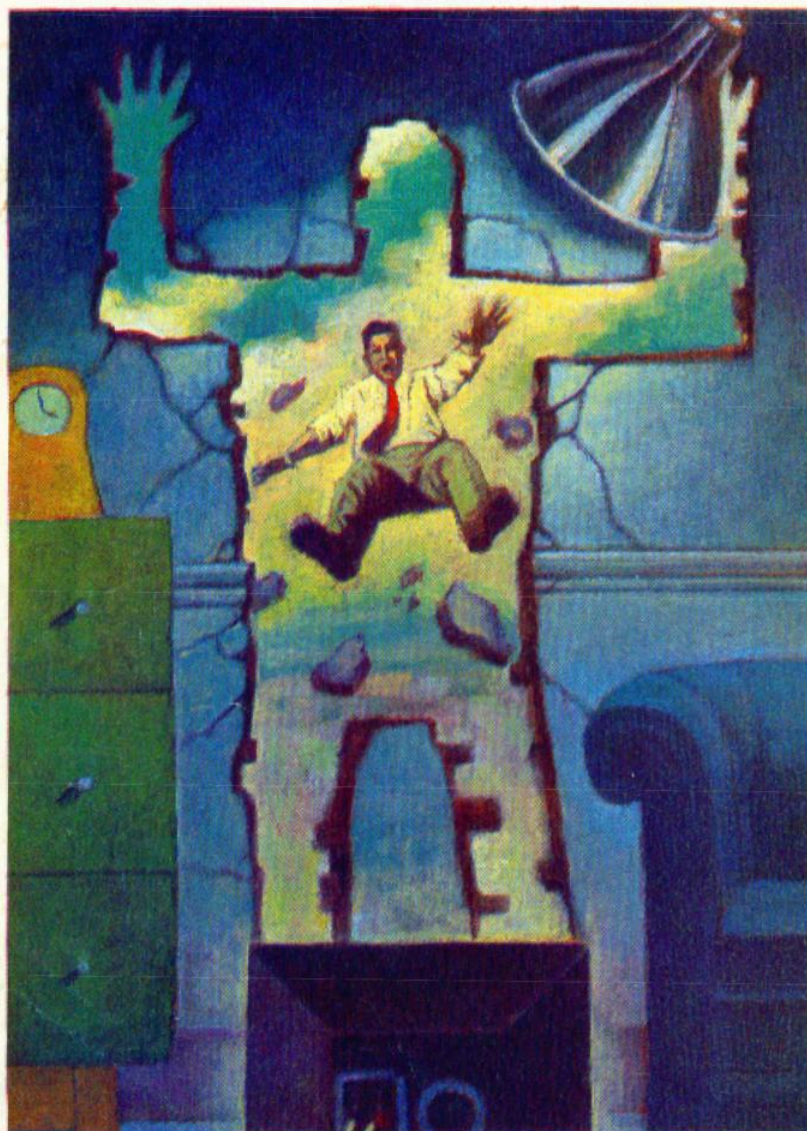


Roaring subwoofer



Using motional feedback, Russel Breden's subwoofer produces flat response down to 15Hz – despite its relatively small enclosure. Feedback also makes feasible an infinite baffle rather than a reflex design, resulting in tighter bass.

Inspired by Peter Baxandall's 'Low-cost, high-quality loudspeaker' series of articles in *Wireless World*, I built my first sub-woofer back in 1978. It was a 2.4ft³ reflex using a KEF B139 driver and tuned to 30Hz.

Originally, this sub-woofer was passive, but it soon became apparent that adding a dedicated power amplifier and second-order low-pass filter produced the flexibility required to interface with my existing speakers. Built in the days when home computers were a distant dream, and Thiele/Small analysis was little known, it is surprising that it worked at all. As it was, it gave me a sense of what was missing from most of the other systems around at that time.

Nearly twenty years later, things haven't changed that much. Off-the-shelf speaker systems available today rarely produce an output below 60Hz.

Going lower

For a system to produce bass extending to at least 30Hz normally requires large boxes and expensive drivers. But by creative use of electronic circuitry, both size and expense can be cut to reasonable proportions. These techniques however require the sacrifice of hi fi's most sacred cow – flat amplifier response.

Today, we have all the tools necessary to design economical audio systems with a flat response, even though the system's component parts may be far from linear. The work of Thiele – extended by Small – provides comprehensive details about the response of a driver in an enclosure. All that is needed is to design electronic circuitry to compensate for the non-linear response of the speaker/driver combination.

Advantages of activity

A motional feedback system operates by sensing the speaker cone's motion and feeding this back into the power amp.

Providing negative feedback in this way forces the amplifier to produce a signal that corrects both for amplitude irregularities and the distortion generated by the speaker. The result is an acoustic output which is flat against frequency – even though both speaker and amplifier are operating in a decidedly non-linear fashion.

Motional feedback is not the only way of achieving this. You could use electrical equalisation, for example, but this would not reduce system distortion. In any event, motional feedback is an intellectually satisfying technique using well understood principles.

To produce a correcting signal, the speaker must be fitted with some form of transducer. In this design, I have used dual-coil drivers, one coil of which forms the pickup. As the cone moves, it generates a voltage signal in the coil which is proportional to cone velocity. This signal is then processed and used for correction purposes, see panel.

One major objection to motional feedback is that the feedback loop could try to force the driver beyond its limits. Thi

Fig. 1a). Power amplifier and motion feedback mixer for the subwoofer. Since feedback is derived directly from the voice-coil of the driver, it is possible to produce a very flat response, and reduce distortion.

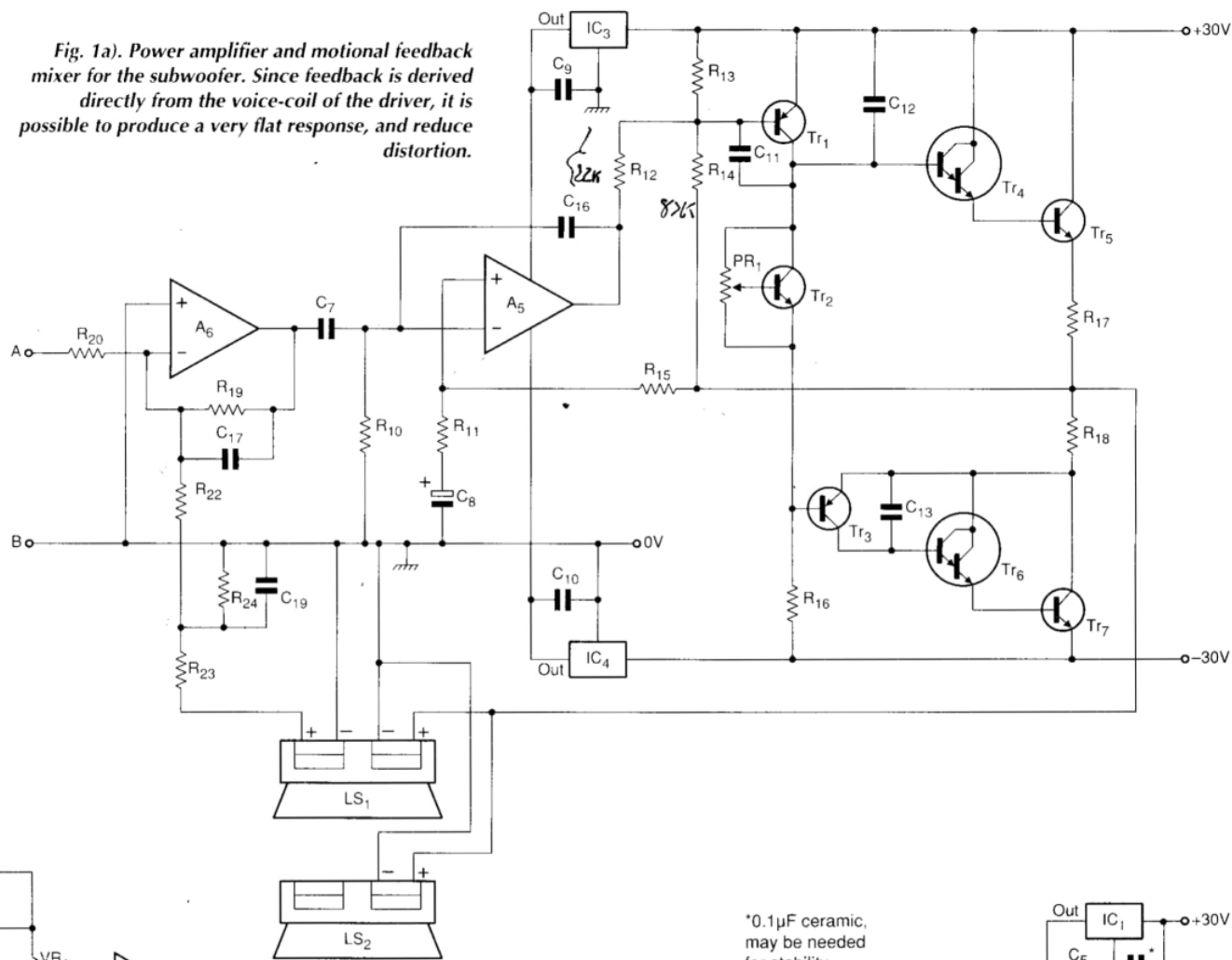


Fig. 1b). Filter chain preceding the subwoofer power amplifier. This section mixes the left and right stereo signals and allows phase to be reversed. It also contains a variable filter allowing the crossover frequency to be set anywhere between 45 and 120Hz. This allows the subwoofer to be adapted to suit most existing hi-fi systems.

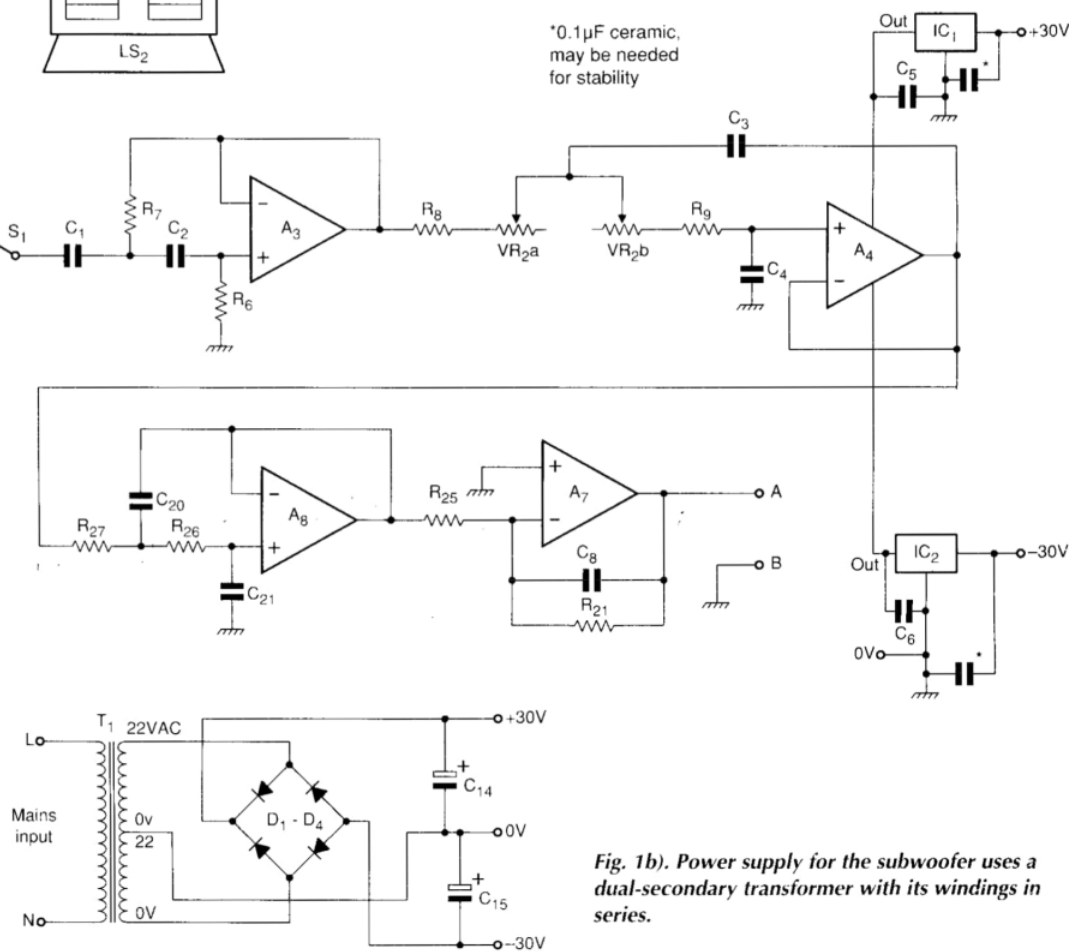
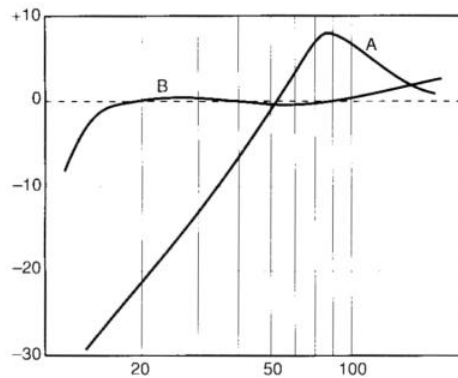


Fig. 1b). Power supply for the subwoofer uses a dual-secondary transformer with its windings in series.

Fig. 2. Curve A is initial system response while curve B is final system response but excluding low-pass filtering. Both curves are measured near field, i.e. 1cm from cone.



problem can be avoided by choosing a sealed box or infinite-baffle enclosure. Use can then be made of the natural roll-off of the driver to ensure that excursion limits cannot be exceeded (see panel).

Using an infinite-baffle enclosure means that the response is essentially that of a second-order filter. To produce a flat response down to dc – even if it were possible electrically – would require infinite cone excursion. To match such a driver, you would need an amplifier of infinite gain since the gain of a high-pass filter is zero at dc. For every octave of base extension, the power and cone excursion required increase fourfold. Obviously this cannot be carried too far. However the system described here has a flat resonance, with a -3dB point at 15Hz.

This frequency is at least an octave below the nominal cut-off of many subwoofer systems. Furthermore when I examined the output acoustically, sine wave distortion was below 2% at 15Hz.

The reason for going so low is mainly to minimise phase shift. There is little musical information on most recordings below 30Hz. Too rapid a roll-off at this frequency produces phase errors between fundamentals and harmonics, leading to a muddy sound. One of the joys of listening to this system is the speed at which bass notes are delivered without the overhang associated with reflex systems.

One problem that has to be considered is how much sound-pressure level, spl, can be generated. Here, we are concerned with a domestic environment. Many more-than-adequate subwoofers use a 10in driver in a reflex cabinet.

Since reflexing is out in this design, I have chosen a pair of 10in drivers, operating in parallel. This has the advantage that both drivers contribute useful output over the entire range whereas a reflex system's vent only contributes output around its resonant frequency. Additionally, the price paid for vent output is a rapid bass roll-off that must worsen the transient response of the system.

For the purposes of analysis, the circuit can be split into three parts. First the power amplifier; the requirements from this element of the circuit include high power output. Furthermore, because the load impedance is around 3Ω, fair-

ly large current swings are required. An extended low frequency response also implies that the power supply will tend to sag under load. Precautions must be taken to make sure that this does not affect sound quality.

Taking into account these factors I make no apology for the use of rugged TO3 output devices, namely Tr_5 and Tr_7 , which are a pair of 2N3055s. These in turn are driven by a pair of Darlington's, Tr_4 and Tr_6 . A quasi-complementary output stage is used with Tr_3 providing the necessary phase inversion for the lower output transistors.

Nested feedback loops

The entire circuit is based on TL074 quad op-amps, one of which is used in the power amplifier. However the low operating voltage and consequent low output voltage swings of this device provide insufficient power. For this reason the amplifier incorporates the idea of nested feedback loops, Fig. 1.

To explain further, the output stage operates from a split 60V power supply. The driver stage is built around Tr_1 . Usually, the output stage biasing voltage is provided by the V_{be} multiplier comprising Tr_2 and PR_1 . Resistors $R_{17,18}$ introduce emitter degeneration in the output stage, stabilising the operating point. Local shunt feedback around both driver and output stage is taken via $R_{12,14}$.

The value of R_{13} has been chosen to produce 0V output for a 0V input from A_5 . This local feedback loop reduces distortion from the output stage to well below 1% before global feedback is applied around the circuit.

Closed-loop gain from A_5 output to the load is approximately five. This allows the op-amp output stage to produce the required voltage swing at the output. An incidental advantage is that the op-amp output sees a relatively high impedance and therefore operates in push-pull, Class A.

Supply voltage for the op-amp is taken from the main power supply through a pair of 15V regulators, $IC_{3,4}$. Capacitors $C_{9,10}$ provide hf decoupling. Op-amp A_5 is the heart of the amplifier. Note that because of the inverting action of the driver/output stage, the inputs are used in the opposite sense.

Input signals are applied to the inverting input and overall feedback to the non inverting. The voltage gain of the amplifier is set by the ratio of R_{15} to R_{11} . Capacitor C_8 reduces the dc gain of the circuit to unity while appearing as a short circuit to ac signals. Resistor R_{10} defines the input impedance of the amplifier.

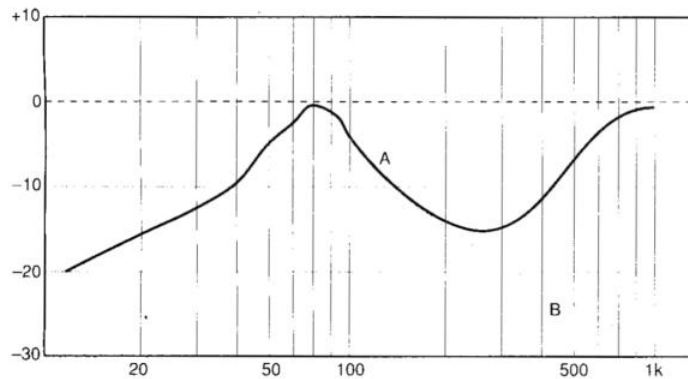
The closed-loop gain needs to be high since most of it is used to equalise the subwoofer and reduce speaker distortion. The other components to be mentioned are mainly concerned with keeping hf stability within the amp. This is the function of $C_{12/13}$ and C_{16} .

It could be argued that most of the parameters of the amplifier just described are well above those required for the circuit function. For example, the slew rate of the op-amps is thousands of times faster than required. Similarly, the distortion level of the amplifier is many times lower than that generated by the speakers. This is simply a reflection on the advancement of commonly available parts. This same level of performance allows supply line voltage rejection ratio of over 100dB, and this is of great importance, as mentioned earlier.

The second aspect of the design, whose overall performance is shown in Fig. 2, involves the manipulation of the pick-up coil voltage to produce motional feedback. Referring to Fig. 1, rather than feed the speaker coil voltage directly into the amplifier's feedback loop, it is fed via the mixer stage built around A_6 . The voice coil output feeds the network comprising $R_{22,24}$ and C_{19} . This is necessary because the voltage follows the impedance curve.

Below 200Hz the output is directly proportional to the velocity of the cone. Above this frequency output rises at

Fig. 3. Curve A is response from the pick-up coil of the loudspeaker and curve B response after high-frequency equalisation.



approximately 6dB/octave. To maintain the velocity curve – not to mention amplifier stability – the output must be suppressed at high frequency. Capacitor C_{19} flattens the curve to a straight line.

From here, the signal feeds the mixer amplifier A_6 , via R_{22} . This is configured as a virtual earth mixer. Feedback resistor R_{19} is shunted by C_{17} , which provides further high-frequency roll-off to the coil signal ensuring that the required response is obtained.

The net result of adding this signal to the amplifier input is that the acoustic response from about 10Hz to 150Hz rises at 6dB/octave. In other words, amplifier output voltage is proportional to cone velocity, and acts as a power differentiator.

To obtain a flat response, the amplifier output needs to become proportional to cone acceleration. Rather than differentiate the feedback signal exactly the same result is obtained by integrating the input signal. This function is carried out by A_7 , which, in conjunction with R_{21} and C_{18} forms the integrator.

At this stage we have produced a flat response speaker system – flat, at least, in the deep bass region. Plotting the response however reveals that the overall response is that of a high-Q low-pass filter. A glance at Fig. 3 reveals that further work needs to be done. The low-pass response is due to the voice coil inductance resonating with the reflected mov-

ing mass.

Rather than complicate the circuitry further, the solution used here is to tame the response by using a low-Q low-pass filter in series with the amplifier. When this has been done the final response is flat within 1.5dB from 15 to 150Hz. The mild penalty to be paid is that the response rolls off at 24dB/octave above 150Hz. Luckily, this is of little consequence in practice since this point occurs at least half an octave – and usually more than an octave – above the roll-off point required by normal speaker systems.

Filtering the input-stage

The main task of the input stage filtering is to extract the bass information from both incoming signals and present this to the power amplifier. In addition, the signal must be manipulated to allow 'seamless' integration of the subwoofer with the existing speakers.

For this design, I decided to drive the sub-woofer directly from the speaker outputs of the existing amplifier. This not only simplifies the design, it is the only rational place to take a signal feed. Once set up the sub-woofer will follow system volume adjustments. This is a particular advantage if, like me, you are always being told turn it down. You can also be assured plenty of drive signal.

Line outputs are rarely standard. The left and right signals

Infinitely baffling

In order to squeeze the maximum possible bass from an infinite-baffle enclosure, the volume has to be carefully calculated. Even motional feedback systems are not immune to the laws of physics. If the enclosure is made too large, the woofer will be driven beyond its excursion limit. If it is too small, maximum power input will not allow full excursion.

In order to calculate the required enclosure volume, Thiele-Small equations are required. When a circular piston is fed with a sine wave, it can be shown that the sound-pressure level generated at 1m into half space, A , is,

$$A(\text{dB}) = 40 \log_{10}(d) + 20 \log_{10}(app) + 40 \log_{10}(f) - 83$$

where d and app are the diameter and peak-to-peak cone excursion respectively, both expressed in mm and f is the frequency of interest.

From the term $40 \log_{10}(f)$, you will see that the available sound-pressure level falls with frequency at 12dB/octave. If the enclosure volume is chosen so that its response lies to the right of A , Fig. 4, then the driver will be protected from excessive excursions. If the response lies to the left of A then the speaker runs the risk of destruction from bass input. Ideal enclosure response coincides with A .

In order to calculate something useful it is essential to examine the efficiency of the driver and relate this to A . Maximum output that a driver can produce in the pass-band is independent of enclosure

size and can be calculated from the following equations, the first of which is for driver efficiency, η_o ,

$$\eta_o = \frac{k \cdot f_o^3 \cdot V_{as}}{Q_{es}}$$

where f_o is free-air resonant frequency, V_{as} is equivalent compliance air volume, Q_{es} is electrical Q and k is 9.64×10^{-10} when V_{as} is expressed in litres. Sound-pressure level in decibels at 1W and 1m distance into half space is,

$$112 + 10 \log_{10}(\eta_o)$$

Maximum sound-pressure level in

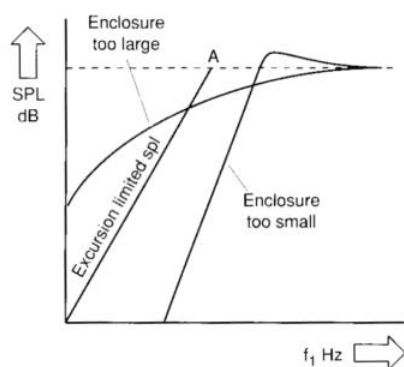


Fig. 4. Excursion sound-pressure level limit rises at 12dB/octave. Speaker responses that intersect the left of A are limited by cone excursion. Right of A are thermally limited but cannot produce maximum sound pressure level. Ideal speaker response coincides with A .

decibels at 1W and 1m distance into half space, B , is

$$P = 112 + 10 \log_{10}(\eta_o) + 10 \log_{10}(p)$$

where p is the available amplifier output in watts continuous.

All drivers mounted in an infinite-baffle enclosure exhibit second-order high-pass filter response whose amplitude, C , is,

$$C = (\omega^4 + (d^2 - 2)\omega^2 + 1)^{-0.5}$$

where ω is f_c/f and d is $1/Q_{tc}$, f_c being the resonant frequency of the driver mounted in the enclosure and Q_{tc} is the Q of the driver in the enclosure. Unfortunately this does not help much because Q_{tc} and f_c are not known until the enclosure volume has been determined. But if you choose f at a low enough frequency, say 1Hz for convenience, then $\omega^4 \gg (d^2 - 2)d^2 + 1$ term. This makes it possible to simplify and rewrite the equation for an approximation of C as, ω^{-2} .

To avoid the excursion limit, the 1Hz response must be $-A+B$ (in dB) down with respect to maximum pass-band sound-pressure level, B . The corresponding amplitude is $10^{((-A+B)/20)}$, which is ω^{-2} .

As f is 1Hz, ω must equal f_c so, $f_c = \omega = 10^{((-A+B)/40)}$. Having obtained f_c , the enclosure volume can be simply calculated from,

$$V_b = V_{as} / ((f_c/f_o)^2 - 1)$$

Calculated volume is slightly conservative, but this no bad thing considering the price of drivers.

from the speaker sockets are passively mixed by R_1 and R_2 . The resulting signal is made available across VR_1 . From here the signal is phase split by $A_{1,2}$.

Avoiding eigentones

Phase splitting is a useful facility for the following reason. When attempting to crossover between speakers and subwoofers, a particular obstacle is avoiding eigentones. At low frequencies, the average room acts as a gigantic speaker cabinet, with resulting resonances, caused by standing waves between parallel walls.

These resonances often occur just where you want the crossover. By judicious use of the controls, you can use phase shift to tame existing boomy speakers or room characteristics. Choice of in or out-of-phase conditions is selected by S_1 of Fig. 1. A small amount of voltage gain is introduced into the phase splitter circuit via $R_{4,5}$. This offsets the gain reduction produced by coil feedback in the power amplifier section. Resistor R_3 couples the op-amps together to provide phase inversion.

Having selected your signal with S_1 , it is then fed into the high-pass filter built around A_3 . This stage defines the lower cut-off point of the system. This is set at 15Hz by the component values chosen. From here the signal is fed into the low pass filter built around A_4 .

Integrating the design

In order to integrate the subwoofer easily, a low-Q second-order filter is used. This stage has a Q of 0.5, critically damped for best transient response.

The -3dB point is continuously variable between 45 and 120Hz. I have yet to find a speaker system which cannot be catered for within this range. Finally, the response of the pick-up coil is modified by the low-pass filter built around A_8 , as described earlier in the text. From here the signal gets fed into the signal integrator A_7 , as already discussed.

I have used separate voltage regulators to power the pream-

plifier section. This may seem like an extravagance but it is a small price to pay for total isolation on the power lines between chips.

On the subject of power supply, Fig. 1, this is completely conventional. Mains voltage is stepped down and full-wave rectified via a bridge, before being smoothed by $C_{14,15}$. The centre tap of the secondaries is used for the 0V line.

Points to watch out for

There are a few points to watch for when implementing the subwoofer. First the cabinet. Initially I intended to build the subwoofer in two enclosures with the intention of siting these below my existing speakers. Since there is no phase information at low frequencies, it is possible, in principle, to the site the subwoofer wherever you choose. In practice the best position is likely to be between the speakers, against the wall. This position will give you an extra 3dB of output for as the system will be driving into quarter space.

Conventional wisdom suggests that corner positions should be avoided as this will tend to emphasise room resonances. Circumstances alter cases, and the extra 3dB of output might be useful.

An advantage of small enclosures, in addition to the improved rigidity, is that they are too small for internal standing wave generation. Since I am not a carpenter and find woodwork a chore I built my cabinets from 15mm chipboard. Fig. 6, available everywhere and in a variety of finishes.

Panel fixing is easiest using Araldite rapid fairly liberally along the seams. The drivers require 230mm diameter cut-outs. They should be mounted on gaskets made from self adhesive draught excluding strip. Before mounting the drivers, fill along the panel seams with filler or silicone sealant to ensure airtightness.

When assembling the electronics ensure that PR_1 is adjusted to short $T_{7,2}$'s base to collector. For obvious reasons it is desirable to set the quiescent current in the output stage before mounting the electronics.

Inductive motional feedback

Inductive pickup is probably the simplest form of motional feedback control. In order to understand its operation, it is necessary to realise that the pickup voltage is proportional to cone velocity. Figure 5 shows the relevant curves. Curve A is the unequalised speaker response and corresponds with the cone acceleration.

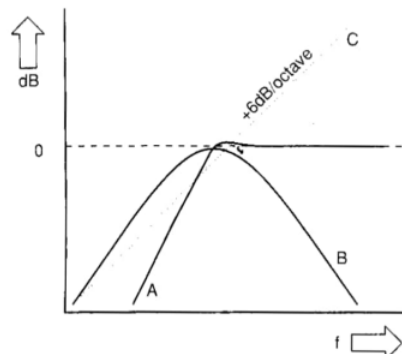


Fig. 5a. Unequalised speaker response, A, resulting velocity curve picked up by the second voice coil, B, and resulting response curve from the speaker, C.

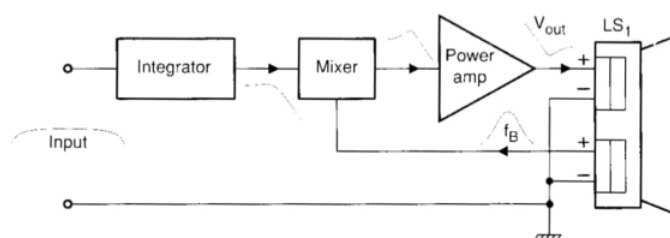


Fig. 5b. Motional feedback using a second coil within the loudspeaker to produce the feedback signal. Error correction signal is introduced via a mixer amplifier.

Curve B is the resulting velocity curve as picked up by the coil. When this voltage is used as negative feedback, the resulting response from the speaker increases with frequency at 6dB/octave, Curve C.

To obtain a flat response, the feedback voltage would need to be proportional to cone acceleration since this is identical to the system response. This would imply differentiating the coil voltage before feeding it back. The alternative, used in this design, is to integrate the incoming signal so that this falls at 6dB/octave. When fed from this signal the overall response of the speaker is flat. There is no difference in system performance either in amplitude or phase response

between differentiating the pickup signal or integrating the input.

Figure 5b shows the basic circuit in block form. Rather than complicate, and possibly destabilise, the amplifier, the feedback is introduced via a mixer amplifier. This is a virtual-earth circuit which effectively adds the input and feedback signals. This is then used to drive the amplifier.

Closed loop gain of the amplifier produces the corrected signal to the speaker. The advantage of motional feedback is that errors in both the enclosure response and in the driver are considerably reduced by negative feedback.

Motional-feedback subwoofer parts

Resistors

Unspecified types are 1% metal film

$R_{1,2}$	47k	2	—
$R_{3/24}$	10k	4	—
R_4	110k	1	—
R_5	100k	1	—
R_6	150k	1	—
R_7	75k	1	—
$R_{8,9}$	15k	2	—
R_{10}	39k	1	—
R_{11}	2k2	1	—
$R_{12,25}$	22k	2	—
R_{13}	330	1	—
$R_{14,15}$	82k	2	—
R_{16}	4k7	1	—
$R_{17,18}$	0.47/3W	2	—
$R_{19,20,26,27}$	560k	4	—
R_{21}	680k	1	—
R_{22}	180k	1	—
R_{23}	43k	1	—
VR_1	4k7 log pot	1	—
VR_2	22k lin dual pot	1	—
PR_1	10k hor. preset	1	—

Capacitors

$C_{1-4,18}$	100n Mylar	4	—
$C_{5,6,9,10}$	100n cer. disc	4	—
C_7	10 μ /50V	1	—
C_8	100 μ F/25V	1	—
$C_{11/12/13/17}$	1nF Mylar	4	—
$C_{14/15}$	6800 μ F/63V	2	—
C_{16}	270pF cer.	1	—
C_{19}	47nF Mylar	1	—
C_{20}	2n7 Mylar	1	—
C_{21}	4n7 Mylar	1	—

Active devices

$IC_{1,3}$	78L15	2	—
$IC_{2,4}$	79L15	2	—
A_{1-8}	TL074	2	—
$Tr_{1,3}$	BC327	2	—
Tr_2	BC337	1	—
$Tr_{4,6}$	BDT65C	2	—
$Tr_{5,7}$	2N3055	2	—
D_{1-4}	1N5408	4	—

Miscellaneous

Heat sink, see text	
TO3 mounting kits	4
Volt DVC250/1, 8 Ω drivers	2
22-0-22V sec. 120VA transformer	
SPST changeover switch	1
Control knobs	2

VEROBOARD.

MB6 CASE.

SKTS.

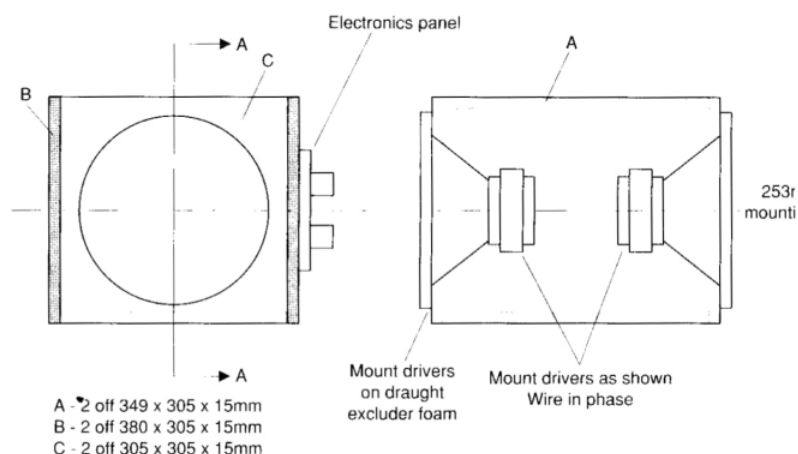


Fig. 6. Subwoofer enclosure details. Since the enclosure is small, it is rigid and inhibits standing waves.

With the speakers disconnected, test that the output is within 50mV or so of 0V. Quiescent current is set up by slowly adjusting PR_1 for a 20mV drop across R_{17} and R_{18} . Although the heat sink gets rather warm under conditions of high drive, I have not found it necessary to use thermal feedback via Tr_2 . But there is no reason why Tr_2 cannot be adhered to the sink.

I mounted the electronics within the enclosure. The output stage requires a large heat sink, of at least 1.5°/W. I used a 120 by 100mm finned sink.

Control panels are always a problem with this type of equipment. I mounted my controls and heat sink on the lid of an MB6 type ABS case which fits into a cut-out in one of the panels. This is secured by six, 30mm M3 screws. Whatever panel you use remember that an air-tight seal is needed.

The drivers are wired up as shown in the Fig. 1a, in parallel and in phase. Ensure that the pickup coil is phased as shown.

Final adjustments

Having set the quiescent current and fastened everything into place, all that remains is to adjust the level and cut-off frequency to suit your system. The best way to start is with the cut-off frequency set high and gain set low. Next, adjust both for the best sound.

Finally, was the effort worth it? Definitely yes. I now get to hear things I have never heard before on my cds. In addition, the clarity and speed at which bass notes are generated and disappear is something of a revelation — after years of reflexed muddiness. ■