Common-Bass Stereo Speaker System

FRANCIS F. CHEN*

One speaker can handle the bass of both stereo channels if the crossover frequency is low enough to avoid directionality.

There are two common problems that the hi-fi hobbyist often faces. One is that his good wife, understanding though she may be in other ways, objects to the "Laboratory Look" in her living room. The other is that his good wife, music-lover though she is, thinks that one-half of the family budget is too much to spend on electronics. Now everybody who reads this magazine knows that you can't get good music reproduction without a certain minimal outlay of cash or without a certain minimal number of components and interconnecting cables, which may not all look beautiful. And *nearly* everybody does not have the inexhaustible capital and engineering time necessary to build one of the "ultimate" systems that so often appear in print.

The more realistic problem of getting highest quality and versatility (important to the hobbyist) within the ever-present restrictions of cost and decor is a challenging one. To solve it, one must give careful thought as to what is important and what is superfluous. To be sure, in the present case the problem of cost was solved not only by "Doing-It-Myself" and by careful choice of components, but also to some extent by the Principle of Infinitesimal Accretion ("You don't mind, dear if I get a couple of EL-34's this week?"); but the latter ploy is at any rate a useful one for the hobbyist to have at his command.

The Common Woofer

The first problem considered was that of the most expensive part of a hi-fi system. Why do good speakers cost so much? The answer is that they do not, as long as one sticks to 8-in. speakers. It is only the 12- and 15-in. speakers that bear the disheartening price tag. It is, however, an inescapable fact that good bass requires large piston area and low resonance. The obvious solution, then, is to use three speakers: two matched 8-in. units for the mid-range and treble of the two channels, and a 15-in. woofer for the common bass. This costs considerably less than two 15-in. full-range units. The savings are partially offset by the need (to be explained later) for three amplifiers instead of two, and for a crossover unit. The latter can easily be constructed, however, using the two-tube circuit to be described later; and it turns out that three amplifiers, of which only one need handle the bass, do not necessarily cost more than two amplifiers, both of which must produce, say, 25 "clean" watts at 30 cps. Again the costliness of good bass has made itself apparent, this time in terms of the power handling capacity needed in the amplifier, and in particular in the amount of iron needed in the output transformer.

At this point I should note that in the choice of a woofer I considered using a battery of, say, 20 or 30 small, cheap speakers in a series-parallel array, instead of using a single 15-in. speaker. The piston area would indeed be large, and one might hope that the distortion would be small in spite of the cheapness of the magnets, because the cone excursion of each speaker would be small. However, besides being a somewhat clumsier arrangement, this method would also entail some risk in that the flimsy magnets may not properly damp the cone motions. Besides, 20 or 30 speakers are not exactly cheap. It was therefore decided to leave the multiple-speaker array to possible future experimentation, in spite of several favorable reports in the literature.

Crossover Frequency

The next problem to consider was the choice of a crossover frequency. Now it is a well-known physical principle that two signal sources cannot be distinguished directionally if they are separated by a distance of the order of magnitude of the wavelength emitted. Since the geometry of my living room requires the speakers to be about 13 feet apart, and since a wavelength of 13 feet corresponds to a frequency of 85 cps, it would appear that a crossover well below 100 cps would be necessary to avoid losing any stereophonic effect through the use of a single woofer for both channels. However, the dividing line is a fuzzy one: one cannot say that at 100 cps there is definitely no directional effect, while at 110 cps, or 150 cps, there definitely is. Aside from the vagueness of the physical principle cited above, the acoustics of the room and the psychology of hearing would also enter in; and indeed, common-bass stereo systems have been made with crossover well above 100 cps. The decision, therefore, was to make the crossover frequency as low as possible, consistent with other limitations.

In listening tests, it may sometimes appear that a low note, below 100 cps, has some directionality. This is probably due to the sudden onset of the note. This initial transient consists of higher frequency components and would be reproduced by the tweeter. For this reason, directionality should be tested only with steady tones.

The "other limitations" mentioned above are the ones imposed by the bass response of the 8-in. speakers. The units chosen were Wharfedale Super-8 FS/AL's, which have high flux density, good efficiency all the way to 15,000 cps, smooth response, and a soft suspension. The free-air resonance of these speakers is around 70 cps. Because of the decor of the living-room, these speakers had to be mounted in different types of cabinets; therefore, differences in cabinet resonances would cause the speakers to be mismatched in the low-frequency region. For this reason, it was felt that the crossover frequency should occur at least an octave above the free-air resonance of the speakers, and a frequency of 150 cps was chosen.

Type of Crossover

The usual type of crossover, an LC circuit inserted between the speaker and the amplifier, cannot easily be obtained at a frequency as low as 150 cps. The reason is that the capacitor has to be large—in the neighborhood of 100 μf. This would be most unwieldy unless one used an electrolytic. However,

* *P. O. Box 31, Kingston, N. J.*

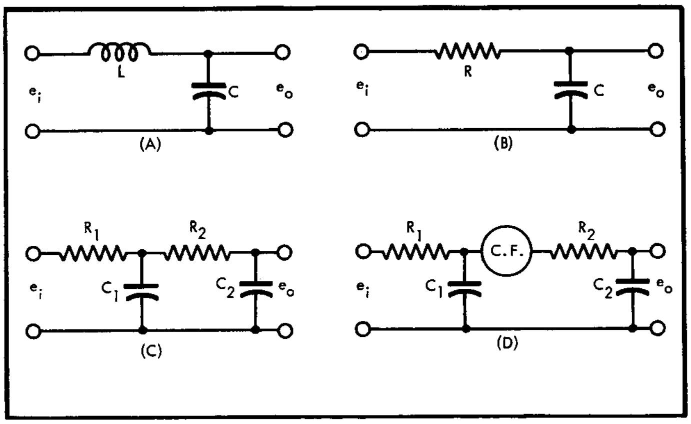

Fig. 1. Simple crossover networks. Only low-pass networks are shown: the corresponding high-pass networks would be similar with the elements interchanged.

electrolytics do not maintain their capacitance value very accurately and, moreover, would have to be used "back-to-back" (requiring double the capacitance in each) in order to "hold off" the a.c. voltage. When used this way the capacitance of the electrolytics can change suddenly in the middle of a loud passage! The inductor, of the order of 10 mh, would also be a large affair, since it would have to be wound with heavy wire to avoid a large insertion loss.

The alternative is to use an "electronic" crossover ahead of the amplifier, and this is the more sophisticated and the more soul-satisfying way of doing it. This would require a separate amplifier for each speaker, but, as mentioned before, this is not necessarily more expensive. By having separate amplifiers and speakers for the highs and the lows, one gains in addition a most attractive bonus: intermodulation distortion (except in the program source) is for all practical purposes completely eliminated!

An attenuation of 12 db per octave is generally recommended for the crossover network. One can achieve this by a half-section LC filter, as in (A) of *Fig.* 1, or by two simple RC filters, as in (C) or (D) of *Fig.* 1. The LC filter has a distinct advantage in that the phase shift is 0 deg. on one side of the crossover frequency and 180 deg. on the other. Thus one can bring the high- and low-frequency speakers into phase by merely reversing the leads on one speaker. However, the use of an inductor at the preamp level would be asking for trouble with hum pickup. This leaves the RC network, with its horrible phase shift characteristics, and leads us to a discussion of phase shift and frequency response in low-frequency RC crossovers.

The Two-Section RC Crossover

Consider first the one-section low-pass filter in (B) of *Fig.* 1, in which the resistor, *R,* and the capacitor, *C,* form a simple voltage divider (operating into infinite impedance) for the input signal, e_i. At any angular frequency, ω, the output voltage e_o will be given by

$$\left(\frac{e_o}{e_i}\right)_L = \frac{1}{j\omega C} \bigg/ \left(R + \frac{1}{j\omega C}\right) \qquad Eq.\ (1)$$

If we define the crossover frequency, ω_o as $1/RC$,

$$\left(\frac{e_o}{e_i}\right)_L = \left(1 + \frac{j\omega}{\omega_o}\right)^{-1} \qquad Eq.\ (2)$$

The corresponding high-pass network, with *R* and *C* interchanged, would give

$$\left(\frac{e_o}{e_i}\right)_H = R \bigg/ \left(R + \frac{1}{j\omega C}\right) = \left(1 + \frac{\omega_o}{j\omega}\right)^{-1} \qquad Eq.\ (3)$$

At frequencies which are low compared with ω_o, *Eq.* (2) shows that $(e_o)_L \approx (e_i)_L$, and at frequencies which are high compared with ω_o, *Eq.* (3) shows that $(e_o)_H \approx (e_i)_H$. At frequencies near ω_o, there is a phase shift, since there is a sizable imaginary part to *Eq.* (2) and (3); and the question arises as to how the outputs from the high- and low-pass networks are to be added.

If the two signals are added together first and then fed into the same speaker, the acoustic output would be found by adding *Eq.* (2) and (3) and then squaring the result:

$$\left[\left(\frac{e_o}{e_i}\right)_L + \left(\frac{e_o}{e_i}\right)_H\right]^2 = \left[\frac{1}{1+\frac{j\omega}{\omega_o}} + \frac{1}{1+\frac{\omega_o}{j\omega}}\right]^2 = 1 \qquad Eq.\ (4)$$

Even though the phase shift is 45 deg. in the critical region around $\omega = \omega_o$, the phases are such that the vector sum is always unity. Thus the frequency response is flat throughout. The same would be true, but not as exactly, if the two outputs were fed to two speakers right next to each other. However, if the two speakers were far apart and the crossover frequency fairly high, the total acoustic intensity would be the sum of those from each speaker; that is, the voltages would not add in phase and would have to be squared before adding:

$$\left(\frac{e_o}{e_i}\right)_L^2 + \left(\frac{e_o}{e_i}\right)_H^2 = \frac{1}{1+\left(\frac{\omega}{\omega_o}\right)^2} + \frac{1}{1+\left(\frac{\omega_o}{\omega}\right)^2} = 1 \qquad Eq.\ (5)$$

This is also equal to *1*! From this standpoint, the single RC network is the ideal crossover; the frequency response is flat no matter how the signals are added.

Incidentally, you can easily see that for the LC circuit of (A) in *Fig.* 1, e_o/e_i is real, and there is no difficulty with phase shifts. However, just *because* e_o/e_i is real, the equations analogous to *Eq.* (4) and (5) cannot both be true; only *Eq.* (4) is true in the case of the LC circuit.

The trouble with the single RC section is, of course, that it rolls off at only 6 db per octave, which usually does not provide sufficient isolation of frequencies. In our particular case, with f_o at 150 cps, this means that the cabinet resonances of the 8-in. speakers at around 70 cps will be only 6 db down—not a very great difference to the ear. The next logical step would be to try two RC sections in cascade, as in (C) of *Fig.* 1. Here, if R_1 and C_1 are equal to R_2 and C_2, the first section would not be working into a large impedance and would not provide as fast a rolloff as it should. This can be improved if the second section is made higher in impedance than the first, or better yet, if the two sections are isolated by a cathode follower, as in (D) of *Fig.* 1.

In this case the response is found by applying *Eq.* (2) and (3) twice:

$$\left(\frac{e_o}{e_i}\right)_L = \left(1 + \frac{j\omega}{\omega_o}\right)^{-1} \left(1 + \frac{j\omega}{\omega_o}\right)^{-1} \qquad Eq.\ (6)$$

$$\left(\frac{e_o}{e_i}\right)_H = \left(1 + \frac{\omega_o}{j\omega}\right)^{-1} \left(1 + \frac{\omega_o}{j\omega}\right)^{-1} \qquad Eq.\ (7)$$

This has been written this way, without the exponent *2,* because by "squaring" we shall always mean multiplying a quantity by its complex conjugate, which

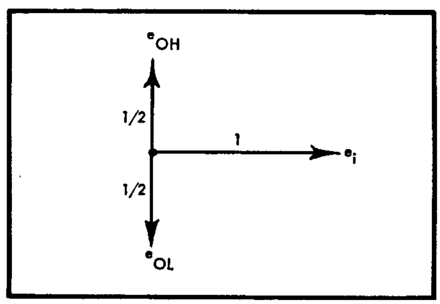

Fig. 2. Behavior of the dual RC network at the crossover frequency.

is not what is required here. Now if we square and then add, we will not get unity:

$$\left(\frac{e_o}{e_i}\right)_L^2 + \left(\frac{e_o}{e_i}\right)_H^2 = \left(1+\frac{\omega^2}{\omega_o^2}\right)^{-2} + \left(1+\frac{\omega_o^2}{\omega^2}\right)^{-2} \neq 1. \quad Eq.\ (8)$$

Moreover, the phase angle between the two signals will change with frequency. At the crossover frequency, $\omega = \omega_o$, each of the terms in the above equation is equal to ¼, so the total intensity is only ½ of what it should be.

Most previous designers have gotten around this either by negative feedback to lower e_i in the flat regions of the response curves or by using different ω_o's in the low-pass and the high-pass sections, so that at the actual crossover point, $(e_o/e_i)^2$ for each section is down only to ½. However, this still does not provide unity gain and zero phase shift at other frequencies. These considerations have been given in detail in two excellent articles by Norman Crowhurst[1,2], the latter of which, unfortunately, did not appear until my system was all finished. The point I want to make here, however, seems to have been missed in these articles, although Mr. Crowhurst touches on this in a more recent article[3] which begins to attack the most basic and difficult problems of stereophonic sound. And that point is, why should RC crossovers be designed so that the separate intensities of the two speakers add up to unity? Isn't it possible that under some circumstances the sound waves from the two speakers add in phase?

This depends on the frequency. At very high frequencies phasing cannot make any difference; there are so many reflections that phasing is all mixed up by the time the waves reach the ear anyway. At 6000 cps, the wavelength is only a couple of inches; and if phasing mattered, the cone of a tweeter would have to lie in the same plane as the cone of the midrange unit, within half an inch or so. This is impossible, since cones are deeper than that. In actual practice, I have been unable to tell the difference when the leads are reversed to a tweeter which crosses over at 8000 cps. At middle frequencies, which are important for the stereophonic effect, phasing makes a difference, but just how is a complicated business. Everyone knows, however, that if the phase were as much as 180 deg. off, the stereo effect is lost for two spatially separated speakers. Our interest now is in what happens near the crossover frequency to a midrange unit and a woofer which are not necessarily separated. In this case the effect of phasing is probably just as great, but easier to analyze. At very low frequencies, it seems to me, phasing must be correct and one must add the (complex) signals to the two speakers together first before squaring, as in *Eq.* (4) to get the total sound intensity. This must be true because the bass reflex principle is known to work; if only total intensity mattered, the back wave from the port of a reflex cabinet would add to the speaker resonance instead of reducing it.

If this is true, observe what the double RC crossover would do. If we add *Eq.* (6) and (7) without squaring them first, we would get (after multiplying numera-

[1] N. Crowhurst, "The RC Crossover Compromise," AUDIO, July 1957.
[2] N. Crowhurst, "Electronic Crossover Design," AUDIO, Sept. 1960.
[3] N. Crowhurst, "Audio Matrixing," AUDIO, Nov. 1960.

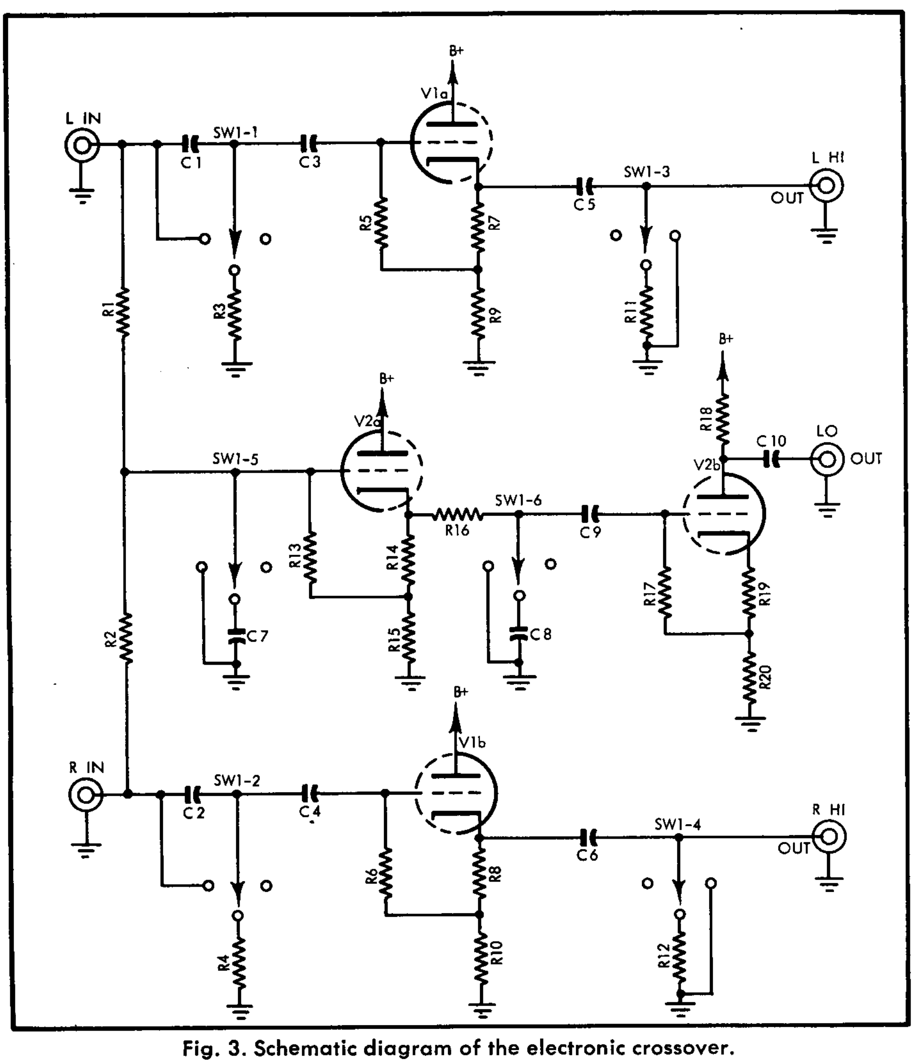

Fig. 3. Schematic diagram of the electronic crossover.

tor and denominator by ω_o^2 and $j^2\omega^2$ respectively),

$$\left(\frac{e_o}{e_i}\right)_L + \left(\frac{e_o}{e_i}\right)_H = \frac{\omega_o^2}{(\omega_o + j\omega)(\omega_o + j\omega)} + \frac{j^2\omega^2}{(j\omega + \omega_o)(j\omega + \omega_o)} \qquad Eq.\ (9)$$

Now let us reverse the leads to one speaker, so that the sign between the two terms becomes minus, thus cancelling the j^2, and then combine over the common denominator:

$$\left(\frac{e_o}{e_i}\right)_L + \left(\frac{e_o}{e_i}\right)_H = \frac{\omega_o^2 + \omega^2}{(\omega_o + j\omega)(\omega_o + j\omega)} \qquad Eq.\ (10)$$

Upon squaring, each factor in the denominator becomes $\omega_o^2 + \omega^2$, and we get unity. Voila! No shifting of crossover points with the resultant mess in phase shifts; no complicated alignment procedure to perfect a feedback circuit.

For those who like vectors, this is what happens at the crossover frequency (*Fig.* 2). The voltages from the high-pass and low-pass filters have magnitude ½ and are shifted ±90 deg. in phase relative to the incoming signal, so that they are 180 deg out of phase with each other. If we square each separately, each becomes ¼, and the sum is only ½. However, if we reverse the leads to one speaker to bring the high and low signals into phase and then add them before squaring, we get 1. Things work out equally nicely at all other frequencies, as long as such simple addition of phased signals occurs.

How low a frequency must one have before this occurs with actual acoustic signals? To determine this, a simple test was performed in the living room. Equal low-frequency signals were fed from an oscillator, through amplifiers, to two speakers. In this test the speakers were separated, but this is unimportant, since the assumption is that the crossover frequency is so low that there is no directionality. The intensity of the sound at the opposite end of the room was observed both by ear and by a microphone feeding an oscilloscope. As the leads to one speaker are reversed, the intensity should go from 0 (complete cancellation) to 4 times the intensity of one speaker alone, if the acoustic waves added in phase. Of course complete cancellation does not occur in actual practice, but at 100 cps, there was a large change in loudness as the leads were reversed. At 150 cps the change was much less pronounced. The conclusion was that the simple circuit in (D) of *Fig.* 1, with two identical RC sections in cascade, should be used with crossover frequencies below 100 cps, and that at our previously chosen crossover frequency of 150 cps, the frequency response of this circuit is only approximately flat but that it should be as good as that of the fancier circuits usually used.

The Two-Tube Stereo Crossover

Before building the circuit of *Fig.* 3, we considered the two electronic crossovers on the market. One, the Marantz, seemed to be carefully designed with feedback; but it was prohibitively expensive. The other, by Heathkit, was unnecessarily bulky for our purpose and seemed to have been designed to provide a peak at the crossover frequency.

The circuit of *Fig.* 3 serves the functions of dividing the frequencies above and below 150 cps for each channel and of adding the low frequencies from the two channels together. Cathode followers are used both to provide high input impedance and to isolate the two sections of the cascaded RC networks. Only two tubes, each a twin triode, are necessary; and the whole circuit can be enclosed in a 3-in. × 5-in. × 7-in. aluminum utility box, which also contains two octal sockets for distributing B+ and filament power to two preamps. A four-prong Jones plug receives power from the bass amplifier.

The circuit is exceedingly simple. In the left channel (the right channel is identical), C_1 and R_3 form one RC network with $\omega_o = 1000$, corresponding to a crossover frequency of about 150 cps. The cathode follower V_{1A} then lowers the impedance level so that the second RC section, consisting of C_5 and R_{11}, with the same ω_o, can be made of such low impedance elements that no output cathode follower is necessary to drive the cable to the amplifier. In the bass channel, R_1 and R_2 serve both to add the left and right signals together and to form the first RC section with C_7. The second RC section, R_{16} and C_8, follows the cathode follower V_{2A}. An amplifier stage, V_{2B}, with a gain of approximately 2 is necessary because the adding network cuts the bass gain by 2. R_{18} and R_{20} may be varied, keeping their sum constant, to change the gain of the bass channel, depending on the gain of the bass amplifier. Here the gain has been made slightly less than 2 because my bass amplifier has higher gain than the treble amplifiers. The final adjustment, of course, is to be made with the level controls on the amplifiers. When testing with a signal source in only one channel, be sure to short the other channel input to ground, or the adding network will not halve the bass gain the way it would in actual use.

The other two positions of the switch SW_1 provide crossover frequencies of 0 and ∞—that is, with the entire program going straight through to the tweeters alone or to the woofer alone. This frill may be omitted, if desired, but is quite useful for checking the system as well as for having music even when one of the speakers or amplifiers is temporarily out of commission. The impedances of the input RC sections have been chosen so that for any of the switch positions the preamps are not loaded by less than 0.5 megohm at any frequency below 20,000 cps. Such a high impedance level is possible only because of the high effective input impedance of the cathode followers. This is the reason, for instance, that the amplifier stage in the bass section cannot be the input stage. The 0.5-megohm input impedance allows the crossovers to be used with any preamp, including those, such as the Dynakit, which will not drive an impedance smaller than 0.5 megohm without internal modification. Several crossover frequencies can be incorporated and selected with the switch SW_1 if one wants to build a more versatile crossover. However, if one goes to a lower frequency than 150 cps by, say, increasing R_3, the input impedance of the cathode follower will no longer be negligible; and if one goes to a higher frequency, the ω_o's for the RC sections should be staggered to provide uniform frequency response, since the acoustic signals will no longer add in phase, according to our earlier discussion. In my unit I have incorporated crossovers at 300 and 600 cps in case the power handling capacities of the speakers have to be used to the fullest; however, the occasion has never arisen.

Precision resistors and capacitors can be used for the elements of the RC networks, but this expense is not necessary. It would be sufficiently accurate to use 20 per cent elements and then adjust R_3 and R_4 and, if necessary, C_1 and C_2 until all three sections gave an e_o/e_i of ½ at the same frequency. If an oscilloscope with a horizontal input is available, a slightly more sensitive method would be to put the input signal from an oscillator on one axis and the output from the crossover on the other and to find the crossover point by finding the frequency at which the Lissajous figure can be made into a circle. The adjustment can be made simply by adding different resistors in series or parallel with R_3 or R_4 until the crossover point occurs at the proper frequency. The exact frequency does not matter so much, of course, as the matching of the crossover frequency in the three sections.

Two warnings should be given to the constructor: first, do not turn the switch SW_1 when the loudspeakers are on, since the discharging of the coupling capacitors will cause a loud pop. Second, be sure the B+ supply is sufficiently well filtered—the amplifier section is sensitive to hum. Although the circuit was designed to operate with a B+ of 300 v, at which the tube currents are 3 ma in the cathode follower sections and 6 ma in the amplifier section, I had to add several filter sections to my B+ source (the bass power amplifier), dropping the B + to 200 v, before the hum was eliminated. Both the crossover unit and my Dyna preamps, however, operate well at 200 v.

Alignment of the Speakers

In order to phase the three speakers properly, the following procedure was adopted. First a signal at the crossover frequency of 150 cps was fed into the woofer (on the left) and the right speaker, and the phase of the latter was adjusted to the position in which the signal sounded louder. Then the left and right speakers were brought into phase by using the white noise signal on the Audio Fidelity Test Record. White noise, I find, is the most unambiguous method to check phasing. The gain of the bass channel was then adjusted to get smooth response with the low-frequency glide tone on the *Popular Science* Test Record No. 1.

The Speakers

The other components in the system and the reasons for their choice will now be described, starting at the back end—the speakers. The cabinet for the woofer is a Karlson, built from ¾-in. plywood, veneered with walnut, and finished with boiled linseed oil (*Fig.* 4). The joints were both glued and screwed (using a total of 130 screws), and weatherstripping was used to provide an air tight seal on the back. The inside surfaces were shellacked, with Fiberglas damping material on the two recommended surfaces. The Karlson has a reputation for making a cheap speaker sound good, and indeed it sounded fine with my old $20 15-in. woofer in it. In one splurge, however, I acquired an Altec 803B, the least expensive of the first-rate woofers, and now the speaker is too good for the cabinet. This speaker resonates at 25 cps in free air, but its output at the lowest frequencies is limited by the cabinet, and, to some extent, by the size of the room. The Karlson also has a peak from 70 to 90 cps; fortunately, this peak is rather broad, presumably because of the exponential slot. Some day I may get around to mounting the 803B in an exponential horn or an infinite baffle, although I would hate to part with my first veneering job, which turned out rather well.

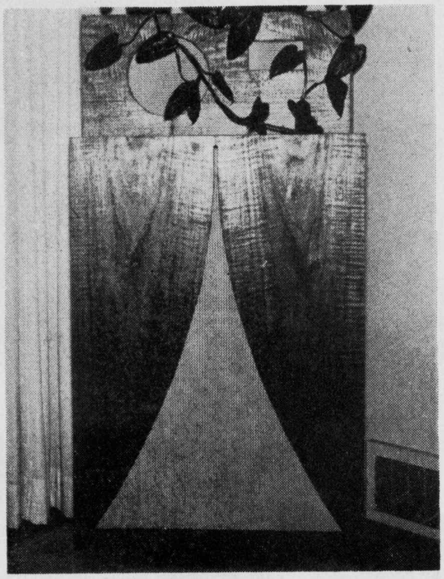

Fig. 4. The woofer and left speaker.

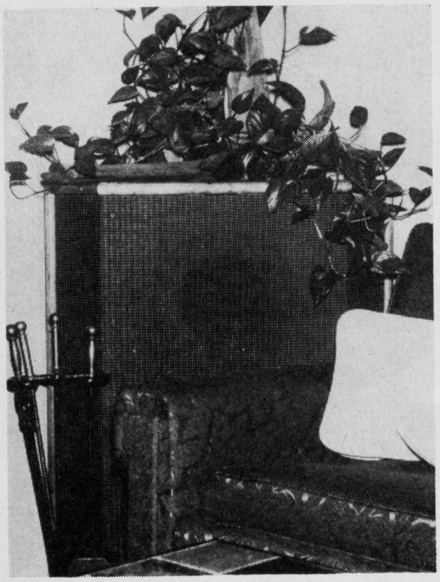

Fig. 5. The right speaker.

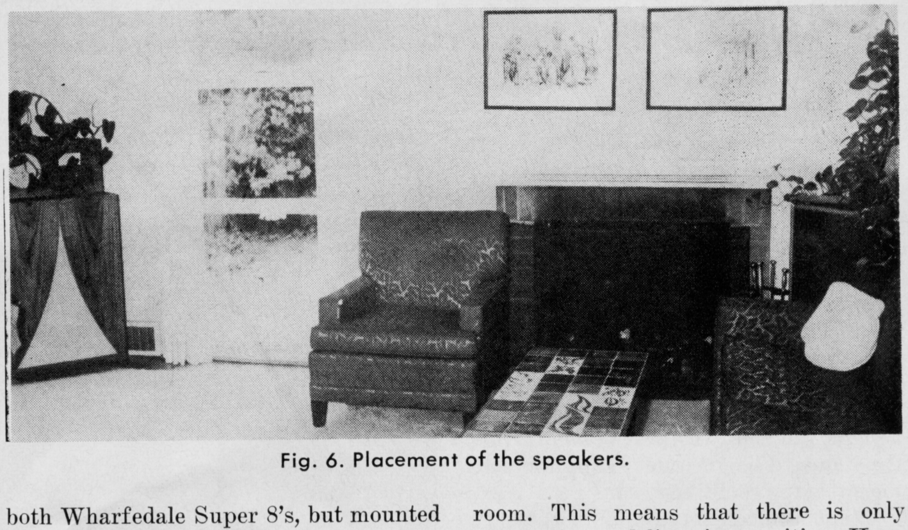

Fig. 6. Placement of the speakers.

The midrange-tweeter speakers are both Wharfedale Super 8's, but mounted differently. The left unit is housed in a small matching cabinet on top of the Karlson, as shown in *Fig.* 4. The back of the cabinet is open, covered only with a grill cloth to keep out the dust. This arrangement was my solution in the monophonic days to the problem of wanting to mount the midrange unit open-backed, but not having provision for it within the Karlson. The fact that the two cabinets are physically separate also allows for flexibility in arrangement of the furniture. The small rectangular opening in the open-back cabinet is not a port but a mounting hole for a University 4401 tweeter. This was originally added to the Super 8 with a crossover at 8000 cps, but it has since been disconnected since the Super 8 was found to need no help at all at the highest frequencies, and the tweeter merely served to unbalance the left and right channels.

The right speaker, as shown in *Fig.* 5, is mounted in an old corner bass reflex cabinet originally built for a cheap 12-in. speaker. Fortunately, the resonance of the Super 8 (a little below 70 cps), is close to that of the original 12-in. speaker, and no retuning of the port was necessary. Moreover, the Super 8 is not being used as a full-range speaker anyway. There was some worry that the difference in cabinetry for the left and right speakers, necessitated by considerations of decor, would unbalance the two channels. However, this did not turn out to be the case. The relative positions of the speakers are shown in *Fig.* 6. The chair between them is of course not the one used for listening, but the balance is so good that even from this chair one can hear a soloist apparently standing in the middle of the opposite wall and staying there.

You have no doubt noted the extreme separation of the speakers—some 13 feet—and the fact that they are "beamed" toward the center of the room. This means that there is only one really good listening position. However, this has not turned out to be a great disadvantage, since I have found that all serious listening has to be done alone anyway. When there is a crowd,

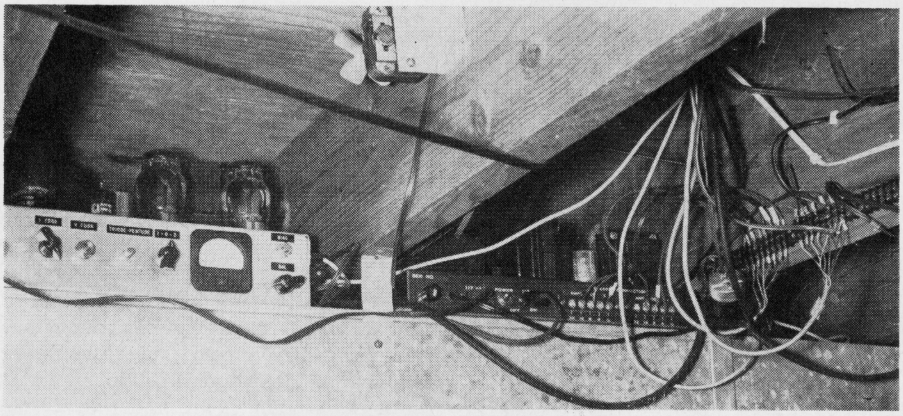

Fig. 7. The amplifiers.

there is always conversation. In spite of the speaker separation, there is absolutely no "hole-in-the-middle." On monophonic sources, the sound comes out of the middle of the wall between the two speakers. If a program has been recorded with too much separation, a turn of the blend switch fills in the hole to any degree desired.

The Super 8's seem to have presence peaks around 2000 and 5000 cps, which are accentuated by the sensitivity of the ear in this frequency range. A broadband RLC filter, constructed with a hand-wound choke, and centered around 4000 cps, was inserted in the speaker circuit to attenuate these peaks and provide a smoother apparent frequency response curve. However, most music did not sound as good with this filter in place as without, mainly because solo instruments would sound muffled and far away. With the filter in, the sound was more nearly like that from "colorless" speaker systems such as the acoustic Research series, but on A-B comparison I almost always prefer the Super 8's as they come. The filtering action of my wife's plants fortunately seems to have a negligible effect.

Amplifiers

The amplifiers, shown in *Fig.* 7, are located on top of a heating duct in the basement. Also visible are the connecting cables going through a hole in the floor to the living room, a patch panel for distributing speaker leads to different parts of the house, and a fan for cooling the output tubes of the bass amplifier. The emphasis, it should be quite apparent, has been on accessibility rather than neatness. The amplifiers are actually so close to the preamps that the standard length cables supplied with the preamps could be used.

The homemade bass amplifier, which also supplies all the preamp and crossover power, employs a modified Dynakit circuit. An ammeter has been added to check the current in the output tubes, the bias and balance being independently adjustable. Controls have also been added for independent adjustment of current and voltage feedback, and for changing from ultralinear to triode operation. These switches have since been found to be unnecessary, the damping being already optimal in the original design.

The treble amplifier is an Eico HF-86 dual 14-watter. This amount of power is entirely adequate for the efficient Super 8's, particularly since the amplifier does not need to supply any bass. With the wife and children safely out of the house, it is possible to turn the volume up to almost the threshold of pain without any sign of distortion.

The bass amplifier will deliver up to 50 watts. Since it is used only for the region below 150 cps, this is more power than is necessary for ordinary program material. However, if there are power peaks in the program sufficiently large to produce distortion, these peaks will occur in the bass, simply because such a large peak in the midrange would be painfully loud. Moreover, running the bass amplifier way below its power rating would practically eliminate harmonic distortion at the lowest frequencies.

Preamps and Control Circuits

The preamps are Dynakit PAM-1's and a DSC-1 stereo control. These were chosen for their versatility, desirable combination and arrangement of controls, and well thought out and sophisticated circuitry, as well as for the distortionless and humless reproduction they are known for. I have had only two complaints with the Dynakits: first, the master volume control had to be changed several times (at Dyna's expense) before one was found that tracked reasonably accurately; second, there is no provision for having both a tape head and a second RIAA input. I believe the latter has been fixed in the PAS-2, which came out after I had bought my preamps. The PAS-2 has several advantages over the PAM-1 plus DSC-1 combination, particularly in cost, but does not have quite the versatility. I still think the volume control should have been changed to a stepped one, even if only 20 per cent resistors are used.

The control panel is shown in *Fig.* 8. Under the Dynakits are two homemade chassis with etched brass panels and knobs to match the preamps. The unit on the left contains the speaker controls, phase reversal switches for the three speakers, a switch for inserting the presence filter mentioned before, and a switch for connecting the left amplifier to the remote outlets, to either an 8-ohm or a 4-ohm extension speaker or to both an 8-ohm speaker and the normal left speaker. The knob marked "VU meter" will be explained later. The unit on the right, under the end of the preamps which are insensitive to hum, contains three relays. Power to the entire system is turned on through a holding relay. This relay can be released, thus turning everything off, either manually or automatically at the end of a tape or record, as selected by the center switch. There is also provision for plugging in a timer to turn the system on and off, for recording radio programs *in absentia*. One of the a.c. switches on the preamps may be used to turn off the amplifiers alone; the other, for turning off the program sources alone.

The Program Sources

The preamps and program sources

Fig. 8. The control units.

are shown in *Fig.* 9. The cabinet was a floor sample picked up for $10, to which lucite doors and my wife's artistic design have been added. The electronic crossover can be seen under the cabinet, together with the seemingly unavoidable mess of cables. Records are stored in the lower part of the cabinet, and tapes on the small shelf in the adjoining bookcase on the right. The tuner is a Bogen R660, the only component left from the first incarnation of this system and the first one due to be replaced, although it still works quite well.

The enclosures in the cabinet are only about 13-in. × 17-in. × 15-in. deep and originally housed only a Miracord XS-200 changer with a GE GC-7 cartridge. It was obvious that no ordinary turntable or tape recorder could fit in such a space, and that expanding to a larger cabinet would raise howls from you-know-who. Fortunately, there are two components of high quality which do not take up any more room than necessary. They are also very reasonably priced.

Fig. 9. The home-decorated equipment cabinet.

The turntable is a Weathers KL-1 kit, mounted on an aluminum plate suspended above a wooden base. The cartridge and arm chosen is the B & O TA-12 combination. In the turntable base are mounted an hour-counter to keep track of the stylus wear, and a Realistic dual VU meter. The latter is connected across the 32-ohm taps of the Eico dual amplifier to give a little more gain in monitoring the signal actually arriving at the speakers. By means of a switch on the speaker control panel, the left meter can also be put across the woofer or an external speaker. The turntable is lighted by a 6-watt fluorescent lamp at the top front of the cabinet. The light also serves as a strobe lamp. To reduce hum, the ballast and starter for the lamp are located in the basement. The lamp is switched by one of the unused loudness switches on the PAM-1's.

The tape deck is a Viking Stereo Compact RMQ quarter-track machine with two built-in recording amplifiers. This and the Tandberg 6 were the two high-calibre decks which would fit in my cabinet, and, unfortunately, the difference in price was a factor of 2. The Viking is an excellent deck, with a .00009-in.-gap playback head and a separate wide-gap record head, making slow-speed quarter-track recording a reality. I find that with sufficient treble boost, the loss in quality at 3¾ ips is quite acceptable for most music, except when there are high-pitched percussior instruments, and I now do most of the recording off the radio at 3¾ ips, getting 6 hours of music on a single 1800-ft. reel. The Viking is made for hobbyists like myself, and between the Viking and the Dyna preamps, versatility is virtually unlimited. At the moment I do not have playback preamps on the tape deck, and the playback heads are connected directly to the "special" input of the Dynakits, so that the tape-monitor switch cannot be used. However, because of the great flexibility, I can still monitor monophonic recordings by using one preamp for the program source and the other for tape playback, and using the channel reverse switch as a monitor switch. The VU meters on the recording amplifiers are very useful; however, they do not light up, and I had to add pilot lights to show when the amplifiers are on.

I now tape all of my new stereo records, using a slightly greater than normal stylus force, and play the records only on special occasions. Aside from reducing record wear, this practice also eliminates the necessity of meticulously dusting the record each time and of changing records every 20 minutes.

After writing this article, I got up the courage to add up the cost of this system. I came to the conclusion that, exclusive of the tape deck, it can be reproduced for less than $500, plus an awful lot of work. For $550 to $600, one can probably buy a "standard" system of similar quality, but without the versatility and luxury features of this system. Although I would hesitate to recommend this common-bass speaker system to the average music listener, I think it deserves consideration by audio hobbyists. The common-bass speaker system is particularly useful when the woofer can be mounted in a wall, using a closet, a garage, or another room as an infinite baffle. The location of the woofer in the room would be immaterial because of the low crossover frequency, and the Super 8's would require very little room. If the Super 8's were properly baffled as full-range units[4], the crossover could be reduced to below 100 cps; then our assumptions of in-phase addition of acoustic signals and of non-directionality of the bass would hold much more accurately. The common-bass system would also be useful to those who have "full-range" speaker systems which have insufficient output below 60 cps, and who wish to add a single woofer to supplement the extreme bass. Æ

PARTS LIST

$R_1, R_2, R_5, R_6, R_{13}$	1 megohm
R_3, R_4	2 megohms
R_7, R_8, R_{14}	2200 ohms
R_9, R_{10}, R_{15}	47,000 ohms, 1 watt
R_{11}, R_{12}	20,000 ohms
R_{16}	10,000 ohms
R_{17}	470,000 ohms
R_{18}	18,000 ohms, 1 watt
R_{19}	1000 ohms
R_{20}	8200 ohms, 1 watt
C_1, C_2	500 pf
$C_3, C_4, C_8, C_9, C_{10}$	0.1 μf
C_5, C_6	0.05 μf
C_7	0.002 μf
V_1, V_2	12AU7
SW_1	6-pole, 3-position

[4] J. L. Grauer, "#4, 80 Pounds, a Super 8, and the Shim Method," AUDIO, Jan. 1961.