only an *effect* similar to inverse feedback. The e.m.f. generated by a coil moving in a magnetic field *is* inverse feedback. Any voltage which aids that voltage is, with regards to it, a positive feedback voltage, but with regards to the total effect it is a negative feedback voltage. There is a good comparison in the rather familiar circuit of an amplifier using a positive feedback loop connected to the cathode end of an unbypassed cathode resistor. The feedback voltage is definitely positive with respect to the signal voltage appearing across the cathode resistor, but just as definitely negative with respect to the over-all effect.

But the story doesn't end here. To some, the advantage of improved damping during hangover periods will be thought to be too highly paid for by the inherent disadvantages existing the rest of the time, since the rest of the time the in-phase feedback is regenerative.

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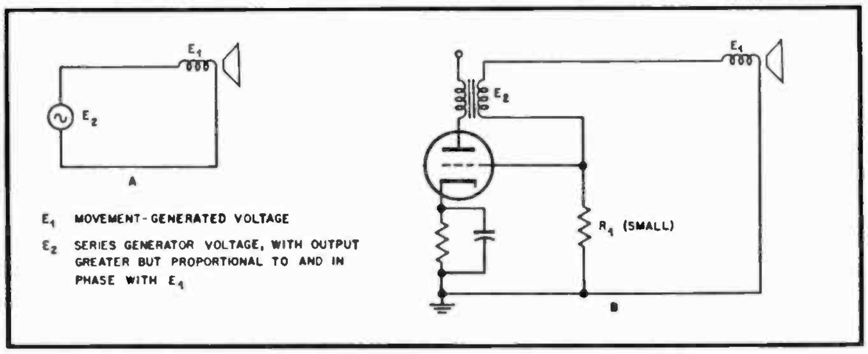

Fig. 1. (A) Theoretical circuit, and (B) practical circuit of feedback obtained from current in a voice-coil.

Damping of Loudspeaker Cabinet Panels

M. RETTINGER*

A discussion of the need for damping of enclosure panels and suggestions for realizing the required result. Considerable improvement in speaker performance can be obtained with relatively little expenditure of money or energy.

THE LOUDSPEAKER IN A CABINET often produces undesirable vibratory motion of the cabinet panels. It is well known that any surface excited into flexural vibration has families of resonant frequencies, so that the radiated spectrum is not the same as that produced by the primary signal forces. How to avoid the effect which reinforces certain frequencies and thus provides selective radiation is an old problem, and the usual recommendation for cure is to brace the enclosure walls. However, increasing the stiffness merely raises the resonant frequencies. True, the amplitudes of the higher resonant frequencies can be more easily damped; yet the total radiation may be more pronounced on account of the larger number of vibrating surfaces.

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Thus stiffening tends more to change the character of the radiation than to eliminate it.

Another undesirable effect of panel vibration is poor transient response of the radiated sound. This is produced by the long buildup and decay periods of the excited panels, compared to the short periods associated with a well constructed loudspeaker. The qualitative effect is one of poor vocal and instrumental definition, of blurriness of tone and lack of crispness and clarity. Such panels are spoken of as having a high mechanical Q or ratio of mechanical reactance to mechanical resistance.

Increasing the damping of a mechanical vibratile surface reduces its radiated sound output. A common example of damping is seen in the application of an undercoat, such as mastic or glassfiber-board, to automobile hoods, fenders, and door panels. Such materials act in two ways: (1) to convert vibratory energy into heat by mechanical resistance; and (2) to lower the vibration amplitude by providing a higher mechanical impedance.

It should be noted that damping is most effective at the resonant frequencies, since the larger bending amplitudes alternately stretch and compress the damping material, which thus dissipates the vibrational energy in its mechanical resistance. Adding mass to the panel causes it to vibrate with a smaller amplitude for the same applied exciting force. If the panel were mass-controlled, its velocity would be F/M where F is the vibromotive force and M is the mass of the panel.

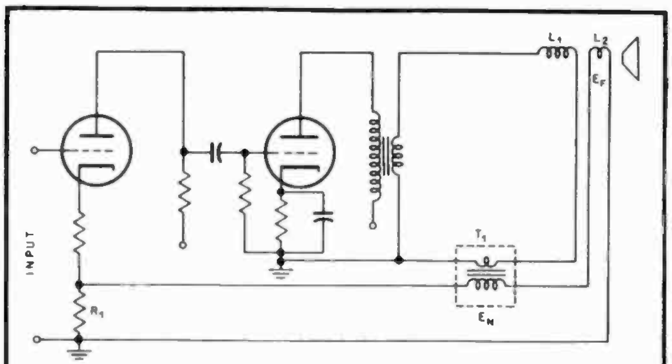
LOUDSPEAKER DAMPING

(*from preceding page*)

The basic problem is one of separating the movement-generated voltage from the driving voltage so that only and at all times will the feedback signal resemble the actual cone movement, in-phase current feedback succeeds in doing this part of the time only. True it serves best during those critical periods of hangover, but hangover is only part of realism. It follows usually in the wake of transients which themselves would be much improved if motional feedback could be put to work on a full-time job.

Source of Feedback Signal

It is not difficult to produce separate and independent movement-generated voltages. Every microphone does it. A separate winding on the voice-coil former suggests itself but this is only a partial answer for it will have some mutual inductance with the voice-coil that will mean voltage will be induced in it other than that due to motion. The apparent solution is to shield the windings from each other magnetically. This seems a difficult problem since the pick-up windings should be as close as possible to the voice-coil from the standpoint of minimizing time lag between driving voltage and pick-up voltage. Such a time lag is due, of course, to sound traveling at a rather slow speed even through a solid. If it were not kept at a minimum serious phase shifts would occur at the higher frequencies and would limit the introduction and effectiveness of feedback. Another approach suggests itself and this is likely the simplest solution: use of neutralizing voltage. Feedback windings wound over a voice-coil, or near one end would be the same as a transformer; current through the voice-coil would induce a voltage in them. If another simple transformer of special design were used in series with the voice-coil to allow the same voice-coil current to produce a secondary voltage equal to that induced in the feedback windings it could be used to cancel the voltage so induced. This transformer could have a low-impedance primary since the secondary can consist of many turns to produce the right voltage. By this device then, it should be possible and feasible to use motional feedback with all the advantages that the principle implies. *Figure* 2 shows a suggested circuit.



L1 VOICE COIL
L2 FEEDBACK COIL
T1 SPECIAL NEUTRALIZING TRANSFORMER
EN NEUTRALIZING VOLTAGE THIS VOLTAGE IS EQUAL TO AND 180° OUT OF PHASE WITH THAT COMPONENT OF EF WHICH IS DUE TO THE INDUCTIVE COUPLING OF L1 AND L2

Fig. 2. Typical circuit employed a movement - generated feedback voltage. The use of resistor R_1 is optional, since the feedback could be applied in any of a number of ways.

Although the author has believed for some time now that a feedback signal due to voice-coil movement would be considerably better to use than any signal existing within an amplifier itself, this positive feedback discussion has helped to stimulate and crystallize thinking on the subject. It can be seen that theoretically the resulting improvements of using an independent feedback signal as described would not be limited to improved damping. It would act to make the voice-coil and cone follow more exactly whatever voltage waveshape was applied, and at all times.

It is felt that much credit belongs to Warner Clements who seems to have stirred up this thing in the first place and to Ulric Childs who gave him some good arguments.

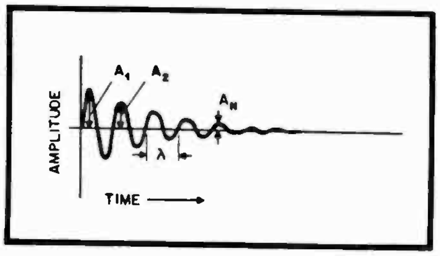


Fig. 1. Damped wave train resulting from stimulus to panel, showing two wave maxima a given number of cycles apart at which measurements are made and on which computations of logarithmic decrement are based.

Damping Panels

There are a number of ways by which damping can be added to a loudspeaker cabinet panel. One means consists in building the panel up in layers of wood, fastened together at numerous points by screws. Bending of the panels causes the sandwiched layers to rub against each other, thus creating frictional damping. Another means is to use laminated panels composed of alternate layers of plywood and felt, cork, rubber, or lead.

A convenient if perhaps not very rigorous way to determine the frictional coefficient or damping constant of such a compound structure is to take a sample of the material between thumb and forefinger of the left hand and rap it with the knuckles of the right hand or strike it with a stick and observe the duration of the resultant sound. A vibratile plate so excited behaves in much the same way that it would if excited by sound waves, and the duration of the noise produced is a fair measure of its damping properties. If it radiates a dull thud the damping is high, while a long ring would indicate low damping. Some experience, of course, is necessary to correlate the sound given off with the frictional resistance, particularly if measurements are to be made on a wide variety of test panels.

In the writer's experience, two types of cabinet panels have proven especially effective as a laminated structure to give damping of a high order. One consists of two ½-in.-thick plywood panels with a ¼-in.-thick neoprene layer sandwiched between them. The other consists of two such plywood panels with a sandwiched filler of corrugated, waffled, or honeycombed plastic of the type frequently used for aircraft panels. Using the damping material as a filler is far more effective than applying it to one side of the panel only, because as a filler it can act in shear, while applied to the exterior of a panel its action is less restrained.

If the damping material not used as a filler is sound absorbent, certain other desirable results occur, of course. These have frequently been discussed in connection with the interior acoustic treatment of loudspeaker cabinets. Briefly, such materials reduce the standing-wave effects in the enclosure and make for an acoustically larger volume. Some compromise may have to be effected in this respect in practice, because the economic aspect of a cabinet is also important. But from a purely idealistic point of view, the most effective way to damp a panel is to use the damper as a filler of the (compound) panel. If additional (sound-absorptive) material is required on the inside of the cabinet, this constitutes an additional requirement.

An exact evaluation of the damping quality of a material is not simple, however. What is usually done in the laboratory is to suspend the panel from the ceiling at one or two points by fine wires or cables. A ball of wood, rubber, or cork —a regulation baseball has been found very satisfactory for this purpose—also hung from the ceiling, is permitted to fall from a predetermined height to strike the panel, to which a vibration pickup has been fastened. It is important that this pickup be very light; otherwise it will itself act as a damper. The output of the pickup is displayed on an oscillograph, so that the successive reduced amplitudes of the transient wave-train can be measured. The quantity of interest is the ratio of successive amplitudes spaced a full cycle apart. Some books prefer to consider the amplitudes when spaced a half-cycle apart; but a study of successive amplitudes on the same side of the time axis appears more common.

What is known as the logarithmic decrement—the vital quality in all these tests—is given by

$$d = \log_e \frac{A_1}{A_2}$$

where A_1 is the first and A_2 the second measured amplitude. In practice, the evaluation of d does not usually comprise taking the logarithm of the ratio of two successive amplitudes (since they may be but slightly different to allow their exact determination), but consists of determining the ratio of two amplitudes which are a number N cycles apart. In that case (see *Fig.* 1)

$$d = \frac{1}{N} \log_e \frac{A_1}{A_N}$$

where A_1 is the first and A_N the nth amplitude.

It is, of course, also possible to excite the panel acoustically with a loudspeaker placed near it, and to record the output from a vibration pickup fastened to the panel as the frequency is varied. This might be termed a steady-state transmission measurement (with and without the damping material applied to the panel). Or the vibration pickup may be fastened directly to a wall of the cabinet with the speaker excited sinusoidally by an oscillator. *Figure* 2 shows a measurement made in this manner, first without and then with the damping material applied to the interior of the enclosure. Since the vibration pickup was inertia-operated, its output was first transmitted to an integrating network to obtain a measure of the change in panel deflection amplitude.

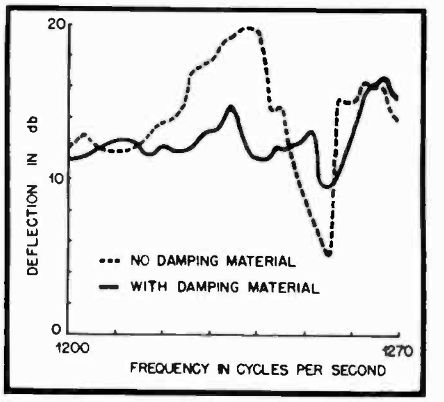


Fig. 2. The effect of resonance on a panel is much decreased when damping material is used as a filler.

Measurements on various types of woods show that the logarithmic decrement of ¼-in. panels is as follows:

WOOD	LOGARITHMIC DECREMENT
Pine	.02
Beech	.03
Plywood	.04

The superiority of the plywood as a material with high internal damping is readily evident from these figures, which pertain to untreated surfaces. When a mastic compound is applied to them and permitted to harden, the values can be still further increased.

The above tests require considerable laboratory facilities, and hence cannot be readily carried out by the average music lover, home owner, or hi-fi enthusiast. A number of less complicated tests have been proposed to show up the resonant vibrations of a speaker cabinet panel. Thus, ripples in a glass of water set on the enclosure can be generated, and fine sand, salt, or sugar crystals sprinkled on the panel can be made to form patterns, so-called Kundt's figures. However, relatively large panel amplitudes are required to achieve noticeable results with these tests. A rather simple procedure consists in placing two pencils on the panel, and a light tin can filled with lead shot on top of the pencils. When the speaker is driven with an oscillator, the lead shot will rattle in the tin pan very audibly when the undamped panel is set into vibration at the resonant frequencies. After properly damping the panel, and properly energizing the speaker again with the same current used previ-

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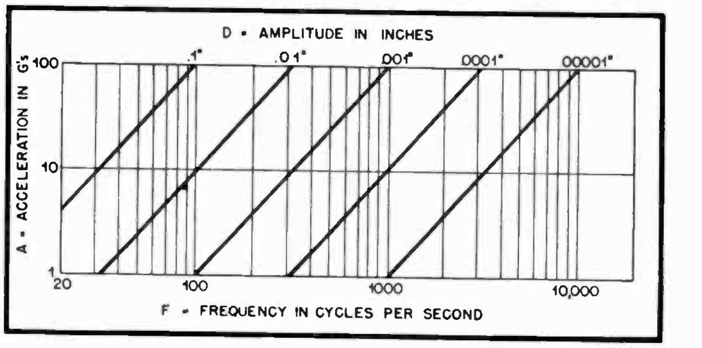
Fig. 3. This chart shows the acceleration required at various frequencies to produce given cone excursion amplitudes. Accelerations are very large and the movements "hit" cabinet panels very hard, so that panel resonances produce maximum effects.



Speaker Treatment for Improved Bass

CAMERON BARRITT*

A simple method of lowering the effective low-frequency limit of inexpensive speakers, but not one that is recommended for high-quality woofers.

A DESIRABLE FREQUENCY RESPONSE characteristic for an audio reproducing system requries adequate lows as well as highs. Amplifiers are available which can handle the necessary range, but the loudspeaker is the bottleneck in the drive toward more perfect audio reproduction. The problem of rendering sufficient highs can be met by various means such as the employment of "tweeters"—some of which can go close to the upper limit of human audibility—but the low-frequency speakers or "woofers" usually fall far short of the lower limit. Sound is propagated at frequencies as low as 16 cps, and 20 cps is usually considered as being audible. LP records are capable of going down to 30 cps, by way of further example, but not many speakers go this low.

The lower limit of a properly baffled speaker is determined by the mechanical resonant frequency of the vibrating piston or cone. A certain amount of propagation occurs below this frequency but response falls off rapidly beyond. The resonant frequency of a mechanical vibrating system is determined by the mass and the compliance—the inverse of stiffness. These two are the electro-mechanical analogs of inductance and capacitance. The mechanical resonant frequency formula is $f_r = 1/2\pi\sqrt{mC}$. The mass of a speaker cone is determined for the most part by constructional requirements for rigidity, and is not readily altered. Furthermore, if one were to increase the mass very much in order to lower the resonant frequency, phase response would suffer and poor transient response might result. For best reproduction of transients a low m/C ratio is required. On the other hand, the compliance of the speaker depends on the "give" in the cone suspension or mounting and no undesirable consequences will result from increasing its value. The total compliance or capacitance of the speaker system also depends on the acoustic capacitance of the air volume in the baffle if it is the totally enclosed type. The equivalent electrical circuit[1] of a loudspeaker mounted in an enclosed baffle is shown in *Fig.* 1. Notice that the two analogical capacitances are in series. If the baffle is made as large as possible, it will be so much larger a capacitance than the cone compliance that the limiting value will be in the suspension. The remaining reactance — the inductance representing the quadrature component of the speaker's air load—is usually small enough to be ignored (except in the case of the R-J speaker enclosure where it is exalted to make possible a smaller speaker enclosure). Consequently that leaves just the cone suspension compliance on which to work in improving the bass range of a speaker.

Since the baffle is of no concern, one can use the "free-space resonant frequency" of the speaker as the working criterion. Although the mass of a speaker cone can not be controlled

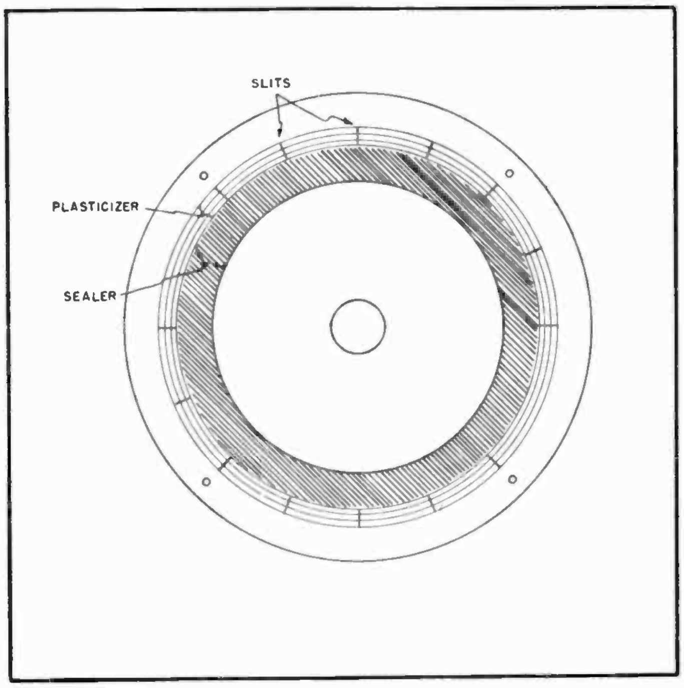


Fig. 2. Method of slitting cone, as described by the author, and location of shellac sealing ring.

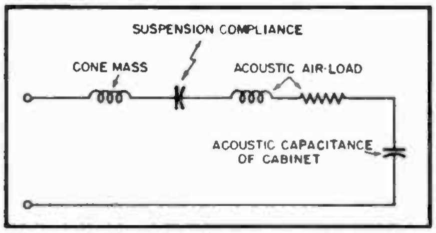


Fig. 1. Equivalent electrical circuit of a loudspeaker mounted in a baffle.

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[1] Harry F. Olson, "*Elements of Acoustical Engineering*." McGraw-Hill.

DAMPING OF CABINET PANELS

(from preceding page)

ously, no rattling of the lead shot should occur.

Figure 3 shows the acceleration for various cone amplitudes. It is seen, for instance, that at 100 cps, a cone vibrating with an amplitude of 0.1 in. undergoes an acceleration of 100 G's. Since a modern dive bomber can rarely exceed 10 G's without shattering, the acceleration for loudspeaker diaphragms is very large indeed.

For the same reason, there are great forces acting also on a speaker enclosure which is closely coupled to the speaker. One means of reducing the transmission of these forces into the speaker panels is to employ a damping material such as a rubber or cork gasket between the speaker and the cabinet. If the speaker is not very heavy, the gasket may be cemented to both the speaker and the cabinet, and the speaker thereby more effectively isolated from the cabinet.[1]

[1] RCA makes a speaker cone with a rubber damping ring. This ring should not be used for cementing the speaker to the cabinet, because the ring is intended for another purpose.

enough to vary this frequency very much, the cone suspension can be treated to make it more compliant and this would lower the resonant frequency the same as would increasing the capacitor in an electrical "tank" circuit. There are two methods of doing this, and by utilization of both it is possible to lower the frequency response limit 30 per cent or more. A relatively inexpensive 8-inch speaker treated by this system can be made equal in bass-range capability to a good quality 12-inch speaker.

Methods of Increasing Compliance

The first method is called "slotting" by the makers of Permoflux loudspeakers. As shown in *Fig. 2*, the speaker is made less stiff by radial incisions in the suspension. The improvement derived from this treatment depends on the merit of the suspension design beforehand. The suspension area of some good modern speakers is extremely thin, as can be seen when it is held up to a light, and the small amount of improvement possible is effected with a relatively small number of slits. In older speakers with stiff suspensions it appears that more slits are better. A wise approach is to make a small number of slits at first—about sixteen for example—then if the improvement is not what was anticipated, to try more between the first set. A pair of drafting dividers is used to space the slits evenly. They are set by trial so that when "walked" around the edge of the first corrugation they return to the same spot with integral spacing. Then they are walked around again allowing the points to prick the cone and thus mark the correct location for each incision. A sharp razor blade is used to slit carefully as close along a radial as possible, from the middle of the first corrugation (which is the cone edge) across the rim to the area where it is cemented to the "basket." The rim should not be torn and the cuts should go all the way to the bottom of the following corrugations. If the suspension at the apex of the cone, the so-called "spider," is also of the corrugated type, slitting here will afford an additional lowering of the resonant frequency. Often this is difficult or impossible to do, however, and might affect the high-frequency response.

The second means of improvement is called "plasticizing." Plasticizing is accomplished by applying a suitable chemical agent to the suspension to make the fibers more pliable and yielding. The plasticizing described herein was accomplished with Dibutyl Phthalate which can be obtained from large chemical supply houses.[2] Dibutyl Phthalate appears to be a good plasticizer in not drying out over a period of time; but it does have the marked disadvantage of "creeping," that is, spreading through the cone by capillary action. This would be disastrous if allowed as the whole cone instead of just the suspension would thus be made pliable and piston action would suffer, resulting in low efficiency and weakening of high frequencies. To prevent this condition the edge is first made impervious to capillary travel by sealing the pores with an application of shellac, varnish, or cement. This means the addition of a small amount of mass to the cone, and a further lowering of the resonance point is sometimes noticeable but rather negligible. A small soft brush is used to paint the cone in a swath about ¾-inch wide (for large speakers) just inside the curvature of the first corrugation—on *both* sides of the cone. Clear shellac was used in most of the treatments attempted, and with a fair amount of success, but it tends to break down under reaction from the Phthalate if the cone is very well felted. Recent but rather inadequate experience with so-called "radio" cement would seem to recommend it because it does not appear to be affected by the plasticizer. It is important that a good job be done in blocking the creep of the Phthalate, because if it once gets started through, it is difficult to stop it. One should apply several costs of the sealer if inspection deems it advisable. When this operation is finished and the sealer is dry, the plasticizer may be applied with a cotton swab. Distributing it in even amounts around the periphery of the suspension is desirable, but there is no worry about spreading it laterally as the Phthalate spreads itself. If it should begin to work past the barrier after ageing, another swath of sealer may be applied to stop it.

[2] Eimer & Amend, New York City, for example.

Results Obtained

By utilizing one or both of these two methods of suspension treatment, the results shown in the chart may be obtained. The improvement in low-frequency capability is evident. We have thus far stressed only the advantages in bass, but no really deleterious effects to the higher ranges have been noted. It would seem that the slitting operation could conceivably cause edge effects resulting in irregularities in the middle frequency range. One of the most pronounced peaks of a speaker's middle frequency response is due to the resonance of the edge.[3] The mechanical alteration of the edge might aggravate the peak somewhat on the less expensive speakers. A small change in frequency of some of the peaks in a loudspeaker's curve has at times been observed, but the extent of the variations has not been quantitatively ascer-

[3] Corrington and Kidd—"Measurements on loudspeaker cones," *Proc. I.R.E.* Sept. 1951, p. 1021.

(*Concluded at bottom of next page*)

TABLE I

Results of Speaker Treatment

Speaker	Resonant Frequency Before Treatment	After Treatment	Treatment
3" Cinaudagraph P2A1	230	205	Eight slits
		190	Sixteen slits
4"×6" RCA 446S2	195	155	Sixteen slits
		147	Thirty-two slits
		140	Same, plus cement sealer
		130	Same, plus three applications of plasticizer, ageing after each
5"×7" RCA 257S1	110	80	Thirty-two slits
		75	Same, plus shellac sealer
		58	Same, plus five applications of plasticizer, ageing after each
6"×9" RCA 269S1	100	80	Thirty-six slits
		72	Same, plus sealer and plasticizer
		70	Same, plus ageing
		63	Same, plus second application of shellac sealer and ageing
		55	Same, plus four slits in spider and more ageing
8" R&A (British) 880P	108	90	Twenty-five slits
		80	Fifty slits
		72	Same, plus plasticizer
		62	Same, plus second application
		58	Same, plus two more applications of plasticizer, and slitting spider
10" PM 10-12	75	63	Thirty-six slits
		48	Same, plus three applications of plasticizer and ageing
12" GE S-1201D	73*	70	Sixteen slits
		68	Same, plus plasticizer
		50	Same, plus four slits in spider, three more applications of plasticizer, and ageing. (Capillary travel is very marked in this cone.)

* Some speakers of this type come with a resonant frequency as low as 63 cps.

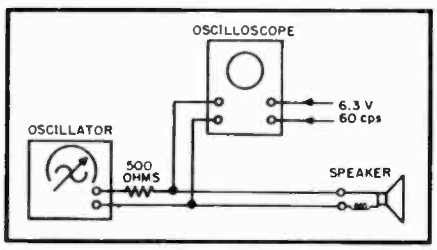

Fig. 3. Circuit arrangement used to determine resonant frequency of loudspeaker.

Improving Loudspeaker Performance

DAVID B. WEEMS*

How to improve low-frequency response from inexpensive loudspeakers with the expenditure of less than a dollar and an hour or so of your time.

Among the most important characteristics of the ideal speaker are wide-range frequency response and low distortion. The fundamental cone resonance of a speaker is closely associated with both its bass range and its distortion content at low frequencies, because the output falls off rapidly with an increase in intermodulation effects and frequency doubling below resonance. It is a well known fact that the proper baffling of a loudspeaker in such mountings as bass-reflex and horn-loaded enclosures lowers the resonant frequency, and an amplifier that presents a low impedance to the speaker will minimize the effects of resonance. We are concerned, however, with the qualities of the speaker itself and how we can improve them.

The fundamental resonant frequency of a loudspeaker may be determined mathematically from the formula:

$$f_r = \frac{1}{2\pi \sqrt{M_c C_{ms}}}$$

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where f_r = resonant frequency of the speaker in cps,
M_c = mass of cone, voice coil, and air load in grams,
C_{ms} = compliance of the suspension system in centimeters per dyne.

Assuming a constant air load, examination of this formula shows that as either the mass or the compliance of a cone is increased, the resonant frequency is lowered. While increasing the mass of the cone lowers resonance, it also limits high-frequency response and results in a deterioration of transient ability. Increasing compliance thus seems to be the best approach toward lower fundamental cone resonance. High compliance also aids in obtaining wider range and lower distortion.

It may be argued that higher compliance in cheap units may produce greater intermodulation distortion due to the movement of the voice coil beyond the area of maximum flux density. There are several answers. The first is that at normal volume levels such movement will probably not occur. Secondly, if it does occur the use of muitiple woofers will eliminate it and at the same time provide one of the most practical means available to the home constructor of accomplishing truly superior low-frequency reproduction. Finally, regardless of the theoretical possibility of excessive voice-coil movement, the net result of higher compliance for cheap speakers has been better sound. In speaker design we cannot always predict, like Old Man Mose, what will provide subjective enjoyment. With this premise firmly in mind and with courage in our hearts, we shall now proceed to the task of speaker improvement.

Procedure

The materials to be assembled are an

Fig. 1. Materials needed for treating speakers—chamois, solvent, cement, tweezers, scissors, razor blade, scalpel, and in some cases clothes-sprinkler head.

Fig. 2. Removing the dust cover with tweezers.

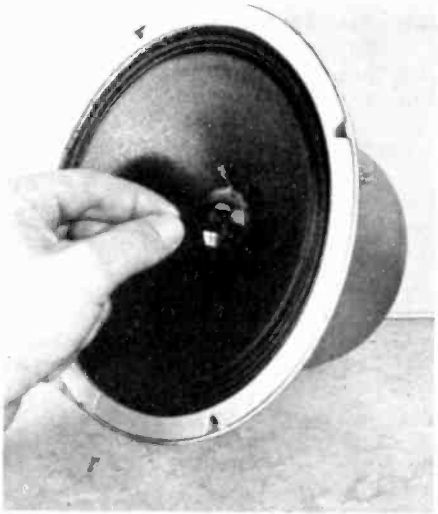


Fig. 3. Inserting shims to hold the voice coil in position.

SPEAKER TREATMENT

(*from preceding page*)

tained because of the apparent inconsequence. The plasticizer, however would have no pernicious effect as long as it is kept within its prescribed area. If one were treating a speaker intended only as a woofer, there is nothing to lose in the complete treatment and much to gain. If both methods are applied, the slitting operation must be done first because slitting a plasticized rim is quite difficult.

In checking progress the circuit shown in *Fig.* 3 has advantages. The 60-cps 'scope input is used to calibrate the audio oscillator, an operation which should be performed at frequent intervals, particularly with a beat-frequency oscillator When working with low frequencies and such small increments, accuracy of calibration is important. By setting the oscillator to 60 cps and varying the calibration adjustment until a stationary one-to-one Lissajous figure is obtained (circle, ellipse, or inclined line, depending on phase), the oscillator can be set readily without much trouble. To find the resonant frequency of the speaker turn the 60-cps input gain to zero and hold the speaker up in the air. Vary the frequency of the oscillator until the maximum deflection is noted on the 'scope. The frequency indicated by the dial is the resonant frequency. In most cases it can also be detected audibly by noticing when the cone tends to chatter, which is caused by its greater displacement at resonance. The 'scope method, which depends on the rise in speaker impedance at its resonant frequency, is reliable and can also be used for a speaker mounted in a baffle thus serving to indicate the reduction in bass range that the enclosure might cause.

As a before-and-after test of this treatment, LP records with low organ notes[4] are very useful. Often the improved speaker will bring out notes in the lower register which were not even noticed before treatment.

[4] Columbia ML 4120: Saint Saëns, Symphony No. 3; Columbia ML 4329: Poulenc, Concerto in G Minor are good examples.

inexpensive 8-to-12-inch speaker, plus (*Fig.* 1) a small piece of the softest, thinnest, and most pliable chamois available, radio solvent (optional), household glue, tweezers, small scalpel (optional), barber's scissors or other pointed scissors, a new razor blade, and an aluminum clothes sprinkler head. The chamois may require some shopping around because of the enormous variation of thickness and pliability, but usually the small, most inexpensive pieces found at the five and dime will include a few that have the desired characteristics. The aluminum sprinkler will only be required for certain very inexpensive units as will be described.

The operation should be performed on a clean bench or table. First, tear away the dust cover with the tweezers (*Fig.* 2), taking care not to disturb the voice-coil leads. Next, place shims of film negative or paper around the center pole to maintain the voice-coil form in position, as in *Fig.* 3. Usually four narrow shims located at equal intervals around the pole are sufficient. The solvent may then be used to saturate the juncture of the outer rim of the cone, the frame, and the gasket. While the solvent is acting on the glue, the rim of the cone may be cut away at the middle of the inside corrugation (*Fig.* 4) leaving a "lip" past the smooth part of the cone. The entire outer corrugations and gasket may now be lifted out. If solvent was not available, this part can be loosened with a knife after it is cut loose from the cone.

The spider should be cut away with either the scalpel or pointed scissors, leaving four strips about ½ to ¾ inch in width at equal intervals (see *Fig.* 5). Now attach four strips of Scotch tape to the outer rim of the cone, evenly spaced, and then fasten them to the frame without applying tension to the cone. If desired, the shims may be removed and the speaker tested for trueness of centering. If it will respond to loud musical passages that include low frequencies without rattling, the assumption may be made that the centering was adequately done, and the shims may be replaced as before.

The Scotch tape should be replaced, one piece at a time, by pieces of chamois about 2 inches wide (see *Fig.* 6). The chamois pieces should first be glued to the lip left when cutting away the rim of the cone, then to the frame. The chamois should be tightened just enough to remove wrinkles but not stretched. After this operation the speaker may be tested again.

Now the four large gaps around the cone rim may be filled by cutting large pieces of chamois to fit them. These large sections should be installed very loosely, leaving a deep wrinkle between the cone rim and the frame as shown in the end-product photo of *Fig.* 7. Their purpose is to eliminate the exchange of air between the rear and front of the cone, not to support and impede it; so they are left loose, allowing only the four narrow strips to suspend the cone. The gasket should be trimmed of original cone remnants and reglued into position. The dust cover may or may not be reused, depending on the ultimate purpose of the speaker as described later. In every instance the glue should be used sparingly, especially on the cone. When the chamois is glued to the frame, the glue should be spread near the outer rim of the frame only; this will allow a longer chamois suspension and preserve its pliability.

When the job is completed, we have a cone that is suspended at four points by the spider, and at four points on soft chamois at the outer rim. It would be difficult to conceive of a more compliant suspension. An ordinary replacement speaker, purchased for about five dollars in a radio supply store was thus treated and then compared to a 15-inch speaker costing more than 10 times as much. The cheaper 12-inch unit appeared to be distinctly cleaner, although the overall response was admittedly somewhat rough and shrill due to inherent characteristics and the removal of cone resonance as a contributor of bass.

This phenomenon of shrillness may be removed by two general methods, the use of the speaker as a woofer only, or frequency correction of the speaker itself. The latter may be achieved by a mechanical means that the writer has found surprisingly effective and that seems to smooth the high range. It is to be recommended only for inexpensive speakers of the replacement class (and then only on 10 and 12-in. units), but these are the units that most need further alteration.

The method consists of gluing a small perforated aluminum dome over the center of the cone in place of the dust cover. This dome may be obtained from an aluminum clothes-sprinkler head. The sprinklers are obtainable from most variety stores for about a nickel. The rim that clamps the dome in place should be removed by inserting a sharp knife blade under it and lifting it away from the dome. A small amount of household glue should then be placed around the rim of the dome, and after positioning, a small weight on the dome will hold it in place until the glue has set. The speaker should now be ready to mount and try out.

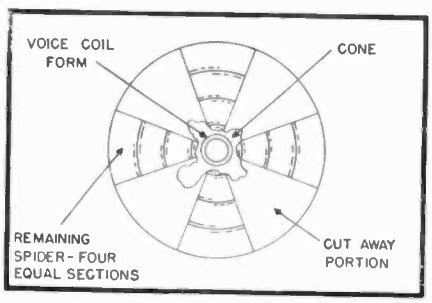


Fig. 5. The spider is cut away except for four sectors which hold it in position while allowing greater compliance.

Fig. 4. Slitting the outer edge of the cone to remove it from mechanical engagement with the frame.

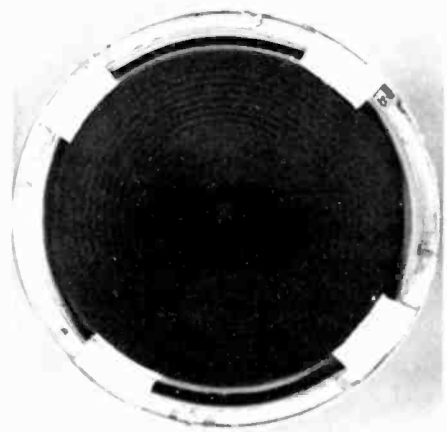


Fig. 6. Four strips of the soft chamois are glued in place to hold the cone centered.

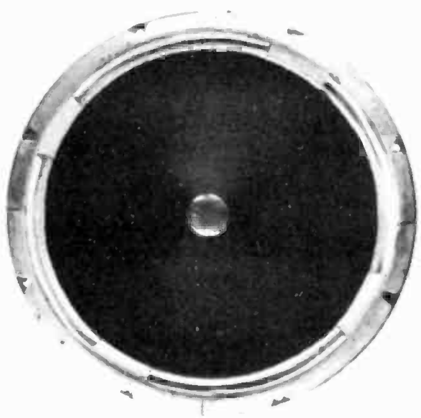
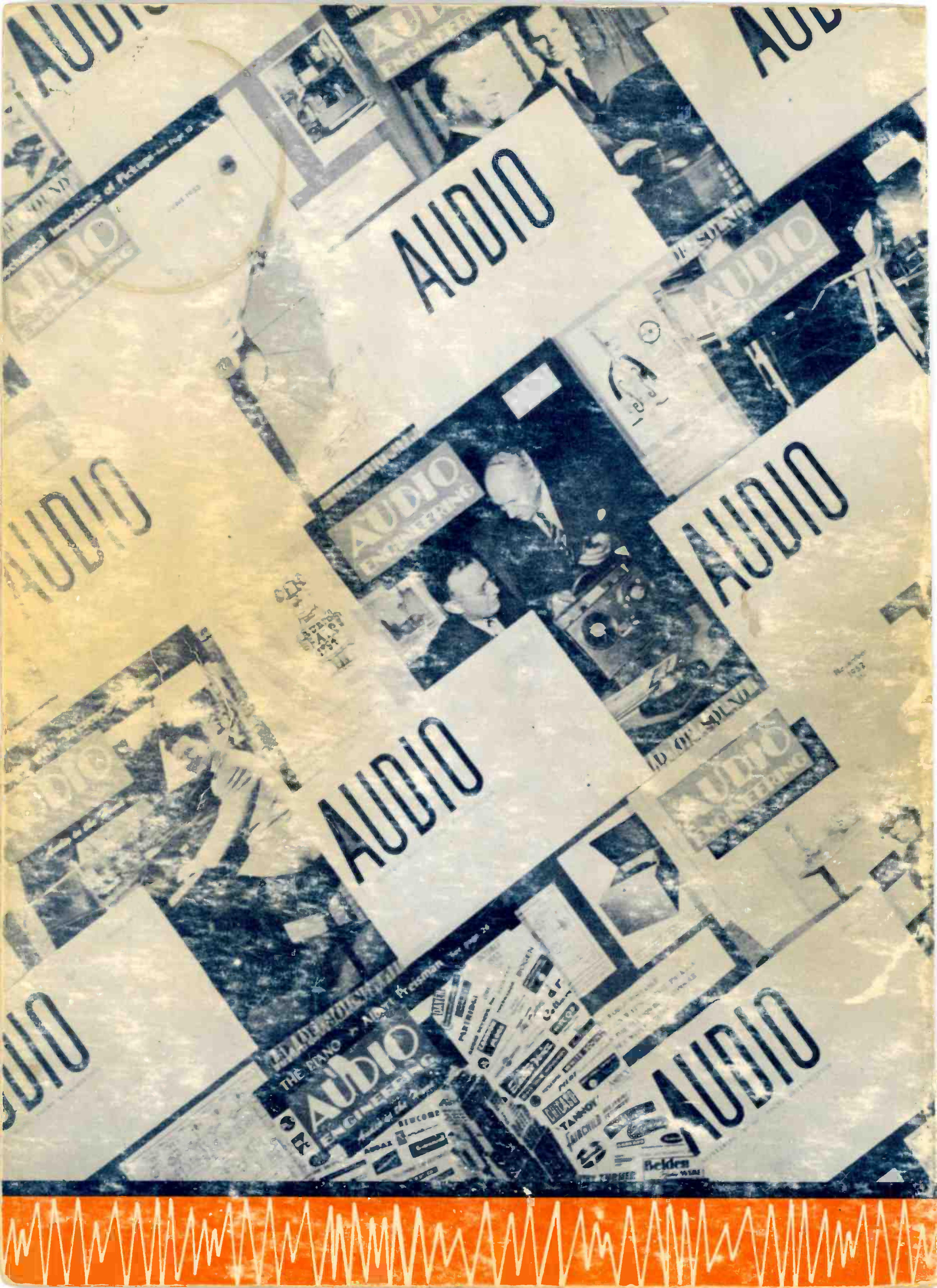


Fig. 7. Spaces between the four original chamois strips are filled with loosely hung chamois to help seal front off from rear air without restricting cone movement.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