Low-cost High-quality Loudspeaker

Design for frequencies above 100Hz. 1-Construction and assembly

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This loudspeaker may be built by the home constructor for a total expenditure in the region of £6. While it does not have the extended bass response of some much more expensive loudspeakers, it is nevertheless unusually free from the colourations and hangover effects which are unfortunately still a feature of the majority of commercial designs[1]. Consequently, on many types of programme material, it will be found to give considerably better reproduction than is obtained with many much more costly loudspeakers.

On speech, both male and female, the loudspeaker reaches a very high standard of performance. Using a high-grade capacitor microphone out of doors, an almost deceptive degree of realism can be achieved in the reproduction of familiar voices.

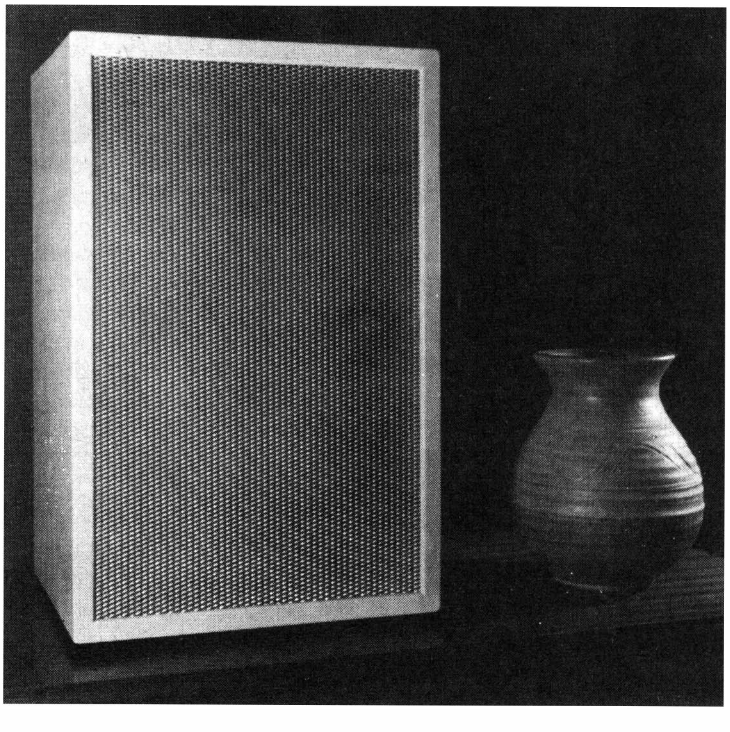
While many people who have heard the loudspeaker on music seem to find the bass response fairly adequate, direct comparison, particularly on organ music or large-scale orchestral music, with a good loudspeaker, such as a B.B.C. monitor[1], leaves listeners in no doubt that the reduced response below about 100 Hz constitutes the main shortcoming of the design*. Consequently it is recommended that, where space permits, the basic low-cost loudspeaker should be augmented, at frequencies below about 100 Hz, by a separate woofer. Because this has to cover a frequency range of only about one octave, in a rather uncritical part of the spectrum, there is much latitude in its choice and almost any old 12-in. unit, such as can be bought second-hand for a pound or two, can be pressed into service.

In a stereo system, one such woofer can be shared very satisfactorily between the two channels, and because these very low frequencies convey almost no sense of position, the woofer can be placed in any convenient position in the room. When circumstances permit, the possibility of mounting the woofer unit in a hole in the floor, ceiling or a wall may be worth considering, as it saves the space and labour of cabinetwork.

Suitable circuit arrangements for such a three-speaker stereo system will be discussed in Part 2 of this article, and it will be shown how the relative levels from the woofer and the other two loudspeakers may be adjusted to give nicely balanced reproduction in listening rooms having different acoustical properties.

A complete stereo system on the above lines can thus be built for no more than about £15, and is capable of a surprisingly high standard of reproduction. Even direct comparison with a pair of Quad electrostatic loudspeakers does not always reveal any obvious shortcomings, though careful listening over a period of time makes it evident, in particular, that the lower intermodulation and hangover distortion of the electrostatic speakers results in greater clarity and separation of instruments particularly at high volume levels. Nevertheless, the low-cost system is capable of quite impressive volume and clarity in the reproduction of orchestral and choral music in rooms of normal living room size, and in much music of a quieter nature listeners have shown no marked preference for one or the other speaker system.

The complete low-cost loudspeaker. The size of the cabinet is 18in. × 12in. × 10in.(deep).



Evolution of the design

The following thoughts were significant in the evolution of the present design, which aims to satisfy an evident demand for the best possible quality of reproduction at a really low price:

(a) Large loudspeaker units suitable for a wide frequency range are expensive and need augmenting by a tweeter for really first-class results.

(b) Smaller circular units, e.g. 8 in., often suffer from undesirable hangover effects in the lower-middle-frequency range [1,4,5] and the unpleasant sound of these cannot be fully removed by electrical equalization. However, it was mentioned by Dr. G. F. Dutton of E.M.I. at the discussion following Mr. Shorter's paper[1] that the use of elliptical rather than circular diaphragms gives a marked reduction in hangover distortion, which is caused by diaphragm vibration persisting in low-damped radial modes after the cessation of the signal.

(c) The surprisingly good results given by a commercial loudspeaker known as the 'CQ Reproducer', which used a cheap

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* Of course, some small-scale music of great beauty contains almost no frequencies below 100 Hz and is therefore virtually unaffected by this shortcoming.

elliptical unit almost the same as that employed in the present design, served further to direct the author's attention to the virtues of elliptical diaphragms, and preliminary measurements on such a unit showed that it had an axial frequency response which, if its main departures from levelness were to be corrected by a cheap and simple electrical equalizer, would give a sufficiently uniform and wide-range response to meet the requirements of very high quality reproduction—except that some sacrifice of performance at very low frequencies seemed virtually unavoidable.

(d) The use of a single unit to cover the whole frequency range also simplifies matters by avoiding the problem of the unnatural changes in polar response which are liable to occur in the crossover regions of multiple-unit systems[1].

(e) While the exploitation of cabinet panel resonances to modify the frequency response over certain ranges is a dodge which has sometimes been employed with a degree of success in cheap designs, it was felt to be such a tricky and unpredictable technique that it would probably be much better avoided.

(f) The notion that very high flux densities are essential for good transient response, while a widely propagated belief, is not in accordance with much practical experience†.

Consequently the fact that the cheap elliptical unit being considered had a rather small magnet was not regarded as of much significance in this context.

(g) Of much greater significance was felt to be the fact that only quite small diaphragm excursions, in the region of ±1 mm, can be made without running into considerable suspension non-linearity and non-linearity caused by the rather skimpy nature of the coil and magnet geometry. Indeed it is still a source of some surprise that such substantial volume can be obtained in practice without these non-linear effects giving any obvious subjective impairment of the reproduction.

The basic recipe adopted thus involves no more than the use, in association with a simple electrical equalizer, of a particularly suitable, though quite cheap, elliptical unit having a plasticized surround, mounted in a totally enclosed box made rigid by internal bracing and containing felt damping material to reduce standing-wave effects and provide some additional damping of the main diaphragm resonance.

The size of the box is such that the stiffness of the enclosed air at low frequencies, referred to the diaphragm is about equal to the mechanical stiffness of the diaphragm suspension, resulting in a resonant frequency of about 100 Hz. This size of box is quite convenient to accommodate, and the improvement in bass performance given by even quite a large increase in volume would not be great. Moreover, the greater the overall stiffness, the less will be the intermodulation distortion when strong low-frequency signals are fed to the unit, e.g. at 40 Hz, at the same time as higher frequencies. The size of box adopted is thus thought to be a good all-round compromise.

While the use of a vented enclosure has been carefully considered, such an arrangement would either result in a considerable increase in intermodulation distortion in the presence of large inputs at very low frequencies, or, if the Helmholtz resonant frequency were made low enough to avoid this danger, the response at very low frequencies would be at a lower level than that at higher frequencies, requiring further electrical equalization. For normal circumstances, the simple totally enclosed box seemed to be the best choice, therefore.

The use of an equalizer of fixed design, not adjusted to suit individual loudspeaker units, will obviously be satisfactory only if the variations in response of the units in production is sufficiently small. All that can be said is that of several units checked, bought over a period of several years, all have had the same main features in their frequency responses, fairly closely matched by the equalizer characteristic. Thus, while it cannot be guaranteed that every loudspeaker built to the present design will have quite such a good frequency response as the prototype, it seems virtually certain that the equalizer will always effect a marked improvement in the results.

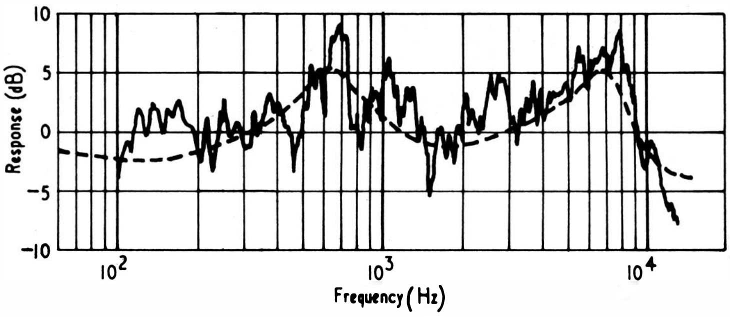

Fig. 1. Full-line: unequalized axial frequency response of loudspeaker. Broken line: inverse of equalizer frequency response.

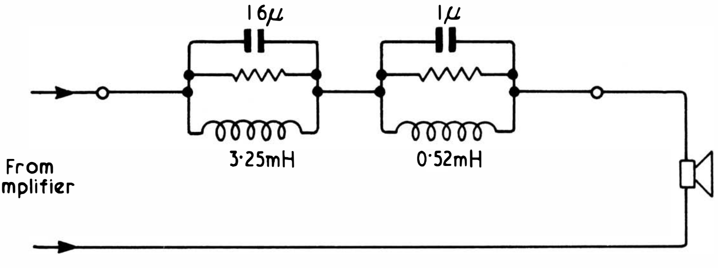

Fig. 2. Basic equalizer circuit.

Frequency response curves

The full-line curve in Fig. 1 shows the measured axial frequency response of the loudspeaker without the equalizer ‡. It will be seen that, ignoring the numerous small wiggles (which appear in virtually all loudspeaker response curves if the frequncy is varied slowly enough), the main features of this curve are a region of excessive output centred broadly just below 700 Hz, and another similar region centred at about 7 kHz.

The basic equalizer circuit designed to correct the Fig. 1 response is shown in Fig. 2. However, because 16 μF is an inconveniently large capacitance, the practical equalizer circuit is arranged as in Fig. 3. The full-line curve in Fig. 4 shows how the equalizer causes the voltage across the speech coil to vary with frequency for a constant amplifier output voltage. Referring to Fig. 1 again, the broken-line curve is an inverted version of the full-line curve in Fig. 4, and shows that the equalizer characteristic is quite well matched to the main features of the loudspeaker response. (The broken-line curve in Fig. 4 simply shows the effect of removing the damping resistors from the equalizer circuit.)

Fig. 5 shows the overall axial response curve of the loudspeaker with the equalizer incorporated and it will be seen that most of this lies within ±3 dB limits from 100 Hz to over 10kHz,

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† A weak magnet may give rise to a peak in the frequency response in the region of the main resonant frequency of the diaphragm[2]. While this is not necessarily undesirable if it occurs well below 100 Hz, where some degree of ringing does not seem to give subjectively noticeable impairment of transient response, it can in any case be damped down by acoustic means, e.g. by a close-fitting felt cover over the loudspeaker unit, if this is thought desirable. At higher frequencies, many of the diaphragm resonances are so weakly coupled to the coil that little electromagnetic damping can occur even if the flux density is very high.

‡ This measurement was made out of doors using a small home-made omnidirectional capacitor microphone at a distance of 2 ft 6 in. from the front of the loudspeaker and at a height of 4 ft above ground, on axis. The microphone was used in the r.f. bridge system described in Reference 6, and its pressure calibration was obtained by developing a constant alternating force on the diaphragm by means of an oscillator voltage applied in series with a d.c. polarizing voltage. To avoid any significant error at high frequencies due to pressure doubling, the capsule was then placed in front of the loudspeaker with its ½ in. diameter diaphragm in a horizontal plane.

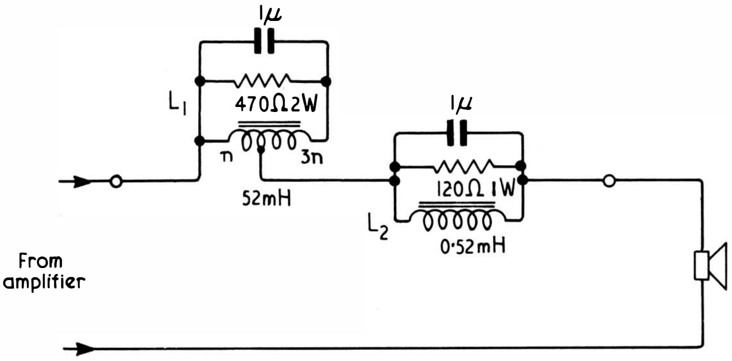


Fig. 3. Practical equalizer circuit.

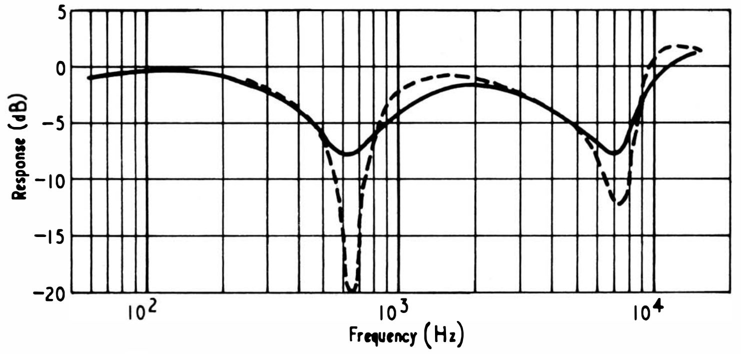


Fig. 4. Measured frequency response from amplifier output to speech-coil. Full line: with damping resistors as in Fig. 3. Broken line: without damping resistors.

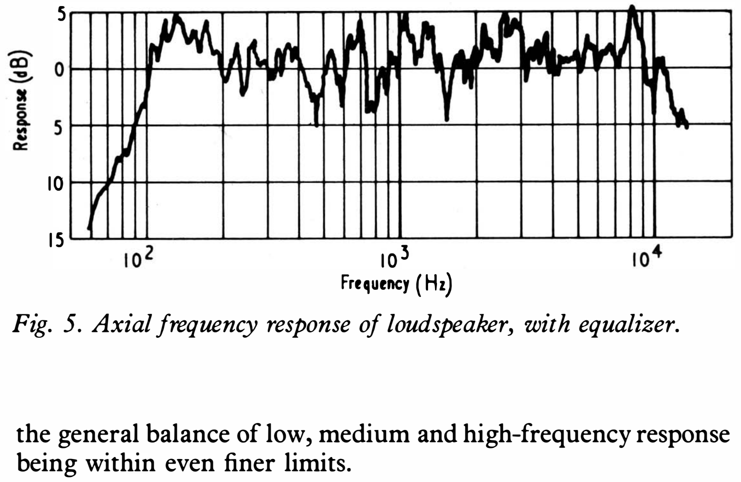


Fig. 5. Axial frequency response of loudspeaker, with equalizer.

the general balance of low, medium and high-frequency response being within even finer limits.

The loudspeaker unit. The unit is an Elac 15-ohm 9in × 5in elliptical unit, Type 59 RM/109, manufactured by Electro Acoustic Industries Ltd., Stamford works, Broad Lane, Tottenham, London N.15. The retail price is £2 6s 2d. When ordering state that 15-ohm impedance is required.

Constructing the equalizer

The equalizer circuit has already been given in Fig. 3. The inductors both employ 0.014 in. silicon iron laminations, Inter-Service No. 421. These are conveniently obtainable in kits from The Belclere Company Ltd., 385/387, Cowley Road, Oxford. Each kit consists of a stack of Silcor 107 laminations, a bobbin and a steel shroud. For each equalizer, two kits are required:–

Kit GN/Silcor ($\frac{7}{16}$ in. stack) Price 6s 6d
Kit GX/Silcor ($\frac{3}{4}$ in. stack) Price 7s 0d

It is essential to specify 'T' and 'U' laminations when ordering, as the firm now normally supplies 'E' and 'I' types, but has agreed to supply 'T' and 'U' laminations for this equalizer when requested to do so.

L_1 winding. The tapped inductor L_1 uses the larger GX size core stack. First wind on 110 turns of 28 s.w.g. enamelled copper wire in four neat layers. Then wind on, *in the same direction,* 330 turns of 34 s.w.g. enam., making 440 turns for the whole bobbin. The 330 turns need not be wound in accurate layers—just wound on reasonably tidily. There is no need for any insulation between sections, but the outside of the winding should preferably be protected by empire cloth or thick paper.

L_1 core. Place all the T's through the bobbin tunnel from one side. Place all the U's in the shroud, with small pieces of cardboard, or $\frac{1}{16}$ in. s.r.b.p. ('Paxolin') $\frac{1}{4}$ in. × $\frac{3}{4}$ in. as shown in Fig. 6, to prevent the steel shroud coming too close to the core gaps.

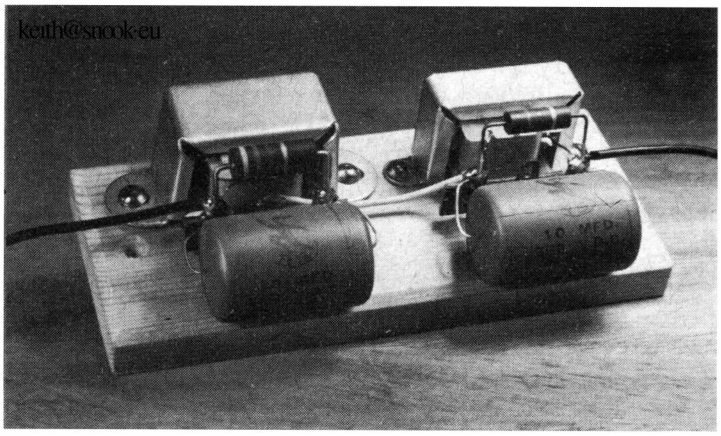
Each of the three gaps should be 0.025 in., which must, of course, be formed by inserting suitable insulating material. In the absence of other facilities, use may be made of the fact that the outside cover of *Wireless World* has been made of paper of thickness approximating closely to 0.005 in. for at least ten years! Thus insert five thicknesses of this paper in each gap. It will be found convenient to cut strips of widths approximately $\frac{1}{8}$ in. and $\frac{5}{8}$ in. for the outside and central gaps respectively and to fold these strips in zig-zag fashion to form five thicknesses. With these gap-spacers in position, the shroud should be screwed down tightly onto the wooden baseboard, $\frac{3}{8}$ in. No. 6 roundhead woodscrews being suitable.

L_1 connections. The enamelled wires from the bobbin should be carefully bared with sandpaper or emery paper and soldered to the three tags of a tagstrip screwed down to the baseboard as shown in Fig. 6 and the photograph below. The beginning of the winding (inner end of the 28 s.w.g. section) should go to the tag nearest to the end of the baseboard, the outer end of the 28 s.w.g. section and the inner end of the 34 s.w.g. section going to the middle tag to form the tapping point. The outer end of the 34 s.w.g. section goes to the tag nearest the middle of the baseboard.

L_2 winding. The untapped inductor uses the smaller, size GN, core stack. Wind on 86 turns of 24 s.w.g. enam. in four neat layers and cover with empire cloth or paper. (For winding this and the other inductor, a simple gadget may be improvised, using bits and pieces from the junk box, Meccano, etc., for rotating the bobbin. There is no need for a turns counter—the number of turns is small enough to be counted without difficulty mentally!) If an aluminium or other non-ferrous shroud is used instead of the sheet steel one employed in the prototype (supplied by Belclere), the winding turns should be increased to 95. Also, because of the increased shunt loss resistance then obtained, the damping resistor value (Fig. 3.) should be reduced from 120 ohms to 68 ohms. The use of an aluminium shroud for inductor L_1 has no significant effect on the inductance value or losses, owing to the much smaller air gap.

L_2 core. Insert all the T's through the bobbin tunnel from one side and place the shroud over the core so that the tops

The equalizer.



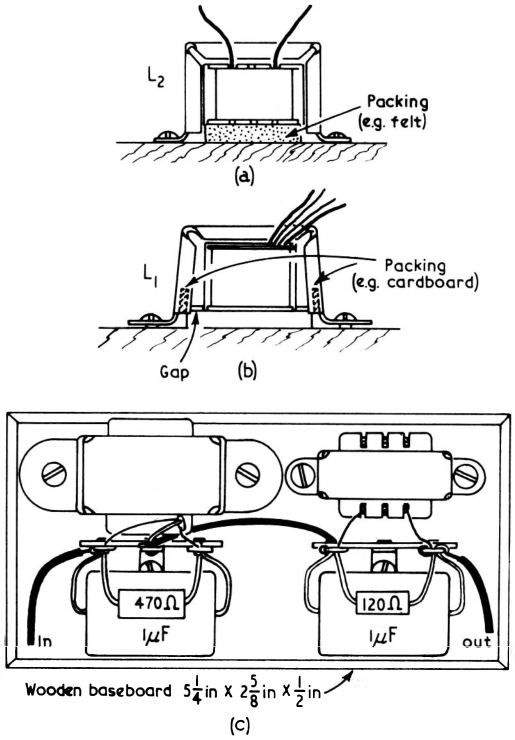


Fig. 6. Constructional detials of equalizer.

of the T's lie inside the top of the shroud. A piece of $\frac{3}{16}$ in. thick soft packing material measuring 1 in. × $\frac{3}{4}$ in. should now be obtained. This is placed between the bobbin and the wooden baseboard so that the bobbin and laminations are pressed securely up into the shroud when the latter is screwed down.

No U laminations are employed for this inductor.

Wire supply. The three gauges of enamelled copper wire required (24, 28 and 34 s.w.g.) may be conveniently obtained, in a minimum quantity of 2 oz each, from Post Radio Supplies, 33, Bourne Gardens, London, E.4.

The three 2-oz reels contain enough wire for at least four equalizers.

Other components. The other components required for the equalizer are all readily obtainable, including tag strips, from Radiospares Ltd. through any radio dealer. Tubular 1 μF paper capacitors, 250V d.c. wkg., ±20% tolerance, are suitable.

Tests. Provided the above instructions have been carefully carried out, it is virtually certain that the equalizer will function correctly. However, if an oscillator is available, it is worth while to check that, with a constant voltage fed to the series combination of equalizer and loudspeaker, the voltage across the loudspeaker varies with frequency approximately as shown in Fig. 4.

The exact position of the lower-frequency dip is slightly dependent on the a.c. voltage at which it is determined. With a source voltage of 2V r.m.s., the dip will occur about 20 Hz lower in frequency than with a source voltage of 0.2V r.m.s. For voltage levels above 2V r.m.s., the fall-off in dip frequency with increasing level is more gradual. (This effect is due to the fact that the initial a.c. permeability of silicon iron is rather low compared with its value at higher flux densities; the effect is well diluted by the presence of gaps, however, and does not seem to give rise to any subjectively noticeable distortion.)

The performance of the equalizer may be regarded as satisfactory provided the measured results fall within the following limits:

(a) Low-frequency dip, with a test voltage of about 2V r.m.s. applied to the combination of equalizer and loudspeaker, 580 to 800 Hz.

(b) High-frequency dip (almost independent of test voltage) 6200 to 7900 Hz.

(c) Magnitude of dips (almost independent of test voltage), relative to response at 1700 Hz, −5 to −8 dB.

In the unlikely event of the performance falling outside any of the above limits, adjustments may be made as follows:

To correct (a), adjust gap of L_1. Increasing the spacer thickness by 0.005 in. will raise dip frequency about 40 Hz.

To correct (b), the value of C_2 may be modified or, alternatively, the number of turns on L_2 may be adjusted. Removing 5 turns will raise the dip frequency about 400 Hz.

To correct (c), alter the appropriate damping resistor value.

Equalizer intermodulation distortion. Variation in inductance of L_1 and L_2 with the instantaneous value of large low-frequency signal currents flowing through them is a possible cause of intermodulation distortion. A test showed, however, that the inductance of L_1 dropped by less than 2% when a direct current of 0.25A was passed through the whole winding, equivalent to a current of 1A through that part of the winding traversed by low-frequency signal currents. The effect in L_2, because of the much larger gap, will be even smaller. It is obvious, without further calculation, that the distortion caused will be considerably smaller than that introduced by non-linearities in the loudspeaker unit itself.

Alternative equalizer design. While the above method of constructing the equalizer is attractively cheap, some readers may find it more convenient to use Mullard Vinkors. Brief winding details are:

L_1: 99 turns of 28 s.w.g. plus 297 turns of 34 s.w.g. on Mullard 35mm, μ_e = 63, Vinkor. (LA2102 core and slug, DT2180 bobbin, plus DT2151 or DT2187 casing or DT2234 mounting clip.)

L_2: 49 turns of 24 s.w.g. on Mullard 25mm, μ_e = 63, Vinkor. (LA2302 core and slug, DT2179 bobbin, plus DT2149 or DT2185 casing or DT2228 mounting clip.)

The resistor R_2 across L_2 should be 68 ohms as compared with 120 ohms in the Belclere version—this allows for the lower losses of the Vinkor. R_1 remains at 470 ohms.

Equalizer kit. A complete kit of parts for the equalizer, including ready-wound inductors on laminated cores, may be obtained from Peak Sound (Harrow) Ltd., 32 St. Judes Road, Englefield Green, Egham, Surrey. The price is £1 16s. The author understands that this company will also supply a kit of parts for the complete loudspeaker (including cabinet parts).

Constructing the cabinet

While some readers may prefer to buy a cabinet and adapt it to the present design, there must be many others who, like the author, find woodwork an enjoyable and rewarding pastime and would prefer to make their own. For this reason full details and a few constructional hints and tips are given.

The author used $\frac{1}{2}$-in. gaboon plywood, but veneered chipboard ('Weyroc') is a satisfactory alternative. While minor changes to the dimensions shown in Fig. 7 may be made to suit individual requirements, the volume of the cabinet should preferably be kept about as shown.

The cabinet is held together by a combination of Cascamite glue and $\frac{7}{8}$ in. countersunk woodscrews, screwed into the inside of the cabinet sides through the $\frac{1}{2}$-in. × $\frac{1}{2}$ in. wooden strips shown. The local timber yard supplied the latter, and also the 1-in. × $\frac{1}{2}$-in. strip for fixing the speaker mounting board, in ramin, a beautifully straight-grained, but quite cheap, hardwood ideally suited to the purpose.

The bottom of the cabinet should be fixed with screws only (no glue) so that it can be removed easily for extracting the speaker mounting board, should this become necessary at any time for renewal of the expanded aluminium or Tygan covering—see photograph on page 246.

The use of mitred corner joints at the top corners of the cabinet allows the veneer to extend right up to the corner, and unlike some other corner joints [7] can be cut by the amateur without

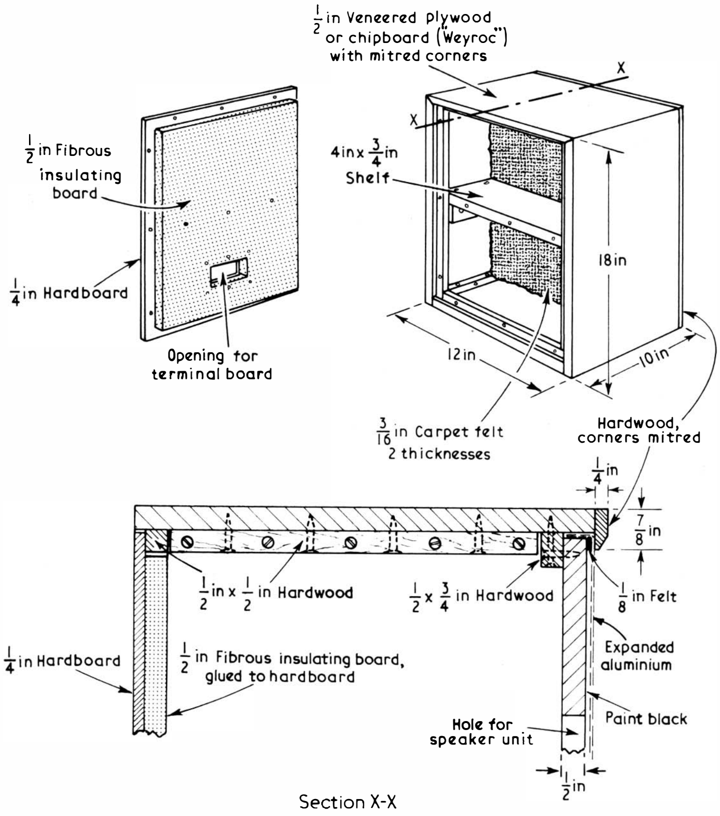


Fig. 7. Cabinet construction details.

special rabbet-cutting tools. The author had not previously tried cutting mitred joints for box corners, but found it surprisingly easy to produce a thoroughly neat job.

The aim should be to cut the wood *very* slightly off 45°, so that when the screws are tightened into the $\frac{1}{2}$ in. × $\frac{1}{2}$ in. strip, the mitred joint is sure to close tightly at the outside of the box.

Either Tygan or expanded aluminium may be used for the front of the loudspeaker, according to choice[7]. While expanded aluminium causes slightly less acoustic obstruction, many people prefer the appearance of Tygan. Expanded aluminium may be obtained from The Expanded Metal Company Ltd., P.O. Box 14, Stranton Works, West Hartlepool, Co. Durham.

Components of cabinet, showing removable base, and the elliptical loudspeaker unit.

A suitable type is List Ref. No. 363A in plain aluminium (22 s.w.g.). It is made in standard sheets 4 ft × 2 ft, in which the 'long way of mesh' (which must be horizontal when mounted in the loudspeaker, for satisfactory appearance) runs along the 4-ft dimension. Half a sheet, 2 ft × 2 ft is therefore a sensible quantity to order for two loudspeaker cabinets. A piece 2 ft × 6 in. will be left over, but will probably come in useful sooner or later.

Tygan may be obtained from A. C. Farnell Ltd., 81, Kirkstall Road, Leeds 3. A book of samples may be obtained. The material is available in cut pieces 27 in. × 24 in. or at any length × the width of the roll, which is 54 in.

Two of the four cabinets made by the author are wax polished with expanded aluminium fronts. After very thorough sandpapering, finishing with No. 1 sandpaper, one thin coat of white French polish was put on quickly with a cloth rubber, followed by wax polishing with Meltonian white shoe polish. Nothing could be much easier and quicker than this finish, which is nevertheless very pleasing. The other two cabinets, however, are painted white and have pattern U528 Tygan.

If expanded aluminium is used, it is important that it should be fixed in such a way that it cannot rattle. The author glued a $\frac{1}{4}$ in. wide strip of $\frac{1}{8}$ in. thick felt round the periphery of the front surface of the speaker mounting board, thus separating the main area of the expanded aluminium from the board. To make doubly sure the aluminium would not vibrate against the board, $\frac{1}{2}$ in. squares of $\frac{1}{8}$ in. felt were stuck to the board at four positions round the outside of the speaker hole. This felt, the speaker mounting board and the edge of the cone-fixing cardboard on the speaker unit should be painted matt black, to prevent any of these being visible through the expanded aluminium. As previously mentioned, the 'little louvres' of the expanded aluminium should be horizontal rather than vertical; there is also a right way up to mount this material which gives minimum transparency from the usual viewing angles. Even the two sides of a piece of expanded aluminium will be found on careful inspection to be slightly different, and it is worth putting it the same way round if two cabinets are being built for stereo working.

When Tygan is used, the strips to space it from the speaker board need not be of soft material, and $\frac{1}{8}$ in. hardboard is suitable**. The Tygan may be stuck round the edges of the board with Evo-stik impact adhesive, care being taken to keep the warp and woof running parallel to the board edges. If this is not got quite right at first, it is possible to pull the Tygan off in the appropriate place and reposition it slightly—but every effort should be made to get it right first time nevertheless.

Finally, the heat treatment recommended by Mr. Briggs in Reference 7 should be applied—a bar-type electric radiator 'should be held about six inches away from the mesh for about five seconds, when the heat begins to contract the fibre. Remove the radiator *immediately* a slight movement is seen in the Tygan, otherwise excessive contraction will be induced'.

Before finally mounting the speaker unit, check that the coil leads are correctly positioned and in no danger of rattling against the diaphragm or the speaker chassis.

Wooden strips of cross-section 1-in. × $\frac{1}{2}$ in. should be screwed and glued edgewise on, using $1\frac{1}{4}$ in. No. 8 countersunk screws, to the insides of the cabinet sides, top and bottom at a distance from the front edges sufficient to accommodate the thickness of the the speaker mounting board after it has been fitted with expanded aluminium or Tygan. This distance is $\frac{3}{4}$ in. in the

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** It might be thought that with cloth there would be no need to space it away from the board. It is easily demonstrated, however, that if the cloth is touching the board but is not stuck to it, buzzing sounds are produced at certain low frequencies. The trouble with sticking the Tygan to the front of the board is that it is liable to make the outline of the speaker hole visible from the front.

author's cabinets, but it is as well to tailor it to suit the speaker boards as made. The aim is to make the latter an easy sliding fit between the 1 in. × $\frac{1}{2}$ in. strips and the inside of the front mitred mouldings when later fitted. The board is secured in place by screws into it through holes drilled in the 1 in. × $\frac{1}{2}$ in. strips.

The $\frac{1}{2}$ in. × $\frac{1}{2}$ in. strips shown may then be screwed and glued suitably in place and the carcase of the cabinet assembled. The bottom, however, should not be glued in position, so that it can be removed by undoing screws only, as previously mentioned. The strips should, of course, be spaced from the back of the cabinet by an appropriate amount, to accommodate the thickness of the back cover.

The cabinet back could be made of $\frac{1}{2}$ in. plywood or Weyroc like the sides, but the author used a slight modification of a B.B.C. recipe[1], $\frac{1}{4}$ in. hardboard being glued to $\frac{1}{2}$ in. builder's insulating board, as shown in the photograph on the right. This gives a composite board which is considerably lighter than wood of the same thickness and which also possesses desirable self-damping properties.

The choice of cross-section for the mitred front mouldings of the cabinet exerts a subtle effect on the appearance, and can be left to individual preference. The moulding should be glued to the carefully planed front edge of the cabinet, but a few 1 in. or $1\frac{1}{4}$ in. panel pins will make it much easier to position the mouldings nicely and ensure that they remain properly positioned while the glue sets. The panel pins should be punched down below the wood surface, and each hole filled with plastic wood made by mixing a drop or two of Durofix with plenty of wood dust obtained from sandpapering a nearby part of the same moulding. Allow to dry very thoroughly before finally sandpapering flush—the pin positions should then be almost invisible.

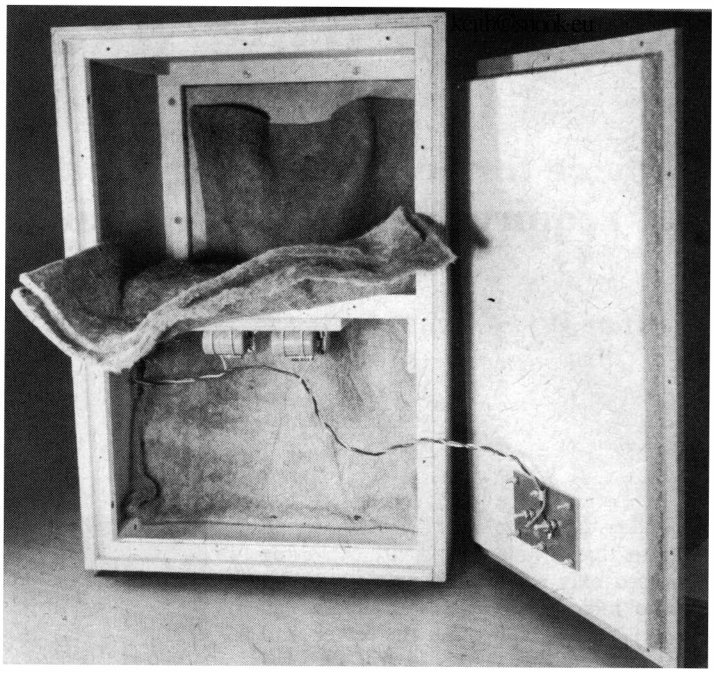
The moulding attached to the bottom of the cabinet should not be glued near its corners, otherwise the bottom will cease to be easily removable.

The $\frac{3}{4}$ in. thick 'shelf' provides a firm anchorage for both the sides and the back of the cabinet (the back being screwed to the edge of the shelf) and thus reduces the tendency for these parts to vibrate in 'drum' fashion.

There are obviously various possibilities for the signal connections. The author made a rectangular cut-out in the back of the cabinet, $2\frac{1}{2}$ in. × $1\frac{1}{4}$ in., and fitted behind it a $\frac{1}{8}$ in. s.r.b.p. ('Paxolin') board carrying two nickel plated 2 B.A. screw terminals—all available from Radiospares Ltd. through radio dealers.

It is important to keep the cabinet reasonably free from air leaks. One easily overlooked source of leak can arise when expanded aluminium is used, if it is bent right round the edges of the speaker mounting board and onto the back surface of the board. Even though the board is held tightly by screws against the 1 in. × $\frac{1}{2}$ in. strips fixed to the cabinet sides, there is nevertheless an air leak round the edges of the board through the interstices of the expanded aluminium. It was found experimentally that the diaphragm displacement at 40 Hz was reduced *several times* on sealing this leak, and there must, of course, be an accompanying reduction in intermodulation distortion. There is a good case, therefore, for cutting the expanded aluminium only $\frac{1}{2}$ in. larger than the speaker mounting board all round, and fixing it with tacks into the edge of the board, thus obviating the leak.

After finally assembling the cabinet, with the unit in place, about 30 in. of red-and-black flex should be soldered to the speaker tags, of which one is marked red by the makers. One piece of ordinary carpet felt, about $\frac{3}{16}$ in. thick and measuring 14 in. × 11 in.†† should now be tacked in place with six tacks spaced out round the unit, producing a sort of roughly fitting felt hat over the unit. This will provide considerable damping of the low-frequency resonance and will consequently reduce the acoustic output in the 100 Hz region. Without it, there may be a slight tendency towards colouration of male speech. With two thicknesses of felt tacked down more closely with a larger number of tacks, the bass response will be decidedly thin. If no woofer is to be used, some constructors may prefer the compromise of omitting this felt cover altogether—speech may then sound a little too full in the bass, but the musical reproduction may be thought better.



Interior of completed loudspeaker, with the two-layer felt 'curtain' partly removed to show the felt 'hat' over the drive unit. The removable back (right) is made from hardboard and insulating board.

After dealing with the above, a 'curtain' made from two pieces of approximately $\frac{3}{16}$ in. carpet felt, each measuring about 19 in. × 13 in., should be tacked loosely in place inside the cabinet as a sort of diaphragm dividing the space into two halves, with the loudspeaker unit in one half.

The equalizer should now be screwed to the shelf and wired in series with one of the leads from the unit to the terminal board. The terminal connected to the red lead should be marked appropriately if stereo operation is envisaged.

The back should be thoroughly screwed on, using three screws along each edge plus three more along its middle to fix it to the edge of the shelf.

Finally, four rubber feet, available from most hardware shops, may be screwed to the bottom of the loudspeaker—or a piece of felt may be stuck on if preferred.

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†† The pieces used by the author weighed approx. 1.6 oz.

Low-cost High-quality Loudspeaker

2. Using the loudspeaker: Determining the electro-mechanical constants of the drive unit

by P. J. Baxandall, B.Sc. (Eng.), F.I.E.E., F.I.E.R.E.

An amplifier capable of giving a sine-wave mean output power* of about 10 W into a 15-Ω resistance load is quite sufficient for operating the low-cost loudspeaker. The use of higher output levels will give gross distortion and may cause permanent damage to the loudspeaker.

Measurements on the loudspeaker (described later in the article) yield an efficiency figure of about 1%, so that a 10-W amplifier is capable of producing from the loudspeaker a sine-wave mean acoustic power output of about 100 mW. From reference 8, this would be expected to give, in an average living room of, say, 1,500 cu ft, a sound intensity nearly 100 dB up on the standard reference level of 2×10^{-5} newton/sq metre r.m.s. This is about the intensity experienced in a good seat in a concert hall during very loud climaxes of orchestral music, though many musical people choose to listen at considerably lower levels at home. (An independent check with a microphone of known sensitivity, at 1 metre on axis out of doors, also gave an intensity of about 100 dB up on 2×10^{-5} N/m^2 for full output from a 10-W amplifier.)

Comparison in the author's living room of the measured sound intensity of a grand piano, and of the reproduction of a recording of this piano via the low-cost loudspeaker, showed that the loudspeaker, when driven by a 10-W amplifier (not overloaded), could produce an intensity very nearly, but not quite, equal to that of the pinao itself when played at extreme fortissimo.

As mentioned earlier, the use of a woofer is really well worth while, adding depth and warmth to the reproduction, improving the naturalness of the balance and reducing listening fatigue.

In a mono system, the simple arrangement shown in Fig. 8 has been found very satisfactory, provided the woofer sensitivity is high enough. A suitable recipe for the choke is as follows:

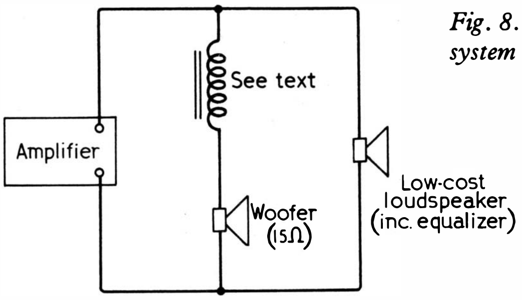


Fig. 8. Mono loudspeaker system with woofer.

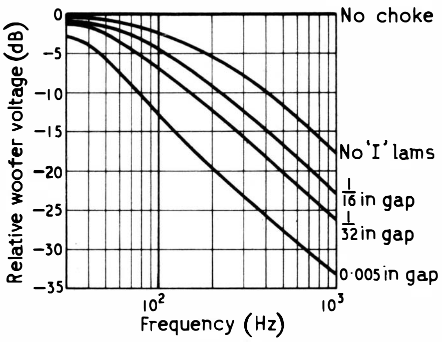


Fig. 9. Variation of woofer voltage with frequency for various choke gaps.

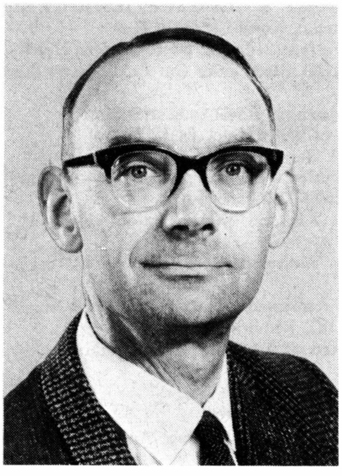
Core, bobbin and shroud. Belclere Kit LX, in Silcor (1¼-in. stack of 0·014-in. laminations, Inter-Service No. 417, maximum dimension 2¼ in.).

Winding. 300 turns of 24 s.w.g. enam.

The choke gaps may be adjusted, using cardboard or other insulating material, to vary the output of the woofer, and Fig. 9 shows the effect of so doing on the voltage across a 15-Ω woofer mounted on a baffle board. The best setting may be determined subjectively.

In a stereo system, the simplest arrangement is to feed a single woofer plus choke from the power amplifier of one channel only. This may seem very crude, because the woofer does not receive the proper sum signal, but the fact remains that it is fairly satisfactory in practice. It is conceivable, of course, that one might come across a stereo record with nearly all the low bass in the channel not feeding the woofer, but the author has yet to meet such a case among classical records!

If, however, the above simple solution does not seem attractive, there is more than one possible way of feeding the woofer with a genuine sum signal. A very satisfactory method is to connect the woofer plus choke between the live output terminals of the two power amplifiers and introduce a simple unity-gain phase-inverting stage at a suitable point in one channel, probably between the control unit and the power amplifier. The connections to one of



Peter J. Baxandall held an amateur radio licence (2AZS) while at King's College School, Wimbledon, but became increasingly fascinated by the problems of low-distortion sound reproduction, electrical musical instruments, etc. He obtained his degree in electrical power engineering at Cardiff Technical College in 1942, later becoming a radio instructor in the Hankey Radio Training Course there. He moved in 1944 to T.R.E. (now R.R.E.), Malvern, and worked on microwave techniques for the first two years before joining F. C. Williams's team on electronic circuit research work. He has latterly co-operated with industry in developing transistor power amplifiers.

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* The author does not like the term "r.m.s. output power", despite its almost universal use nowadays. The product of r.m.s. voltage and r.m.s. current is not r.m.s. power, but *mean* power.

the smaller loudspeakers must, of course, also be reversed, to restore the loudspeaker outputs to their correct phasing. (The alternative scheme of reversing the connections to one half of the stereo pickup is not such a good idea as it might at first sight appear to be—quite apart from the fact that it does not cater for the stereo radio aspect of the problem. This is because, while it will work all right under stereo conditions, the normal control unit switching arrangements connect the two halves of the pickup in parallel under mono conditions, giving, ideally, zero output if the connections to one half have been reversed.)

An incidental advantage of feeding the woofer from the two power amplifiers as just described is that the signal level is 6 dB higher than when fed from only one amplifier, assuming the two amplifiers give in-phase contributions to the woofer at these very low frequencies.† If the bass output is too powerful, it may, of course, be reduced by decreasing the choke gap, but it is probable that the extra output at these very low bass frequencies will be felt to be beneficial in most rooms.[1]

An alternative and equally effective method for obtaining the sum signal for feeding the woofer is shown in Fig. 10. The transformer can employ a Belclere Kit LX, as used for the woofer choke, each winding consisting of 150 turns (in one section) of 24 s.w.g. enamelled wire. The laminations should be interleaved, i.e., no gap.

The connection of a transformer, as in Fig. 10, directly across the output of some transistor amplifiers of the type having no output transformer, is, however, inadvisable, being likely to lead to the breakdown of one or both of the output transistors should an accidental very-low-frequency overload occur—caused, for example, by mishandling the pickup. This is because the transformer inductance presents the transistors, at very low frequencies, with a low value of almost purely reactive load, giving an instantaneous combination of high collector current and high collector voltage not met under more normal load conditions. No trouble with such breakdown effects is likely to be experienced with valve amplifiers, however.

An economical scheme, which feeds the woofer with a genuine sum signal without requiring a transformer, and which can be used quite safely with a transistor amplifier, involves connecting two resistors, of about 15 Ω each, in series between the live output terminals of the two power amplifiers, the woofer and its series choke being fed from their junction. While this arrangement draws extra power from the amplifiers, and gives reduced electromagnetic damping of the woofer, it has been found to work quite nicely in practice. The resistors should preferably be wire-wound, with a rating of at least 3 W each.

Yet another arrangement, which may be favoured by readers possessing a spare mono amplifier, is shown in Fig. 11. It is here assumed that the amplifier input impedance is at least 100 kΩ. If it is lower than this, the impedance values in the circuit should all be reduced appropriately.

The circuit shown in Fig. 12, which is believed to have been used on the Continent, has the advantage of requiring neither special iron-cored components nor an extra amplifier. It cannot be strongly recommended, however; the resonant interaction of the motional impedances of the three loudspeakers leading to peculiar dips and peaks in the frequency response which are difficult to predict or control satisfactorily.

A system employing a single woofer operated off the sum signal can give considerably less turntable rumble than a normal system, and this is a very real advantage. Rumble vibrations tend to be largely in a vertical plane, and the single woofer is non-responsive to vertical stylus movements, which normally represent the stereo difference signal. When two separate full-range loudspeakers are used, vertical stylus movements give, ideally, equal and antiphase outputs from the loudspeakers, but, because of acoustic effects in the room, the outputs do not, in general, cancel at the listener's ears—indeed, if they did, there would be no stereo effect! The loss of stereo effect inherent in the use of a single woofer does not seem to matter, provided, as in the present scheme, it is confined to very low frequencies only.

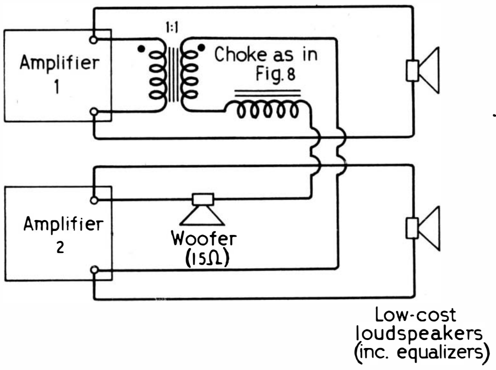

Fig. 10. Stereo system with phase-inverting transformer for feeding woofer.

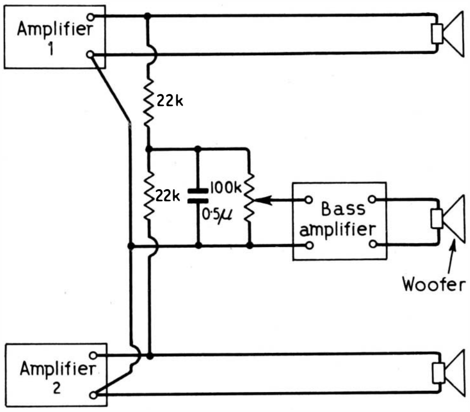

Fig. 11. Stereo system with separate amplifier for feeding woofer.

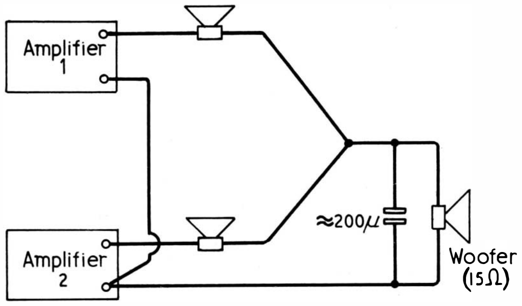

Fig. 12. Simple stereo system with capacitor across woofer—not strongly recommended.

Measurements on the drive unit

The electro-mechanical constants of the loudspeaker drive unit were determined by the following set of measurements and calculations.

The unmounted loudspeaker unit was placed face upwards on a table and fed at low level from an oscillator via a 1,000-Ω series resistor. An oscilloscope (10 mV/cm sensitivity) connected across the speech coil enabled the oscillator to be set to the diaphragm resonant frequency, f_o, as indicated by a maximum waveform amplitude. A series of ordinary brass balance weights was then carefully placed on the diaphragm near the coil, giving modified values of resonant frequency. A graph of $(1/f_o)^2$ against the mass added was then plotted. This was a good straight line, with an intercept at −5·7 gm, so the effective diaphragm mass was taken to be of this magnitude. The total mass coresponding to a particular resonant frequency could then be obtained, enabling the diaphragm

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† An objection to the scheme might seem to be that each amplifier will "see" a load impedance of only half the woofer-plus-choke impedance. However, because the impedance of a nominally 15-Ω speaker is much higher than 15 Ω in the region of its resonant frequency, except, perhaps, if it is mounted in a phase-inverter cabinet, and because of the high impedance of the choke at higher frequencies, it is found in practice that there is little reduction in the apparent power-handling capability of the system.

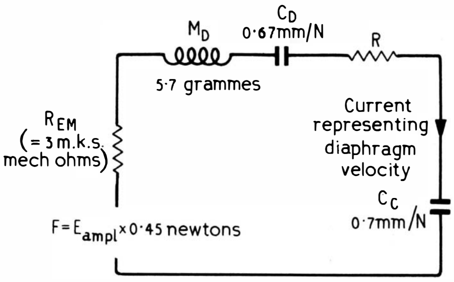


Fig. 13. Analogous electrical circuit representing mechanical system of loudspeaker unit and cabinet.

suspension compliance to be determined from the relationship:

$$f_o = \frac{1}{2\pi\sqrt{MC}} \qquad (1)$$

where: M = mass in kilogrammes, C = compliance in metres per newton, f_o = resonant frequency in Hz.

The value obtained for the compliance was 0·67 mm/N.

A little wire tripod was then made, whose feet rested on the junction between the central dome and the paper diaphragm. This tripod carried a fine pointer arranged to indicate displacement against a scale of millimetres fixed to a wooden "bridge" resting on the loudspeaker frame. With care, it was found possible to estimate tenths of a millimetre. Measured direct currents were then passed through the coil, in both directions, and a graph of displacement against current plotted. This was fairly straight, with a slope of 4·5 mm/amp, up to about ±1 mm. From this figure, and the compliance figure previously obtained, a force/current relationship of 6·7 newtons/amp was deduced.

The aim was to obtain all the values of the mechanical elements represented by the analogous electrical circuit of Fig. 13[8,10]‡ The suffixes D and C are used for quantities associated directly with the diaphragm and cabinet respectively. F is the force which would be produced by the speech coil if it were prevented from moving. This force, as already mentioned, is 6·7 N/A. If E is the amplifier output e.m.f. in volts, and if we take the sum, R_{tot}, of the coil resistance (13·5 Ω) and the amplifier output resistance as 15 Ω, then $F = 6{\cdot}7 \times E/15$, i.e., $F = E \times 0{\cdot}45$ newtons.

R_{EM} in Fig. 13 is the mechanical resistance introduced by electromagnetic damping, i.e., it is the ratio of the force produced by the speech coil when blocked to the velocity it would have if completely free and massless. It may be shown that:

$$R_{EM} = \left(\frac{F}{I}\right)^2 \times \frac{1}{R_{tot}} \qquad (2)$$

where: F/I is in N/A, R_{tot} is in ohms, R_{EM} is in m.k.s. mechanical ohms.

For the present loudspeaker unit, R_{EM} comes out at 3 m.k.s. mechanical ohms.

C_C is the compliance associated with the volume of air enclosed in the cabinet, referred to the diagram. It is obvious that the larger the diaphragm area, the greater will be the increase of air pressure in the cabinet for a given diaphragm movement, and that a given increase in pressure will produce a force on the diaphragm proportional to its area. Hence:—

$$[\text{Air compliance}]_{\substack{\text{w.r.t.}\\ \text{diaphragm}}} \propto \frac{\text{cabinet volume}}{(\text{diaphragm area})^2} \qquad (3)$$

(An alternative method of calculation involves the use of acoustical impedances rather than mechanical impedances. Acoustical impedance is pressure/volume-current rather than force/velocity. Compliance is then volume change/pressure and is a function of cabinet volume only and not diaphragm area.)

When the air in the cabinet is suddenly compressed, its temperature rises. If the increased pressure is maintained, the air will cool down again, giving a further volume reduction. Thus, for very slow changes, the compliance is higher than for faster changes. When, as normally applies for a loudspeaker cabinet, even at low audio frequencies, there is no time for the air to cool down after compression, the operation is said to be adiabatic, as compared with isothermal for very slow changes.

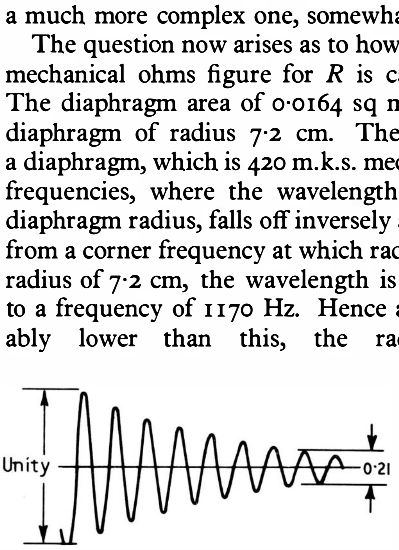
The effective diaphragm area for the Elac unit used, measured to the "mid-point" of the surround, is approximately 0·0164 sq m. The effective cabinet volume is approximately 0·0252 cu m. This leads to the result that, with adiabatic operation, the compliance of the air, referred to the diaphragm, is 0·70 mm/N.

It will be noticed that the compliance due to the air is about equal to that of the unit itself, giving a rise in resonant frequency by a factor of about $\sqrt{2}$. The calculated resonant frequency in the cabinet is 114 Hz, which agrees quite reasonably with that determined experimentally. (A complicating factor is that it is found in practice that the resonant frequency of the unit depends considerably on the applied voltage at which it is measured.)

The remaining element to be determined in Fig. 13 is the mechanical resistance R, representing diaphragm suspension losses and radiation resistance. The latter varies rapidly with frequency, but the value of R at the resonant frequency is of particular interest. One of the simplest methods for determining R is to connect the contacts of a relay in series with the speech coil and a d.c. supply such as a dry cell. The relay is operated at some quite low frequency, e.g., 1 Hz, by means of a multivibrator, or in some other convenient way. When the contacts open, the electrical resistance in the speech-coil circuit becomes infinite, making R_{EM} zero. The only resistance effective in the mechanical circuit is then R, and a damped oscillatory voltage appears across the coil, as shown in Fig. 14. The Q value may be determined from the rate of decay of the oscillation, and a convenient fact is that the Q value is equal to the number of *half* cycles that occur while the oscillation amplitude is decaying from a value of unity to a value of 0·21 of unity. This test performed on the present loudspeaker, with no felt in the cabinet, gave a Q value of 15 and a natural frequency of 110 Hz. The reactance of the 5·7 gm diaphragm mass at 110 Hz is 3·9 m.k.s. mechanical ohms, so that, with $Q = 15$, R is $3{\cdot}9 \div 15$, i.e., 0·26 m.k.s. mechanical ohms. Thus, when the unit is fed from a *low* impedance source, the total mechanical resistance is 3·26 m.k.s. mechanical ohms, and the Q value is 3·9/3·26, i.e., 1·2.

It was interesting to observe that, while a nice simple exponentially damped sine wave was obtained with the cabinet properly sealed, quite a small leak, such as that mentioned earlier, caused by incorrectly-fitted expanded aluminium, converted the waveform to a much more complex one, somewhat as sketched in Fig. 15.

The question now arises as to how much of the above 0·26 m.k.s. mechanical ohms figure for R is caused by radiation resistance. The diaphragm area of 0·0164 sq m is the same as for a circular diaphragm of radius 7·2 cm. The radiation resistance seen by a diaphragm, which is 420 m.k.s. mechanical ohms per sq m at high frequencies, where the wavelength is small compared with the diaphragm radius, falls off inversely as the square of the frequency[8] from a corner frequency at which radius/wavelength ≑ 0·25. For a radius of 7·2 cm, the wavelength is thus 28·8 cm, corresponding to a frequency of 1170 Hz. Hence at any frequency, f, considerably lower than this, the radiation component of the



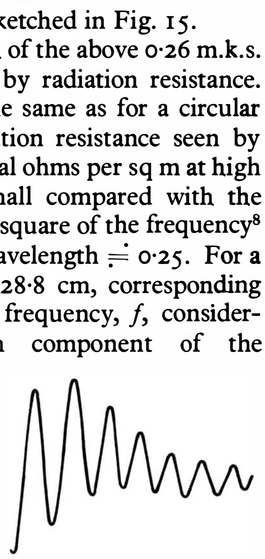


Fig. 14 (Left). Open-circuit coil voltage following interruption of current.

Fig. 15 (Right). As for Fig. 14, but with air leak in cabinet.

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‡ The analogy used is that voltage represents force, current represents velocity. Hence voltage/current, i.e., electrical impedance, represents force/velocity, i.e., mechanical impedance. Just as the reactance of an inductance L is $2\pi fL$, so the mechanical reactance of a mass M is $2\pi fM$ mechanical ohms, etc.

mechanical resistance in Fig. 13 is $0{\cdot}0164 \times 420 \times (f/1170)^2$, i.e., $5{\cdot}0 \times 10^{-6} f^2$ m.k.s. mechanical ohms.

The acoustical power radiated is equal to the square of the diaphragm velocity times this radiation resistance—equivalent to "$P = I^2R$" in an electrical circuit. Hence the radiated power will be independent of frequency if the diaphragm velocity is proportional to $1/f$. At frequencies well above resonance in the Fig. 13 circuit, the reactance of M_D becomes the dominant mechanical impedance, giving a velocity of $F/2\pi f M_D$. As already discussed, $F = E \times 0{\cdot}45$ newtons, M_D is $5{\cdot}7$ grammes. The radiation resistance is $5{\cdot}0 \times 10^{-6} f^2$ m.k.s. mechanical ohms. Hence the power radiated is:

$$\left[\frac{E \times 0{\cdot}45 \times 10^5}{2\pi f \times 5{\cdot}7 \times 10^{-3}}\right]^2 \times 5{\cdot}0 \times 10^{-6} f^2\,\mathrm{W}$$

$$= E^2 \times 0{\cdot}80\,\mathrm{mW}.$$

A 10-W amplifier designed for a 15-Ω load will give an output voltage of 12·2 V r.m.s., so that, from the above, the acoustic output from the loudspeaker at 10-W level is 120 mW. Hence, for practical purposes, the efficiency may be taken as 1·2%.

In the Fig. 13 circuit, the power output is $(\text{current})^2 \times$ (radiation resistance). But radiation resistance is proportional to $(\text{frequency})^2$. Hence power output is proportional to $(\text{current} \times \text{frequency})^2$. Now the voltage across the inductance is proportional to (current × frequency), and this leads to the useful idea, pointed out by D. E. L. Shorter in reference 2, that the output power is proportional to the square of this voltage, or the pressure produced by the loudspeaker in free space is proportional to the voltage across the inductance. In this context, the circuit may conveniently be redrawn as shown in Fig. 16. This is a well known circuit, whose normalized frequency response, is given in reference 11. With a Q-value of 1·2, as determined above, the response would be expected to exhibit a peak of 2·4 dB just above the resonant frequency and to become asymptotic at very low frequencies to a 40 dB/decade (12 dB/octave) line going through 0 dB at the resonant frequency. The measured acoustic frequency response of the loudspeaker, Fig. 5 in Part I, will be seen to approximate fairly closely to this at low frequencies.

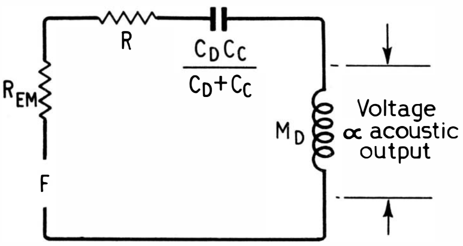

Fig. 16. Rearrangement of Fig. 13.

In conclusion, readers employing Vinkor for mounting clips for the equalizers may find it convenient to order mounting boards. These are DT2233 for the larger core and DT2227 for the smaller core. If, however, the whole equalizer is built on a piece of 1/16th inch s.r.b.p., holes for attaching the clips may be drilled in this and there is no need to employ Mullard boards.

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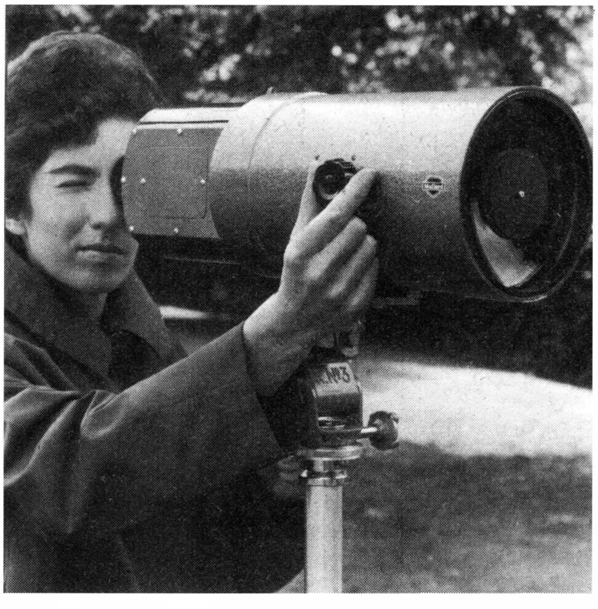
Image intensifier for the visible spectrum

Fibre optics are used for high-efficiency transfer of optical images between three photo-electric stages in a new image intensifier tube introduced by Mullard for "seeing in the dark" in the visible spectrum. Originally produced in collaboration with Government establishments for military purposes, the tube has now been made available for civil use. Possible applications include navigation, aerial reconnaissance, space and underwater exploration, astronomy, nature studies of nocturnal animals, and aiding police and other authorities in night surveillance. The tube also makes possible the use of closed-circuit television in conditions of very low ambient lighting.

The tube uses a wide-diameter objective lens to collect as much as possible of the light reflected by the object or scene being observed. This optical image is focused on to the photocathode of the first stage, producing a corresponding pattern of electrons. These electrons are then directed and accelerated by an electrode system connected to a potential of 15kV, and fall on a phosphor screen. Because of their high velocity they cause more photons to be emitted from this screen than were received by the photocathode. Hence the original image is intensified. To ensure that as much of the light as possible is transferred from the phosphor screen of the first intensifier stage to the input photocathode of the second stage (and so on) the optical image is transferred by fibre optics. The input and output windows of the stages are plano-convex, giving flat images and so simplifying optical coupling. Finally, a visible image is produced on a small screen 25mm in diameter. According to the manufacturers, the sensitivity of the tube makes it possible to clearly recognise objects under starlight conditions.



The image intensifier being demonstrated by Daphne Lamport, project leader of the group responsible for the early development of the tube.