An acoustically small loudspeaker

Unusual design gives low colouration and good off-axis response

by **R. I. Harcourt** B.Sc., M.I.E.E.

This design for an active-crossover loudspeaker system is based on acoustic principles which are well established, and on psycho-acoustic criteria which are subjective in nature. As with all designs, trade-offs are possible. The acoustically small loudspeaker is designed to reduce colourations of the sound, below the limit of audibility where possible. This can be done at the expense of bass distortion, though since 40% second harmonic distortion is inaudible at 80 Hz[1], it is not considered important. In addition, a novel fourth-order, bandpass sub-woofer is described using an acousto-electronic crossover and feedback *Q* correction.

The basic aims of the design were low colouration and a uniform off-axis frequency response. A flat on-axis frequency response is the accepted criterion, but the off-axis response – often compromised in commercial designs – determines the stereo imaging qualities. Colouration and off-axis response depend upon both the drive unit and its enclosure, particularly in the mid-range, where the ear is most sensitive: it is between 1-4 kHz that most of the image is found. To avoid compromising this part of the spectrum, the Jordan 50mm aluminium-coned unit was used for its small size, low colouration, and good transient response – this was the only drive unit found for which the impulse response is published. It must be emphasized that it was not designed as a mid-range unit, and is specified to 22 kHz: the booklet advocates using the unit, together with a bass driver, to form a two-way system[3]. However, to the author's and a colleague's ears, an improvement was obtained with the use of a dome tweeter above 4 kHz, making a three-way system. Whichever way the unit is used, there are no crossovers in the critical range 500 Hz to 4 kHz to detract from the imaging quality by giving rise to an uneven polar response around the crossover frequency[2].

The design of a loudspeaker is often influenced by the ease, or otherwise, of its manufacture. For example, it is rare to find other than a cuboid of wooden construction used for the enclosure. But even ideal drive units are at a disadvantage in a wooden cuboid. The shape, size, materials and construction of an enclosure all have audible effects on the response. The great advantage of home construction is that one is freed from many of these constraints, and this advantage is exploited in the design. A mid/high frequency enclosure is made of modelling clay which does not require firing. Thus, the shape and materials of the enclosure are optimized.

Cavity resonance

Distortions in musical sounds take several forms, and the total harmonic distortion is often quoted. More recently, it has been found that this measurement does not correspond well with how a unit sounds: indeed, sometimes a valve amplifier with a high t.h.d. is preferred to a transistor design. It has become clear that steady-state measurements do not give a good indication of performance, and other measurements have been used. With pickup cartridges and loudspeakers, there is a large variation between units sometimes expressed as "detail" or "dynamics", perhaps due to the presence or absence of masking effects of one sound upon another. More complex effects have been found, which are time-dependent, such as the 1 millisecond forward inhibition of a sound upon a following one, and the 30-120 millisecond backward inhibition of a sound upon a preceding one[4]. These

could be stimulated by delayed resonances in transducers, which often have time-constants within these ranges. With this in mind, and the author having a particular dislike of the sound of delayed resonances, the design for the acoustically small loudspeaker sets out to minimize them.

The cavity resonances of an enclosure constitute an inharmonic series given by the solution to the wave equation for rectangular (or other) boundary conditions. The cuboid has resonances at

$$f_{\mathrm{nx,\,ny,\,nz}} = \frac{c}{2}\sqrt{(n_x/x)^2 + (n_y/y)^2 + (n_z/z)^2}$$

where c is the velocity of sound, n_x, n_y, n_z are integers chosen separately and x, y, z are the dimensions. These resonances can be heard because they present a widely varying acoustic load to the rear of the diaphragm and thus affect its motion. At high frequencies they can be damped using acoustic filling material, but this is not true at lower frequencies, nor necessarily for small enclosures; for both frequency and thickness of material affect the absorption. An acoustically small loudspeaker is of such a size that the lowest, and therefore all, the cavity resonances are outside the passband, and the loudspeaker is used below these frequencies. This applies, then, to a bass unit. Choosing $n_x = 1$, $n_y = 0$, $n_z = 0$ gives the lowest resonance at

$$f = c/2d$$

where d is the largest dimension of the enclosure, and this is the one-dimensional half-wave standing wave.

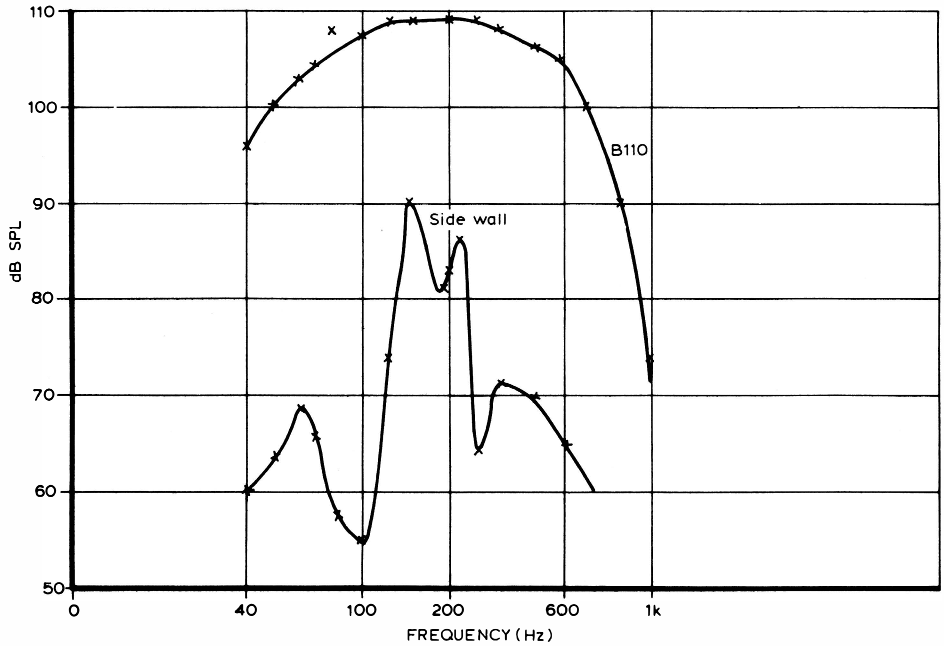

Fig. 1. Near-field transmission of cabinet side compared with that from B110.

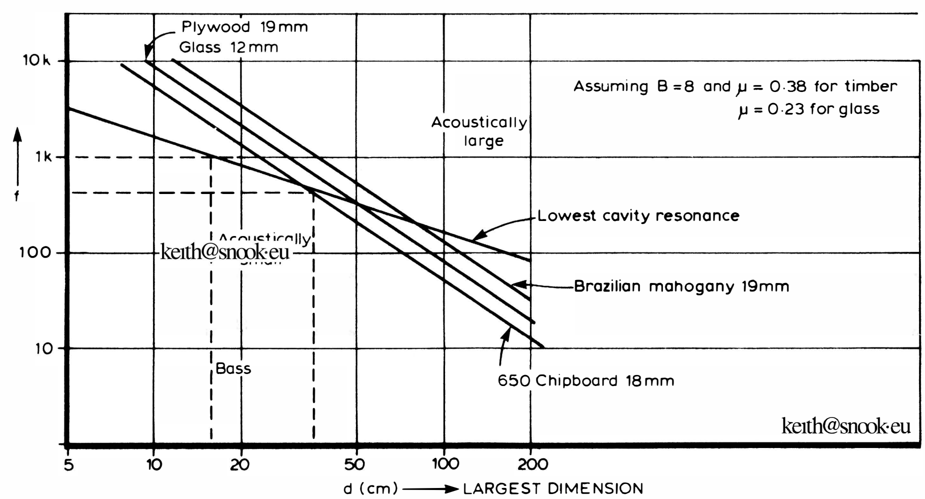

Fig. 2. Cavity and panel resonances for varying maximum dimension.

Panel resonance

Many loudspeakers have a "boxy" sound while reproducing male speech. The box can produce sounds in various ways, one of which is given above. Another way is by the panels of the box vibrating. It has been found[5] that at certain resonant frequencies, the output of the box is within a decibel or so of that of the loudspeaker. As an experiment, some enclosures were made after Linkwitz[2], constructed of 6mm plywood with a 10mm internal layer of roof-patching tar. The transmission of the cabinet side-panel was measured by placing two such units together, fed by sine-waves of equal amplitude but opposite phase, so as to null the sound from the loudspeaker. The microphone was placed 1cm from the side-panel so that the near-field response was measured. The results are shown in Fig 1. After correcting for the relative emitting area of the panel and allowing for two panels, the output from the box at 150 Hz was found to be about 8dB below that from the drive unit. Since the Q was measured to be 5, the box will continue to produce the sound after the drive unit has finished, which constitutes a delayed resonance. In this case the 40dB decay time will be $Q/0.7f = 48$ms.

A panel has a series of resonant frequencies, the lowest one of which is at

$$f = \frac{Bt}{2\pi a^2}\sqrt{\frac{E}{\rho(1-\mu^2)}}$$

where t is the panel thickness, a its dimension (for a square panel), E the Young's modulus of the material, ρ the density and μ the Poisson's ratio. B varies according to the construction, and is higher for a clamped panel than a freely supported one. For loudspeaker enclosures, B is taken as 8. An acoustically small loudspeaker can be made so that the lowest, and therefore all, the panel resonances are above the frequency of operation. The two resonance functions mentioned are plotted in Fig 2. From this can be determined the maximum dimension of an enclosure for it to be acoustically small. It can also be appreciated that most loudspeakers are acoustically large. The graph of the lowest cavity resonance coincides with a criterion for determining the maximum frequency at which to operate a drive unit to ensure wide dispersion, and the maximum enclosure width.

The sound emitted by an enclosure depends upon its dimension and the degree of its motion. For a circular piston, the emitted sound pressure level increases by 12dB for a doubling of its diameter, which implies that as a box is made smaller, so the sound radiated from it decreases. However, the internal pressure within the box increases as the volume is decreased, so that the deflecting force on the panel increases. This is compensated by a decrease in the actual deflection with reducing dimension, according to a square law. The combined effect of all this is a decrease of about 6dB in the emitted sound with a halving of the dimension. All the above factors represent a confluence of ideas pointing to acoustic size as being an important parameter. It is therefore no coincidence that listening tests have revealed a preference for small loudspeakers, provided that these are also well designed in other respects.

The panel transmission loss below the first resonance depends on the stiffness of the material used, not its mass or damping properties. The bass enclosure is best constructed of a thick material of high Young' Modulus. In this respect plywood is better than chipboard and hardwood better than plywood. Glass would seem to be an ideal material, for it has a Young's modulus 75 times that of chipboard, and an enclosure can be fabricated in the same way as an aquarium, using silicone rubber as an adhesive. This is a subject for further work.

Clay enclosure

Diffraction round an enclosure has been found[6, 2] to have a bearing on the frequency response and stereo imaging qualities of an enclosure, and Fig. 3 shows the frequency response of differently shaped enclosures, other things being equal. The sphere was found to give the smoothest response of the shapes tested, since there are no discontinuities in the surface to give rise to frequency-dependent effects. A novel enclosure is made as close to a sphere as practicable, and consists of a short vertical cylinder with domed top, as shown in Fig. 4. The shape is achieved by using modelling clay, which has a high density and large internal losses – it is acoustically "dead". The clay used is sold under the trade name of "Das", and does not require firing. It is not possible to include the bass unit in this enclosure, so only the mid/high-frequency unit or units are placed in it, and it is stacked on top of the bass enclosure. Because of the rounded shape, advantages are obtained in suppressing cavity resonances. The top-to-bottom, one-dimensional standing wave which normally occurs in a pipe is suppressed by the domed top, and similarly the axial one-dimensional standing waves are suppressed by the cylindrical walls. This leaves the two-dimensional waves, and the lowest is calculated to occur at 1.4 kHz, where the damping material used has a high absorption.

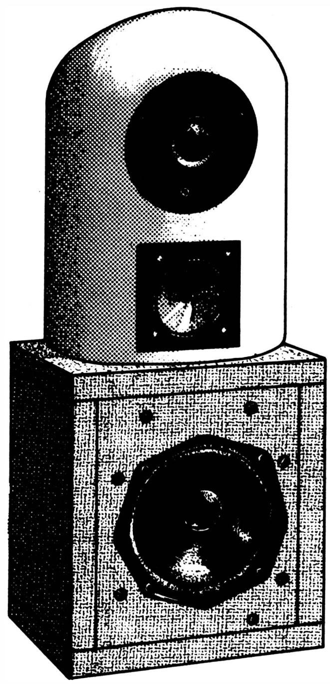

Fig. 4. Author's prototype. Clay enclosure on top contains mid and high-frequency units, while wooden bass enclosure is for B110. Single sub-woofer is not shown.

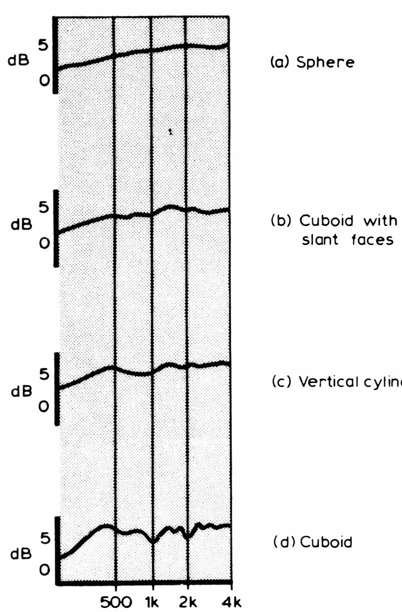

Fig. 3. Frequency response of four different cabinet shapes.

Bass enclosure

The simplest way of making an acoustically small bass enclosure is to make it physically small, and for operation up to 500 Hz the lowest resonance is placed higher, at 1 kHz, where the response is 20dB down. This, combined with the high absorption of the filling material and the internal losses of the enclosure material, will give only very small amounts of unwanted sound. The maximum dimension for the bass enclosure is found from Fig. 2 and is 16cm. The volume of a 16cm cube is 4 litres, and it is clear that this will give insufficient bass extension. A cube is to be avoided, since resonances coincide to give a higher Q, and the bass enclosure is best made with dimensions in the ratio 2.3:1.6:1, this being the ratio used for designing listening rooms. The maximum volume is around 1.1 litres, which is too small, and so a modification is called for. A 5in bass unit requires an enclosure dimension of 15cm on the front panel, and so the box is made square at 16cm: the other dimension is determined by the volume obtained from the design procedure for bass loading. The volume is divided internally by a partition placed to brace the magnet against the back panel, which also suppresses the offending double resonance caused by the square dimension.

An acoustically small loudspeaker does not have to be physically small, and this is achieved by a scheme of internal partitions, in which each subvolume is acoustically small, but is connected to the adjacent one by a low resistance path. The partitions simultaneously brace the panels, effectively sub-dividing them into smaller ones which are acoustically small. The smallest dimension of the box is the width to ensure wide dispersion, and this is equal to or slightly greater than that required to house the drive unit.

Sub-woofer

The bass extension in this design is obtained by a novel sub-woofer, the aim being to achieve economy in space and expense. A 12in bass guitar speaker is capable of producing high levels of bass below 100 Hz, and is inexpensive. However, it has a rather high resonant frequency, which was utilized by placing it in an acoustically small enclosure and using it below resonance, with a second-order filter, to give the required amount of bass boost. The closed-box enclosure acts like a second-order high pass filter, and the flat part of the response above resonance is made to fall off at 12dB/octave using the filter. The portion of the response below resonance which was falling off at an ultimate slope of 12dB/octave is made flat with the same filter. The falling part of the new response is tailored to form half of a 12dB/octave crossover, the other half being the natural fall-off of the bass enclosure

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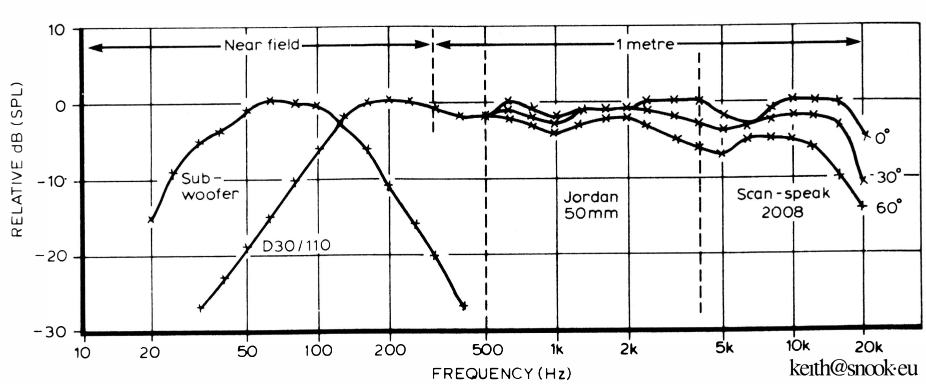

Fig. 5. Frequency response of system plotted using one-third octave pink noise signals.

whether it is ready or not), and to activate the peripheral at the correct time, that is when i/o instruction with address Aq in our case is being executed.

Peripheral status information is made available to the program through an input port. If the ready/unready state of the peripheral is indicated by the '0' and '1' values of signal *r* in Fig. 3, to determine whether the peripheral is ready or not, the programmer proceeds in the following manner. He executes an IN instruction with address Ap. Execution of this instruction copies the signals *rxxxxxxx* in Fig. 3 into the accumulator. If $r = 0$ the process is repeated, otherwise the next (i/o) instruction is executed, which allows the microprocessor to communicate with the peripheral, as shown in Fig. 4. The programmer has several options to determine the value of *r*. We shall describe two such options. He can AND the contents of the accumulator (*rxxxxxxx*) with 10000000 (80 in hex), which modifies them to *r*0000000. If $r = 0$ the zero flag is set, otherwise it is reset. Alternatively, he can shift the accumulator left through the carry flip-flop, as shown in Fig. 5, which shifts the value of *r* into the carry flip-flop.

If we assume that our peripheral is an action/status device, that is a device whose terminal characteristics are shown in Fig. 6, the hardware implementation of a test-and-skip system is shown in Fig 7. Action/status devices are described in Appendix 1 of "System Design with Microprocessors", Academic Press, 1978.

In the next article we shall demonstrate design steps by means of a PRINT problem. This problem, which will also be implemented using the wait/go, interrupt, d.m.a. and d.d.t. modes, has been chosen, first because a printing operation can be readily visualized and secondly, the character printer used can be assumed to have been in existence in the 1940s, that is well before the era of computers and microprocessors. □

WW index for 1979

The index for Volume 85 (1979) of *Wireless World* is now available, from the General Sales Department, IPC Electrical-Electronic Press Ltd, Room CP34, Dorset House, Stamford Street, London SE1 9LU, price 75p including postage. Cheques should be made payable to IPC Business Press Ltd.

We apologize for the unusually long delay in the production of this index. This was due to a combination of editorial staff problems and more general industrial disputes.

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containing the 5in unit. In this fashion, a two-way, second-order crossover is obtained for the price of a single-stage filter, and considerable bass extension with an inexpensive unit. The success of this design is best judged by the observation that, when placed in a corner, 102dB SPL peaks were measured while playing a recording of cannon shots during the "1812" overture, without any sign of stress. Naturally there is a price to be paid, and that is increased harmonic distortion. This could not be heard during music, but sine-waves or pink noise showed it up, and the result was that the source of sound could be located, which is not generally true for such low frequencies. Frequencies below 100 Hz were found to occur infrequently during most music, but bass guitar, bass drum, and organ enthusiasts may prefer some other solution below 100 Hz. That due to Linkwitz[2] is an alternative.

The on-axis response of the units was measured above 300 Hz in situ using third-octave pink noise. That below 300 Hz was measured by taking the near-field response of each unit to eliminate the effects of the room. The results are shown in Fig. 5. The off-axis response was measured above 300Hz in situ, rotating the loudspeaker. Curves are shown for 30 and 60 degrees horizontally off-axis, and show that an integration has been achieved between drive units, and that there are no large steps in the off-axis response to cause shifting or diffuse stereo imaging.

Design and construction will be described next month.

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3. Jordan, E. J., "The Jordan Manual", from the author.
4. Von Bekesy, G., "Auditory Backward Inhibition in Concert Halls", *J. Audio Eng. Soc.* p780 27 No 10 Oct 1979.
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6. Olsen, H. F., "Direct Radiator Loudspeaker Enclosures", *J. Audio Eng. Soc.* 17, No. 1, pp22-29 1969. □

IN OUR NEXT ISSUE

Unique pickup arm

By displacing the horizontal and vertical pivots of an arm from each other it becomes possible to increase the radius of the arc in which the pickup travels across the record and so reduce tracking distortion. This article describes a practical design for home construction.

Amplifier-loudspeaker interface distortion

Matti Otala examines distortion caused by intermodulation between the signal and a delayed, frequency transformed version generated by the loudspeaker and propagated in the feedback loop. Measurements on four power amplifier circuits are discussed.

Designing inductors carrying d.c.

It's difficult to select initially a core for a winding that is carrying d.c. A simple procedure allows different cores to be compared and the optimum one chosen for a particular inductor design.

On sale October 15

An acoustically small loudspeaker

2 – Construction of speaker enclosures and active crossovers

by **R.I. Harcourt** B.Sc., M.I.E.E.

The bass enclosure uses a Dalesford D30-110 5in unit and is 3 litres in volume, giving a 3dB point of 100Hz and a Q of 0.7 (measured values). The KEF B110, which is a rather similar drive unit, could also be used, but has not been tried. The enclosure is constructed of 18mm timber (in order of preference hardwood, plywood and chipboard) to ensure a low level of unwanted sound. The author used chipboard and, using the method outlined in part 1, was unable to measure any panel resonances or other sound transmission through the walls owing to their low levels (at least 30dB down). Dimensions are shown in Fig. 6. The bracing member is used to eliminate one of the two resonances arising from the square dimension, but was also found to remove a chassis resonance of the drive unit which coupled to the box at 320Hz, by bracing the magnet. The member is deliberately made slightly larger than the available space so that when the front panel is fitted the drive unit and rear panel are stressed. The enclosure is filled with 4oz long-fibre wool.

Sub-woofer

The sub-woofer required for adequate bass extension (below 100Hz) has a fourth-order, band-pass characteristic arising from the second-order, high-pass function of the closed-box enclosure system and a second-order boost filter. The 3dB frequency of the sub-woofer enclosure is made the same as that of the 5in bass unit, which is 100Hz. For analysis, the network functions for the band-pass sub-woofer and for the bass enclosure system were combined to produce a bi-quartic function to determine the Q and gain required from the components for a satisfactory 3dB point and low ripple. Normal analytical techniques were found inadequate for the case of the bi-quartic function, and the magnitude function was taken and explicitly solved using a home computer. These functions are shown in the Appendix.

The ideal Q for the enclosures was found to be 0.5, but this low figure was not practicable for the small enclosures used. The figure of 0.7 was taken for the Q and the method of Small[7] was used for the enclosure design. The higher Q implies some ripple in the response, but the computer prediction is that this ripple amounts to only 1.6dB, which is hardly audible, particularly since it is at 113Hz, where room eigentones are likely to give rise to much larger ripples. The frequency response predictions are shown in Fig. 7.

The design procedure for the sub-woofer enclosure was similar to that for the bass unit, but using the Son-Audax WFR 15S. The theoretical Q for the enclosure system was 0.7: however, when the enclosure had been built the Q was measured as 1, and stuffing the enclosure with long-fibre wool did not significantly reduce it. The effect of the higher Q is to introduce about 3dB of ripple into the response. To lower the Q, the old technique of feedback Q correction was used. If the output resistance of the amplifier is made negative by the introduction of positive feedback, then part of the voice-coil resistance is effectively cancelled, giving a lower Q. A theoretical treatment is found in the Appendix. Figure 8 shows the circuit of this arrangement for the case of the Crimson Elektrik ΩE608 amplifier module. The 47k negative-feedback resistor R_7 is part of the CE608 module, and the module must be modified in the following manner. Remove R_6 (1.5k) and C_4 (100μF, 10V) from the CE608 board. Replace R_6 by a 1k2 resistor. Take a screened lead from R_6 to the 100μF, 10V capacitor on the filter board. The 1N4148 back-to-back diodes are an insurance against mains transients, which can otherwise destroy the amplifier when it is used in this manner.

The summing amplifier, which combines the two channels, and the second-order filter circuits are also shown in Fig. 8. The filter is a conventional Sallen and Key type with an f_0 of 28Hz and a Q of 1. The summing amplifier provides the gain necessity for the bass boost of the below-resonance enclosure. Two 27k resistors combine the left and right bass channels within the metalwork housing the active crossovers and amplifiers. A screened lead takes this signal to the summing amplifier, which can be remotely sited. (In the author's installation, the two metalwork kits containing the crossovers and amplifiers are shelf mounted with the loudspeakers remote.) The screened lead takes the combined bass signal to the sub-woofer enclosure, where the filter p.c.b., CE608 amplifier and power supply are sited, enclosed in a Crimson Elektrik metalwork kit.

The WFR15S unit used for the sub-woofer has a high efficiency, which to some extent offsets the amount of bass-boost necessary for the below-resonance design. However, the limiting factor for all loudspeakers at the bass end, whether sub-resonant or not, is the cone excursion capacity, and not the power handling capacity as it sometimes supposed. Since the maximum s.p.l. available from a closed-box enclosure is determined by the area of the cone and the cone excursion, it is possible to increase the s.p.l. by increasing the number of drive units. The WFR15S has a cone-excursion-limited maximum s.p.l. of 86dB at 35Hz, the worst case considered here, since this limit increases at 12dB/octave. This may be compared with 88dB s.p.l. available from two KEF B139 units in a closed box, and with approximately 95dB s.p.l. from the full symphony orchestra. The author has considered using two WFR15S units in two closed-box enclosures placed adjacently, which would produce about 6dB more, i.e. 92dB s.p.l. at 35Hz, but space considerations led to the use of a single unit placed in a corner of the room, which gives a 9dB gain when compared with the free-space radiation figure, at no extra cost.

Clay enclosure

The mid-high frequency enclosure is made of modelling clay (Das) which has the property of setting rock hard without the use of firing. Plastic-wrapped 980g packs are available, and four packs were used for a pair of enclosures. However, the sides were found to be rather thin and six packs would be preferable. The clay is rolled flat with a rolling pin to the required dimensions and moulded round a cylinder of fine-mesh chicken wire. The cylinder is 20cm high and 40cm circumference. Cut a 60mm square hole in the chicken wire for the Jordan 50mm unit, and an appropriate round hole for the tweeter used. (The author has used both the Son-Audax tweeter (available as a smaller-faceplate version, the HD 9×8 D25) and the Scan-Speak 2008 pictured in Fig. 4.) The clay can be worked by those with no previous experience in the art, provided that it is remembered that a little moisture is required for smoothing down and for jointing. Avoid too much water.

Use two packs of Das for the cylinder and one for the base and dome. There will be some left over for patching holes and for surrounding the drive units with a fillet to provide a smooth profile. The wet clay will join to dry clay successfully when a little moisture is used, so it is not essential to obtain a perfect finish first time. The base and dome are made by rolling out one pack of Das to about 50cm square and cutting out two circles. One circle is made into the dome by placing the cylinder vertically in a

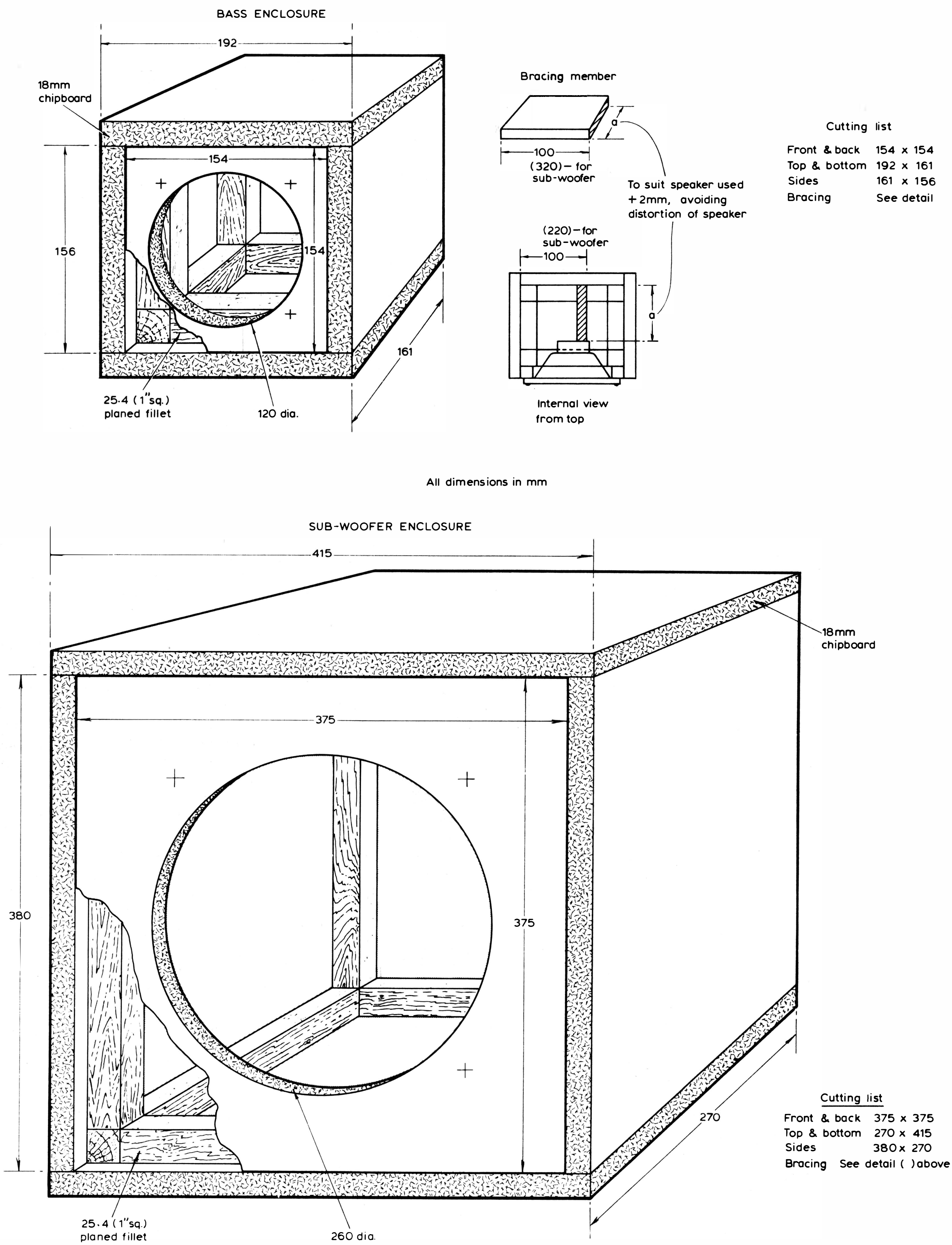


Fig. 6. Construction of the bass and sub-woofer enclosures.

mixing bowl and moulding the circle into the bowl, joining the edges with the end of the cylinder. The moulding is then placed upon the other circle and joined to this, forming the base. It may be found desirable to support the dome from the inside of the enclosure while it is drying to prevent sag. The clay takes about two days to set in an airing cupboard, and when set the drive units are fixed using silicone rubber (bath sealant). Holes are drilled for the wires to exit the back of the enclosure. About 4oz long-fibre wool is used as filling material, before the units are fixed. The adhesive takes 1-2 hours to set, and the spare clay (which should be kept in a plastic bag to prevent drying) can be used to form a smooth fillet around the edges of the units to reduce diffraction. For finishing, Declon acoustically transparent foam is formed into a cylinder and has a circle fixed to the top.

The cylinder of foam is overlapped by 0.5in and stapled together, and the circle for the top is stapled to the top edge of the cylinder. When this arrangement is turned inside out the join is concealed, and the sleeve can be fitted over the clay enclosure. The clay can be sprayed black to match the drive units, and the foam can be sprayed any desired colour, using aerosol paint.

The author has experimented with the positioning of the satellite units, and has found that stereo imaging is best when the units are placed on stands so that the tweeter is the height of the seated listeners' ears, and the units are 0.5-1m away from the walls. This requires stands about 470mm high.

Electronics

The Crimson Elektrik modules for the bass, mid-range and treble units are mounted in two Crimson Elektrik metalwork kits, with interconnecting socketry, as shown in Fig. 9. The kits are stacked on top of each other. One contains a normal two-channel amplifier with power supply, and the stabilized supply for the active crossover modules. The second kit contains four power amplifier modules and the active crossovers. If it is desire to omit the dome tweeters, as recommended by Jordan, then two amplifiers can be omitted also. Full instructions are supplied by Crimson Elektrik for the construction, but the filter p.c.b. for the sub-woofer is not supplied by them. Those wishing to make their own crossovers are referred to reference 2 where Linkwitz gives the circuit and design formulae. The crossover frequencies are 500Hz and 4kHz.

The sub-woofer electronics are housed in a suitable chassis together with a power supply. The author used a Crimson Elektrik metalwork kit and power supply, arranged as shown in Fig. 10. Because there is a positive feedback loop connected

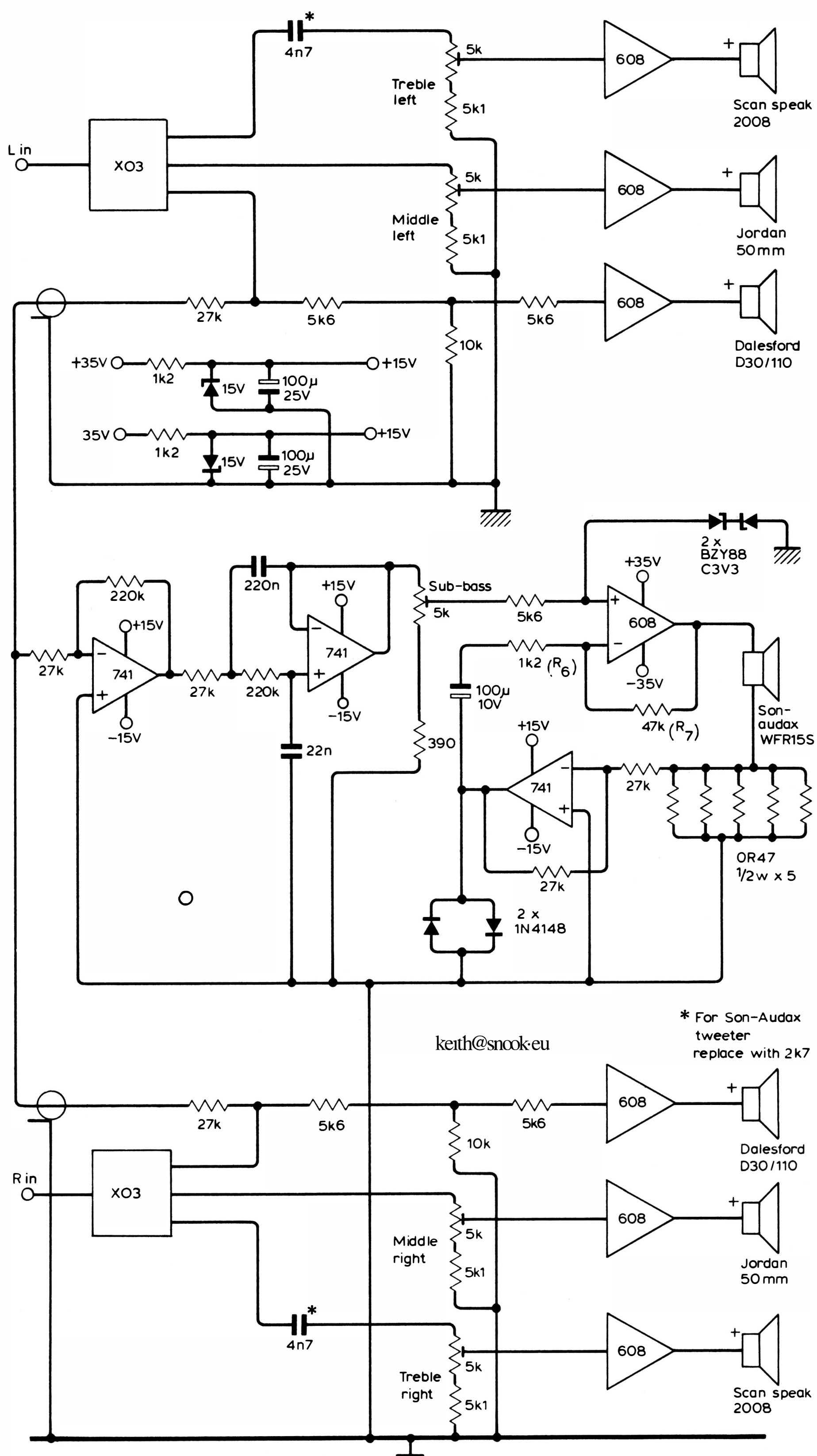


Fig. 8. Summing amplifier and filters for sub-woofer drive.

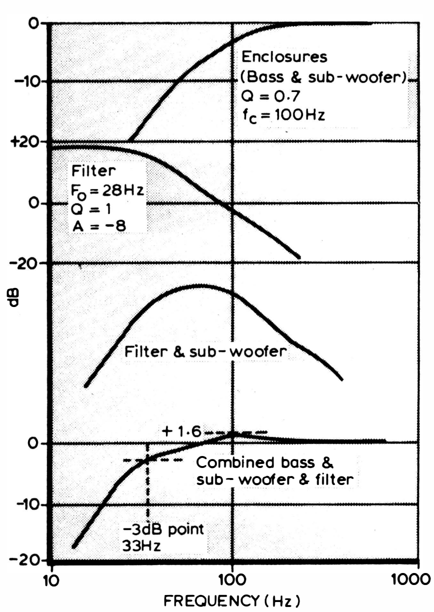


Fig. 7. Computer-predicted curve (bottom) for combined bass and sub-woofer responses.

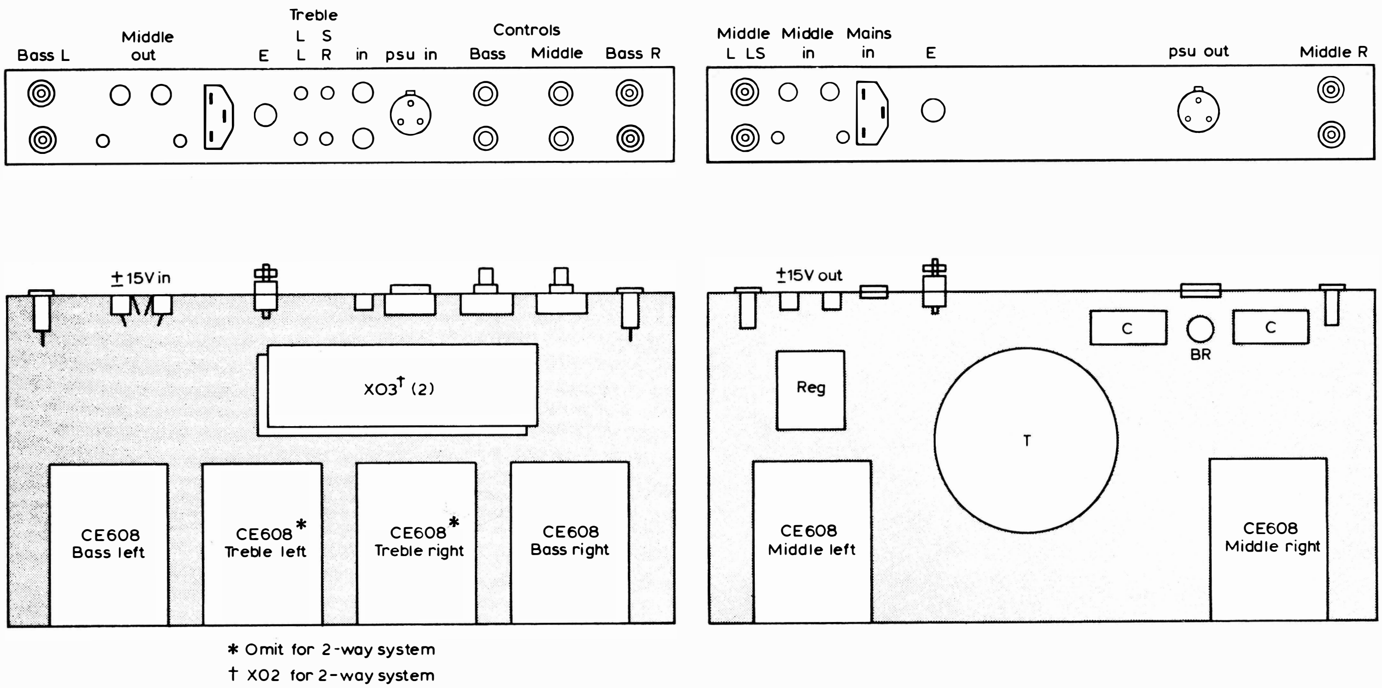


Fig. 9. Suggested layout of amplifiers and filters.

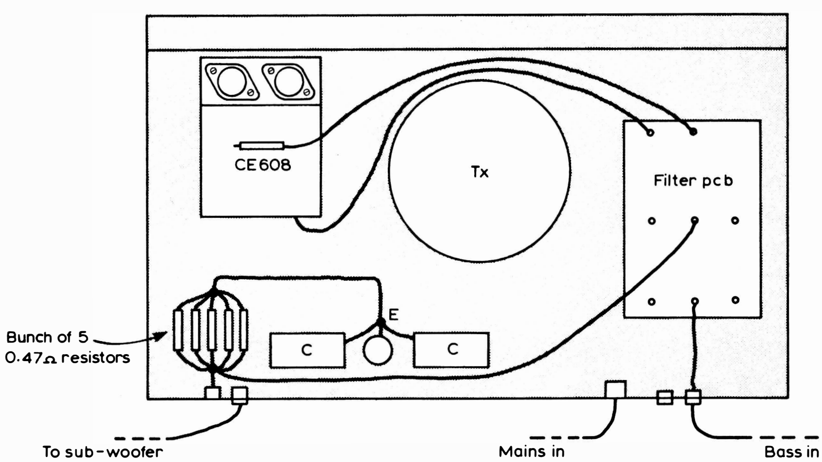


Fig. 10. Layout of sub-woofer amplifier and filter.

between two p.c.bs with screened cable, it is possible that mains harmonics may give rise to hum. To reduce this, use the thinnest screened cable obtainable, and route the cables well away from the mains transformer. The author has found that the screened cables are best positioned by experiment to give the lowest hum.

Listening tests

In each case, the audience was composed of electronics engineers. The first comparison was with the *Hi-Fi For Pleasure* Compact Monitor, a three-way system made from a kit, with much larger enclosures than those of the A.S.L. There was a large difference between the systems and the audience of three were unanimous in preferring the active system. The most noticeable difference was a gain in transient-attack on such instruments as guitar, piano and drums, when using the A.S.L.

The second comparison was with Spendor BC1s. In this case, the audience of two could just detect a difference, but were unable to tell which was in use. There was slightly more colouration or 'warmth' in the lower mid-range (about 500Hz) of the A.S.L. On some material, the greater bass extension of the active system could be heard. Direct comparison of stereo imaging was not found valuable, since the co-sited speakers interfered with each other's sound field.

Other considerations than sonic ones then become important. The larger size of a passive speaker may be a consideration, though price is not since, considering the cost of amplifiers and speakers, the two systems are comparable.

It should be mentioned that high-quality equipment (amplifier, deck, arm and cartridge) was used in the tests, in which analogue, digital and direct-cut records were played. The author is unable to measure or explain the slight 500Hz colouration, and intends to try the effect of lowering the crossover frequency.

Addresses

Crimson Elektrik: 1(a) Stamford Street, Leicester LE1 6NL
Sonaudax Loudspeakers Ltd: Main distributor is Falcon Acoustics, Tabor House, Norwich Road, Mulbarton, Norwich, Norfolk NR14 8JT
Dalesford: A.C. Farnell Ltd, Kenyon Street, Sheffield S1 4BD.
E.J. Jordan Ltd: Stonyway, Bovingdon Green, Marlow-on-Thames, Bucks SL7 2JH
K.E.F. Electronics Ltd: Tovil, Maidstone, Kent ME15 6QP
Scanspeak: Crimson Elektrik
Das is available from larger branches of W.H. Smith
Long-fibre wool can be obtained from Wilmslow Audio, Swan Works, Bank Square, Wilmslow, Cheshire, who also stock Dalesford, K.E.F. and Jordan drive units
Badger Sound Services, 46 Wood Street, Lytham St. Annes, Lancs FY8 1QG, stock most components, including Crimson Elektrik and wool, and are looking into the possibility of producing the sub-woofer electronics as a kit.

References

7. Small, R.H. "Closed Box Loudspeaker Systems" part 1 and part 2, *J. Audio Eng. Soc* 20 No. 10 and 21 No. 1, 1972 & 1973.

Appendix: sub-woofer characteristics

A closed box enclosure has network function

$$GH2(s_n) = \frac{s_n^2}{s_n^2+s_n/Q_o+1}$$

where $s_n = s/\omega_o$
$s = \sigma + j\omega$

The second-order, low-pass with gain A is

$$GL2(s_n) = \frac{A}{s_n^2/h^2+s_n/hQ_1+1}$$

where $h = \omega_f/\omega_o$

The two functions above in cascade give the fourth-order, bandpass

$$GB4(s_n) = \frac{As_n^2}{(s_n^2+s_n/Q_o+1)\,(s_n^2/h^2+s_n/hQ_1+1)}$$

Summing this with the response of the bass enclosure (1 channel only)

$$GH4(s_n) = \frac{As_n^2+s_n^2(s_n^2/h^2+s_n/hQ_1+1)}{(s_n^2+s_n/Q_o+1)\,(s_n^2/h^2+s_n/hQ_1+1)}$$

The magnitude function, which is too long to reproduce here, is then taken and programmed into a home computer. The result of evaluating this function is that, for a filter Q_1 of 1, an enclosure Q of 0.7 and an enclosure f_0 of 100 Hz, the gain required is $A = -8$, the 3dB-down requency of the system is 33Hz and ripple is 1.6dB at 113Hz. Note that the gain is negative, as is usual with second-order crossovers, when the drive unit is connected out of phase to achieve this. In the case of this design, the negative gain is achieved by the summing amplifier, and the driver is connected in phase.

For the D30/110 an enclosure volume of 3 litres was found to give the necessary Q and f_0. For the WFR15S an enclosure volume of 33 litres gave an f_0 of 100Hz, but the Q was too high. To reduce the Q the following expressions are considered from Small[7].

$$Q_{ts}=Q_{es}Q_{ms}/(Q_{es}+Q_{ms}) \qquad 1$$

$$Q_{es}=\omega_s C_{mec}(R_e+R_g) \qquad 2$$

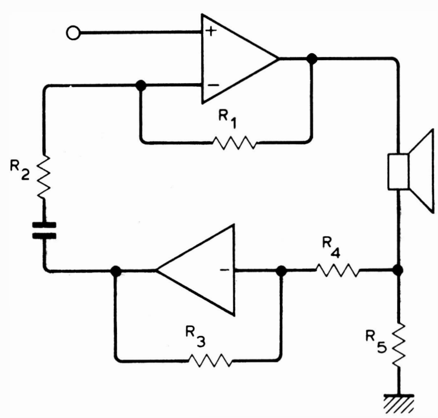
R_g is the output resistance of the amplifier and, by making it negative, Q_{ts} can be reduced to the desired value. The required Q'_{ts} is given by:

$$Q_c/Q'_{ts}=f_c/f_s$$

f_s = 31 Hz, f_c = 100 Hz, Q_c = 0.7. Therefore,

$$Q'_{ts} = 0.22$$

since $Q_{ms} = 4.19$, $Q'_{es} = 0.23$ from eqn. 1.



We know that with $R_g = 0$, $Q_{es} = 0.46$ and that $R_e = 7.3$, so from eqn. 2,

$$\omega_s C_{mec} = 0.063$$

and the required R_g is -3.7Ω.

The amplifier is given a negative resistance by the circuit shown.

By inspection:

$$R_g = \frac{-R_1R_3R_5}{R_2R_4}$$

For the CE608, R_1 = 47k. Choose R_5 = 0.1Ω, $R_3 = R_4$ = 27k. Then R_2 = 1.2k.

List of symbols

c velocity of sound 345 m/s
C_{mec} electrical capacitance representing moving mass
f frequency
f_c resonant frequency of closed box system
f_s free air resonant frequency of drive unit
G(s) response function of s
Q_{es} Q of driver at f_s considering electrical resistances R_e and R_g only
Q_{ms} Q of driver at f_s considering driver non-electrical resistances only
Q_c total Q of system at f_c
Q_{ts} total Q of driver at f_s considering all driver resistances and R_g
R_e d.c. resistance of driver
R_g output resistance of amplifier
s complex frequency variable (jω=σ)
ω_s radian resonant frequency of driver in free air □

The Author

The author was born in London, and lived as a child in Nairobi, Kenya. He attended Ipswich school, and from there went on to Southampton University, where he obtained an Honours Degree in Electronic Engineering in 1967. Appointed as an Executive Engineer in the Post Office HQ, he spent some time carrying out organisation and methods studies, before moving on to the Experimental Packet Switching System (EPSS) for which he helped to produce a mini-computer-based tester. Currently he is with the Mechanization and Building department, of Postal Headquarters, where he is developing a traffic recording system for parcel sorting.

Personal privacy of engineers

Mr F. W. Sharp of the Institution of Electronic and Radio Engineers writes to us as follows:

The suggestion was made in a BBC Radio 4 programme "Reel Evidence" broadcast on the evening of August 26th, that the membership lists of the chartered engineering institutions were freely available and could be used to compile unauthorized information about the work of individuals. As far as the IERE is concerned, this is not so: the Institution's membership list is maintained on a strictly confidential basis. By decision of the IERE Council a "list of members" is now no longer published.
F. W. Sharp