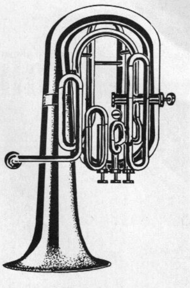


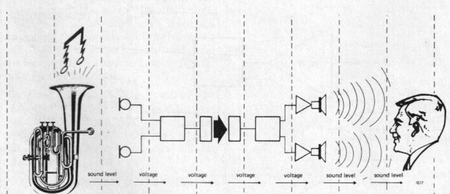
sonant

One of the weaker links in the sound reproduction chain has always been the loudspeaker. The physical problems involved in loudspeaker design are more severe than those occurring elsewhere in the chain — but this is no reason for failing to face them.

A loudspeaker can be defined as a 'more or less linear convertor of electrical to acoustical power'. One can now attempt to make detail improvements to the conversion mechanism (flattening its power-frequency response or reducing its non-linear distortion etc.), or one can look for a different approach to the entire conversion problem (e.g. electrostatic drive!). Still another approach is perhaps more fundamental: combine a power amplifier, a transducer (loudspeaker) and an error-correcting system to form a 'linear convertor of electrical voltage to acoustical amplitude'.
This approach is sufficiently new to merit a new name. We suggest the name 'Sonant'. An example of such a design is the 22RH532 Electronic, recently introduced by Philips (see the article 'Philips Sonant' elsewhere in this issue).
The introduction of a low-distortion voltage-to-sound convertor would mean that all the links of the sound reproduction chain reached an acceptable level of performance. It seems desirable to do a little rethinking about the whole sound reproduction process, in the light of the changed situation.
Figure 1 shows a block diagram of a reproduction chain using sonants. The figure also shows the influence of various links on the frequency characteristic, stereo balance, stereo width and '0 dB' reference level. The influence on the level is important when a loudness-contour correction is to be used.
The reproduction chain includes a 'presonant'. This is the link which applies the corrections necessary to attain the desired overall characteristics in the face of errors elsewhere in the chain. Essentially this function can be performed by a normal control-amplifier. Optimum convenience of operation, however, requires some rather different control arrangements:

– *tonal balance.* Assuming that the recording staff have done their job properly, it should only be necessary to make tonal balance adjustments which will offset the effects of the listening-room acoustics on the frequency characteristic. A once-only adjustment of preset controls is then sufficient.
– *stereo balance.* This is somewhat dependent on the positions of the sonants in the room. A preset adjustment will once again suffice.
– *stereo width.* This is also somewhat dependent on the sonant positions. It is furthermore very dependent on the personal taste of the recording engineer, so that a control providing some range of variation is desirable. Experience has taught that four switched conditions are adequate: mono — 'half stereo' — stereo — 'wide stereo'.
– *level.* The requirements to be met by the contour compensation depend on the actual reproduction level at any given moment. However, the '0 dB level' may be 100mV for example from a disc preamplifier, 300mV from an FM tuner and 1000mV from a tape recorder. It is therefore necessary to match levels from the various inputs. The final requirement is for a *volume control* fitted with the loudness-contour compensation circuit. Once again experience teaches that the number of settings can be drastically pruned. What about: background — moderate — normal music reproduction — shattering?

It will be clear that most of the front panel controls disappear. These were in the past used mainly to compensate for imperfections in the recording- or playback-equipment. The operating procedure now becomes much simpler: 'select programme', 'select loudness' and (for the time being?) 'select stereo width'. The well-known complexities of operating audio equipment have now been replaced by the ease of operation of a TV set (once properly adjusted!).
Design work is at the moment in progress on a complete elektor-presonant, along the above lines.



Aspect	source	microphones	mixing console	transmission path	presonant	sonant(s)	room	listener
Frequency characteristic		correct	variable	flat	see text	flat	varied	
stereo balance		–	correct	–	"	–	"	
stereo width		–	variable	–	"	–	"	
level		–	correct	varied	"	constant	"	

philips sonant

In the article "sonant", elsewhere in this issue, a definition is given of a new type of "active" sound-reproducer design, in which the radiated sound (pressure) level is linearly dependent upon the electrical signal (voltage) applied to the input. This approach differs fundamentally from the classic 'passive' loudspeaker system design, where electrical input power is converted – with limited linearity – into radiated acoustical power. Passive systems are either particularly poor in low-frequency linearity or else they are very bulky. On the other hand, active systems with good low-frequency performance can be quite compact.
A practical realisation of the sonant-principle is to be found in the recently presented Philips 22RH532 "Electronic". This article describes a few aspects of the way in which this sonant operates.

It is well known that the performance of most loudspeakers is far from ideal. The designer of a complete system is invariably forced to make compromises between size, shape, efficiency, directionality, distortion, amplitude response ... and price. The attempt to achieve really good performance via the classic approach nearly always fails. The performance in the bass register, in particular, only becomes acceptable as the enclosure or system becomes too large for domestic listening-room convenience. The possibility now suggests itself of circumventing the mechanical-acoustical problems by electronic means. One possible approach is to make use of a power amplifier with a negative output impedance, which can compensate for some of the drive-unit's DC resistance and so obtain better control of the coil movement. Examples of this approach are the "Electronic loudspeaker" (Elektor sonant) and a commercially-available system called "Servosound".
The difficulty with this approach is that the systems have to use current-dependent positive feedback around the power amplifier, which means that the optimum adjustment is fairly critical. A different approach becomes possible when the loudspeaker is equipped with some form of motion-sensing device. One can then apply overall negative feedback to the complete amplifier-driver system: so-called 'motional feedback'. This approach is what has been realised in the woofer channel of the Philips sonant.
It should be noted that neither of these ideas is new. The negative-output-impedance approach was described as far back as 1940, by nobody less than Harry F. Olson. The motional feedback approach using an accelerometer was first tried at least ten years ago. The difficulty was to design a reliable system, not too expensive – and suitable for mass production!

Principle

Figure 1 is a block diagram of the complete sonant. It is clear that motional feedback is only applied to the woofer channel, the classic approach being adopted for the mid-range and the treble. In contrast to the "electronic loudspeaker", this design employs a single amplifier for the mid- and treble-ranges, in conjunction with a passive dividing network.
The bass register (35–500 Hz) is handled by the feedback system. This subchannel consists of a 40 watt amplifier, the bass transducer with accelerometer and a feedback network. The feedback network actually contains an impedance-matching circuit for the piezo-electric acceleration-pickup, a preset gain control and a set of stabilising filters.
Before examining the system in detail it will be interesting to see what results are actually achieved in practice.

Results

The volume of the actual woofer enclosure is only 15 litres, the overall outside dimensions being 28.5 x 38 x 22cm. A copy of Elektor, folded out flat, is therefore rather larger than the system's front panel! The complete electronic 'works' (amplifiers, filters, power supply) are mounted inside the 'box', as can be seen in the photograph (figure 2).
In a small enclosure such as this the fundamental resonant frequency is about 80 Hz, but the feedback arrangement prevents this having any effect on the amplitude response. This response-characteristic is sketched in figure 3. The 3dB rolloff-points are shown at about 35 Hz and 20 KHz. The dotted curve shows what happens when the feedback is made inoperative – more than an octave of bass response is lost!
A further advantage of the use of feedback is that the distortion is reduced. This is shown in figure 4. One should bear in mind, at this point, that the subjective effect of loudspeaker distortion is quite different to that of the usual kind of distortion in the (power) amplifier. A good amplifier is substantially free of perceptible distortion until it is actually

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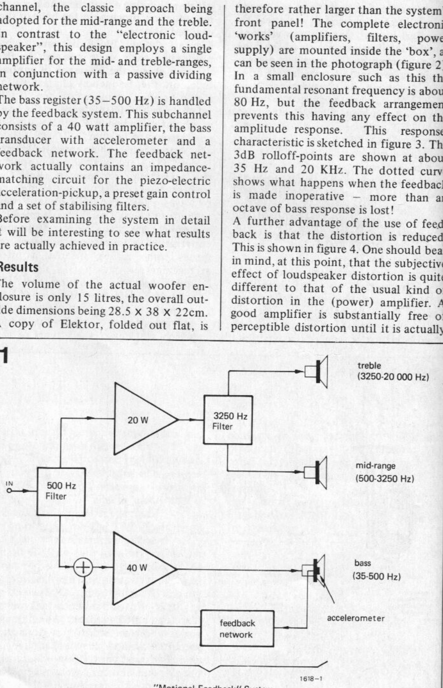


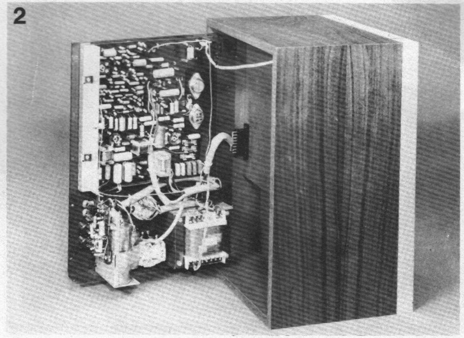
"Motional Feedback" System

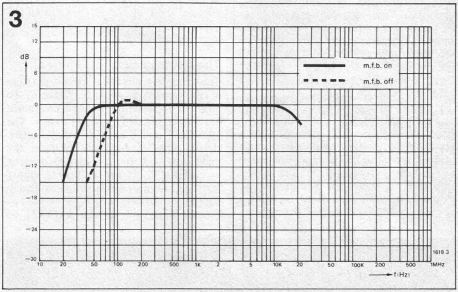
Figure 1. Block diagram of the Philips sonant. Frequencies up to 500 Hz are reproduced via a motional-feedback woofer system.

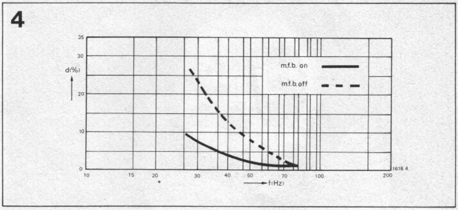
Figure 2. The sonant. The electronic 'works' are mounted on the inside of the hinged rear panel, which doubles as a heat sink.

Figure 3. Amplitude-frequency response of the sonant. The solid line shows the response with motional feedback in operation; the dashed line shows what would happen if there were no motional feedback.

Figure 4. Distortion characteristic at nominal output. The solid line indicates the distortion with motional feedback operative. Removing the feedback results in the much higher dashed curve values. See the text for an interpretation of these curves!



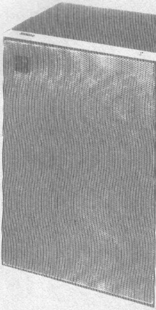


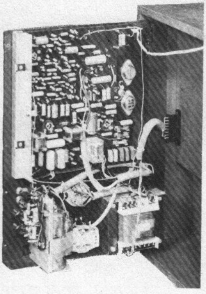


overdriven, when it suddenly starts to produce sharp-edged waveforms containing musically-unpleasant high order components. Loudspeaker distortion at low frequencies, on the other hand, consists mainly of third harmonics. This low-order distortion merely disturbs the balance between fundamental and naturally-present harmonics in the instrument being reproduced, causing relatively acceptable 'colouration' of the sound.
The curve in figure 4 can be viewed as follows. Assume that the loudspeaker is operating without feedback and is being nominally fully driven with a pure 30 Hz tone. The delivered output will consist of 76% fundamental (30 Hz) and some 24% third harmonic (90 Hz). The loudspeaker with feedback will (under equivalent conditions) produce 92% fundamental and only about 8% third harmonic, so that the reproduction will sound much less 'coloured'. At still higher (overdrive) levels the effect will become still more pronounced: operated without feedback the loudspeaker will produce as much as 80 to 90% distortion, reducing to about 30% with feedback operative. The operation of the feedback is then to increase the amount of fundamental produced from as low as 10% to some 70% of the total.

The great increase in the level of the fundamental inevitably makes the reproduction more natural-sounding, more 'realistic'. One then overhears remarks like: "It is as if there is no longer a loudspeaker getting in the way".

It is to be expected that this performance can only be obtained at a price. The price to be paid was already mentioned in an earlier article ('electronic loudspeaker' Elektor no. 1) – it is simply that the amount of sound output obtainable at 'flat' power-response is considerably lower in the feedback case. Maintaining the loudspeaker's amplitude response flat below the point at which the basic system naturally rolls off implies the application of 'brute force' by the feedback drive. Extending the response by an octave would in fact require the amplifier rating to be increased more than 10 times. Any attempt to actually do this is of course to risk destruction of the driver – which in the Philips case is rated (as is the associated power amplifier) at 40 watts. The seemingly-obvious assumption that the power-response should be flat leads to the conclusion that the amplifier, at nominal output of the system above the natural rolloff frequency, always operates well below the level of which it is capable. Fortunately the assumption is incorrect!
The continuous line in figure 5 is a contour for the maximum level encountered in 'typical recordings' as a function of the frequency. It was derived from measurements performed on a large number of disc records. The dashed line is a contour which applies to one or two extreme-case recordings (e.g. Decca's 'Zarathustra'). It may be pointed out that the extremes below 100 Hz are





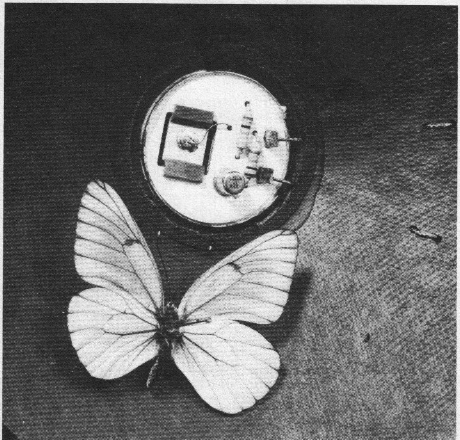
rarely encountered ('Zarathustra' or Saint-Saëns 'Organ Symphony'); but that the treble-range extremes are more common (e.g. percussion and synthesizer-effects in pop-recordings). This higher contour can be exceeded by 6 to 10dB during momentary signal peaks, mainly in the mid-range up to about 3 KHz.
The right-hand vertical scale in figure 5 has been chosen to represent fairly loud music reproduction (as typically encountered in monitoring rooms during classical recording). The dash-dot contour indicates (to this scale) the maximum level of which the Philips sonant is capable in a fair-sized domestic listening room. It will be clear that the system is capable of handling the 'peak programme level' discussed above at all frequencies higher than its 35 Hz amplitude-response rolloff point.

The electronics

The complete electrical circuit diagram of the sonant is given in figure 6. To improve the readability of this diagram it has been divided into sections. The sections A and B are the 'woofer' drive circuit and its feedback system; the mid-range and treble channel consists of sections C, D and E; power supply section G, finally, is controlled by signal-dependent shutdown F.
The first part of section A (T421 to T423) is an active bandpass filter with cutoff frequencies of approximately 35 Hz and 500 Hz. The motional feedback signal is injected via C 506. The operating principles of such filters are (or should be) well enough known. The remainder of section A is a normal class B power amplifier, with an operating bandwidth of 5 Hz to 2 KHz and rated at 40 watts. It meets all the requirements of this application. One or two design details may be worth noting:

- the differential input pair T 424 and T 425, necessary to prevent 'disagreement' between the various feedback paths (C 506/R 603/T 423, R 611/C 504 and R 623/C 509/R 622/C 511/R 619).
- the extensive filtering in the above-mentioned feedback paths, which are designed to optimise the overall amplitude- and phase-characteristics of the system as a whole.
- the application of power-Darlingtons in the output stage.
- diode D 456, which 'clamps' the base-voltage of T 430 whenever this attempts to exceed the supply-rail voltage as a result of the 'bootstrapping' via C 513.

In section B a dotted rectangle is shown enclosing the drive-unit itself, the accelerometer and an impedance-matching stage. The pickup device proper, plus an FET and two resistors, are mounted on a small PC board glued to the leading edge of the drive-coil former (figure 7). The pickup device is a small ceramic plate, suspended in an opening of the PC board by means of rubber blocks. The voltage delivered by the pickup is proportional to the force it experiences, which in turn ($f = M.a$) is proportional to the acceleration. The moving mass is in fact largely due to the solder droplets – so that these must be carefully controlled in size! Inevitable production tolerances can be corrected by means of preset potentiometer R 654. The ceramic plate performs best when looking into an extremely high impedance, which is the reason for using an FET. Installing this FET beside the pickup, rather than on the main PC board, avoids problems with hum and instability.
The circuit around T 433 combines the functions of maintaining the FET at the correct operating point and of extracting the output signal for further processing. The remainder of section B is a filter-amplifier. It delivers a signal strong enough for injection into the main channel, at the same time arranging for unconditional stability of the feedback system.
The power amplifier circuit for the mid-range and treble loudspeakers (section C) closely resembles that in the woofer channel. In this case however the 500 Hz high-pass filter is built up around the first transistor of the amplifier itself T 439). The output stage is biased to a quite high value of standing current (about 200mA), i.e. in class AB, to eliminate any possibility of crossover distor-



amplifier passes through the dividing network (section D) to drive the loudspeakers of section E.

The circuit of section F is a kind of automatic supply switch. With mains applied and the on/off switch depressed there will be DC on the '+2' and '+3' supply rails. An input signal delivered to the sonant at a level higher than about 1mV will, after being amplified and rectified, cause the Schmitt trigger in section F to change state and so pull down relay S 402. This turns on the feed to the power amplifiers, within 1 second of the arrival of a signal. If the signal is interrupted for more than about 3 minutes, the circuit assumes that the sonant is no longer required and shuts down the power amplifiers.

The actual power supply circuit (section G) is fairly standard. Special attention has been paid to the feed for critical circuits ('+3'): T 451/T 452 provide extra smoothing by what is in fact gyrator-action. An additional effect of this arrangement is to cause the feed voltage to rise slowly after application of the mains, thus eliminating 'thump'. The circuit including R 726, R 727 and D 471 is an interesting design-gimmick: the supply indicator lamp L 414 is arranged to glow dimly so long as the sonant is dormant with mains 'on', via R 726 and R 727. When the amplifier feed is turned on, however, D 471 effectively short-circuits R 726, so that the lamp glows at normal brightness. This provides, if nothing else, a 'standby' condition for the lamp! However, our most recent information indicates that this feature has been omitted from the latest models: the lamp switches on and off together with the amplifiers.

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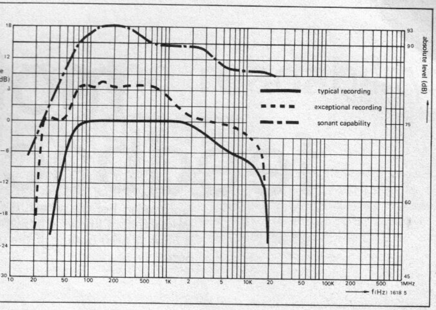

Figure 5. Contours of maximum level against frequency. The solid curve indicates the levels encountered in typical music recordings; the dashed curve shows higher levels occasionally encountered. The right-hand scale is chosen to represent fairly loud reproduction (classical monitoring); the dash-dot line shows the sonant's peak programme capability to the same scale.

Figure 6. Complete circuit diagram of the sonant. 'A' and 'B' form the motional feedback woofer-channel; 'C', 'D' and 'E' are sections of the mid-range/treble channel; 'F' is the signal-sensing power-shutdown; 'G' is the power supply section.

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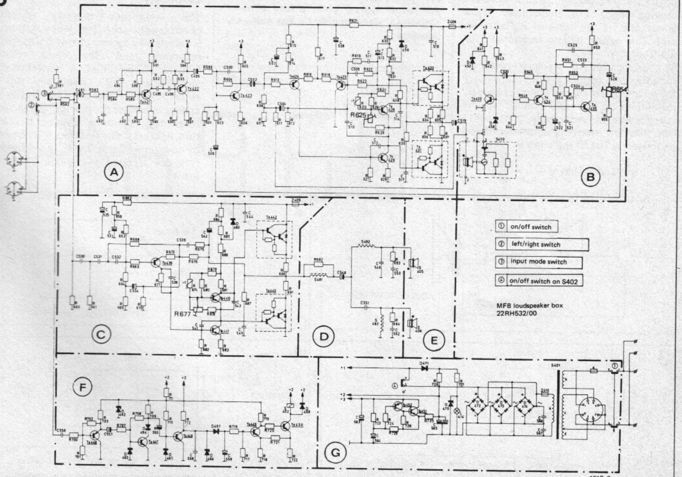

motional feedback

Many people are quite satisfied with the musical pleasure they can obtain from a well-designed reproducer. For the others, the pleasure is heightened by the intellectual satisfaction of knowing how the thing works and why it sounds the way it does. A brief outline of the theory of motional feedback is therefore in order.

A good starting point is to take a look at the arrangement chosen for the 22RH532 'Electronic': a loudspeaker driver fitted with an accelerometer, mounted in a stiff airtight enclosure.
This arrangement is shown in the block diagram of figure 1. The acceleration of the cone is measured, the derived voltage being applied as negative feedback. A short calculation will show that this is the best approach. The radiated acoustic power (P_a) is given by:

$$P_a = u^2 \cdot R_a$$

where u is the 'particle velocity' (equal to the cone velocity) and R_a is the 'radiation resistance' (real part of the air-load on the moving cone). For frequencies below about 500 Hz the value of R_a increases with the square of the frequency (figure 2).
The application of acceleration-dependent negative feedback will tend to keep the acceleration of the cone (a) linearly dependent on the input voltage (v_i) and independent of frequency.
The relation between acceleration (a) and velocity (u) of the cone is:

$$u = a \cdot t \text{ so that } u \sim \frac{a}{\omega} \sim \frac{v_i}{\omega}$$

Figure 1. Block diagram of a motional feedback system making use of an accelerometer.

Figure 2. The radiation resistance (R_a) of the air-load on a typical moving-coil loudspeaker in box increases (up to about 500 Hz) with ω^2. Since it is the radiated power that is proportional to R_a, the output at constant cone velocity would rise at 6dB/octave.

Figure 3. Constant acceleration means a cone velocity that is inversely proportional to ω. This velocity is proportional to the square root of the radiated power. So u can be plotted as a −3dB/octave slope.

Figure 4. The radiated acoustical power is proportional to the product of radiation resistance and velocity squared. Combination of figure 2 with twice figure 3 yields a total slope of 0dB/octave below 500 Hz i.e. flat frequency response!

Since a is independent of ω in this case, the velocity will be inversely proportional to frequency (figure 3).
This leads to:

$$P_a = u^2 \cdot R_a = C \cdot \frac{1}{\omega^2} \cdot \omega^2 \cdot v_i^2$$

where C is a constant. (If we ignore box dimension- and room position-effects!) The relationship between radiated sound power and input voltage is therefore independent of frequency (figure 4). In other words, the amplitude-frequency response characteristic is flat.

Summary

The amplitude-frequency response characteristic (the 'frequency response') is determined by two terms: the radiation resistance and the cone velocity (squared). The radiation resistance rises quadratically with frequency (up to about 500 Hz, figure 2). The velocity decreases in inverse proportion to the rising frequency (assuming constant acceleration, figure 3). The final result is obtained by combining figure 2 with twice figure 3 (velocity squared!) — which yields figure 4. K

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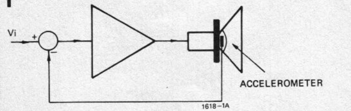

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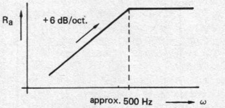

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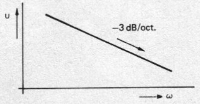

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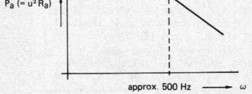


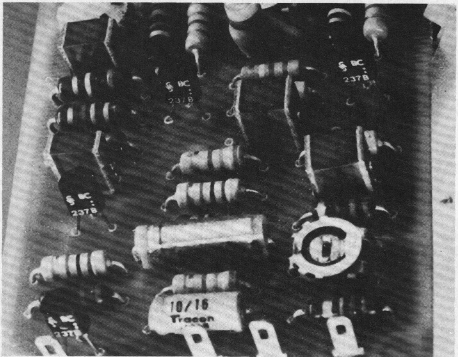
elektor „sonant" loudspeaker system

An earlier article explained the operating principle of the electronic loudspeaker. It was shown that this approach enables a woofer in a relatively small enclosure to provide really good reproduction quality from 40 Hz upwards.
The second step is to describe a complete system, using active crossover networks, based upon the electronic loudspeaker used as a woofer. The system provides unusually good reproduction quality for its modest size and low cost.

An electronic woofer can be designed to have a flat frequency response curve beginning at 40 Hz – or an even lower frequency if desired. This flat response will normally be maintained up to about 400 Hz. Above this frequency it will be necessary to apply additional corrections, which are dependent on the loudspeaker used, the size and shape of the enclosure (and the loudspeaker's position on the front) – and even the position of the enclosure relative to room boundaries.
The simplest design choice would therefore seem to be an application of the electronic loudspeaker to the woofer-range, with reproduction of the mid- and treble ranges by normal units.
Such an approach makes it very desirable to use a separate driving amplifier for the woofer, with crossover networks ahead of this amplifier. The mid- and treble ranges can then be handled by one or two smaller amplifiers, for two- or three-way systems respectively.

Two-way system

The so-called two-way system provides a simple solution which can nonetheless produce excellent results. The block diagram for this is shown in figure 1. The 'crossover' frequency chosen is 340 Hz. The electronic woofer reproduces the range 40-340 Hz, while a separate 10-watt amplifier takes care of the remainder of the frequency range.
Two preset potentiometers, inserted in the circuit ahead of the crossover networks, enable the correct loudness balance to be set up. The adjustment is best done by ear, preferably with the system installed in its working location. In this way the inevitable differences between the efficiencies of the various loudspeaker-drivers used can be equalised.
Care must be taken to connect the two drivers in the correct phase-relationship. The woofer is operated with the 'plus' terminal – the terminal which, when driven positive, causes the cone to move outwards – connected to the 'hot' power-amplifier terminal. The mid-high-range driver is then connected with its 'plus' terminal to chassis, i.e. the 'wrong way around'. On many loudspeakers the 'plus' terminal is identified by the maker, by means, for example, of a red dot.



Three-way system

The three-way system is built up according to the block diagram given in figure 2. The electronic woofer operates as before, for frequencies up to 340 Hz. A second amplifier handles the range 340 to 4800 Hz., while a third unit operates at frequencies above 4800 Hz. This may seem to be the expensive way of doing things; but in practice the additional lower-power amplifiers often turn out to be less expensive than conventional crossover networks. In any case the results are usually better, while the level-balance between the high- mid- and bass-loudspeakers can be very conveniently obtained by adjustment of the three preset potentiometers.
Once again, attention must be paid to correct relative phasing of the drive-units. This is also shown in the block diagram.

The crossover networks

The crossover networks for the two-way system are built up according to figure 3; the three-way circuit diagram is given in figure 4. The circuits are simple, to the point of being primitive, being cascades of passive RC-networks, buffered by means of emitter-followers. Reliable, stable and predictable. Furthermore the two arrangements are so similar that both can use the same printed circuit board. For the two-way system all that is required is a single printed circuit board, with the component layout shown in figure 6.

The three-way system circuit requires two of the boards. One of these is laid out according to figure 7, for the mid-range and treble channels. The other is laid out according to figure 8, with the circuits for bass reproduction via the electronic loudspeaker. This latter p.c. board also includes an input buffer (T_1).

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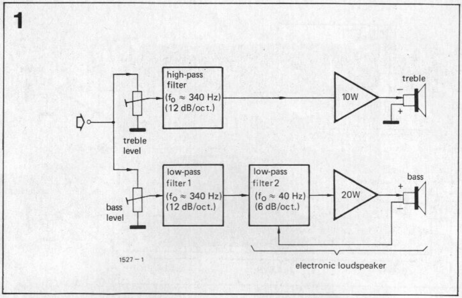

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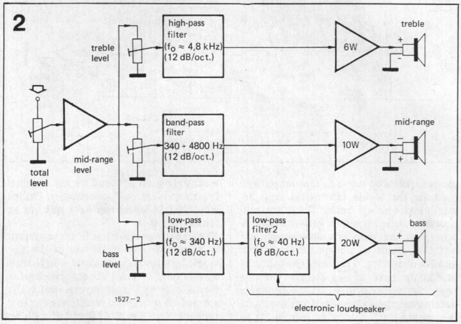

Figure 1. Block diagram of the two-way system.

Figure 2. Block diagram of the three-way system.

Figure 3. Circuit diagram of the two-way system. The motional feedback in the bass-channel is adjusted by means of P_3; the level-adjustment presets P_1 and P_2 enable the correct loudness-balance to be obtained between bass and treble channels.

Parts list for figures 3 and 6

Resistors:

R_1	=	220K
R_2, R_6	=	100K
R_3, R_5, R_{11}, R_{13}	=	1K
R_4, R_9, R_{12}, R_{16}	=	6K8
R_7	=	56K
R_8	=	3K3
R_{10}	=	4K7
R_{14}, R_{18}	=	22K
R_{15}	=	2K2
R_{17}	=	1K8
R_{19}	=	1 Ω
P_1, P_2	=	5K (preset)
P_3	=	1K (preset)

Capacitors:

C_1	=	6n8
C_2, C_5	=	470n
C_3	=	330n
C_4	=	100n
C_6	=	180n
C_7	=	10µ/16V

Transistors T_1 to T_4: TUN

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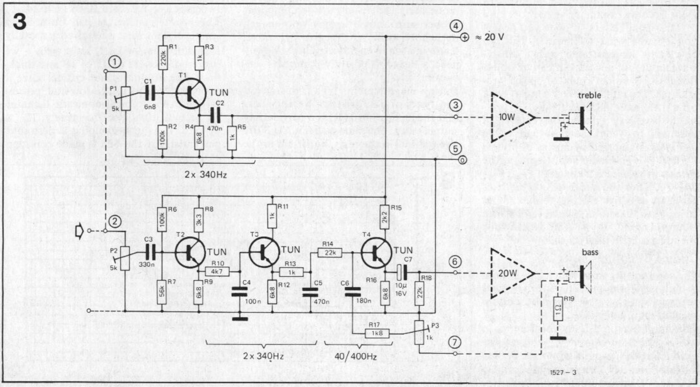


Parts list for figures 4, 7 and 8

Resistors:

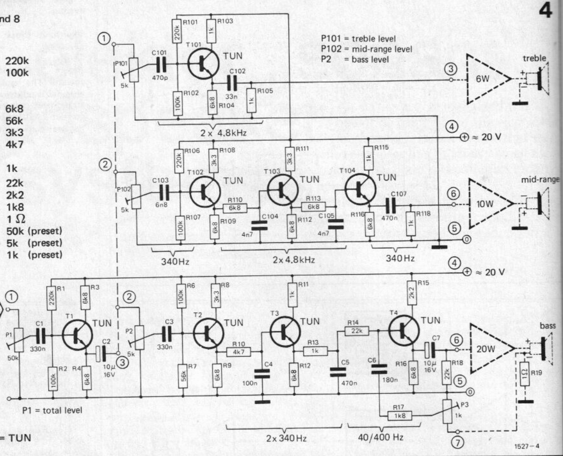
R_1, R_{101}, R_{106}	= 220k
$R_2, R_6, R_{102}, R_{107}$	= 100k
$R_3, R_4, R_9, R_{12}, R_{16}, R_{104}, R_{109}, R_{110}, R_{112}, R_{113}, R_{116}$	= 6k8
R_7	= 56k
R_8, R_{108}, R_{111}	= 3k3
R_{10}	= 4k7
$R_{11}, R_{13}, R_{103}, R_{105}, R_{115}, R_{118}$	= 1k
R_{14}, R_{18}	= 22k
R_{15}	= 2k2
R_{17}	= 1k8
R_{19}	= 1 Ω
P_1	= 50k (preset)
P_2, P_{101}, P_{102}	= 5k (preset)
P_3	= 1k (preset)

Capacitors:

C_1, C_3	= 330n
C_2, C_7	= 10µ/16V
C_4	= 100n
C_5, C_{107}	= 470n
C_6	= 180n
C_{101}	= 470p
C_{102}	= 33n
C_{103}	= 6n8
C_{104}, C_{105}	= 4n7

Transistors:

T_1 to T_4, T_{101} to T_{104} = TUN



Setting-up procedure

The adjustment of the electronic woofer – in either arrangement – is made by means of the preset potentiometer P_3. This is slowly turned up from zero position to the point at which the system starts to oscillate (howl), and then turned back until oscillation just ceases.

Some loudspeakers have inferior magnetic systems in which the degree of electro-magnetic coupling varies during the drive-coil throw. An adjustment made as above can then give rise to a kind of 'after-pong' effect (if once you hear it you'll know what we mean!). This can be clearly demonstrated with a square-wave input; but of course it can be objectionable on some kinds of music programme. The remedy is very simple: back off a little more on preset P_3.

In the two-way system, the balance of loudness between the bass and treble channels is adjusted by means of presets P_1 and P_2. The same adjustment in the three-way system is made with P_{101}, P_{102} and P_2. Turn the preset which controls the least sensitive channel to maximum, then adjust the other(s) until the balance 'sounds right'. It is well worthwhile spending a little time on this.

Choice of components

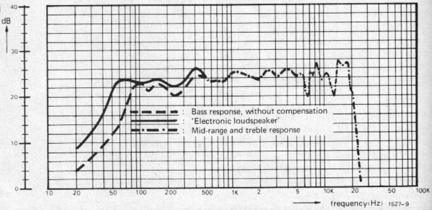
The components from which this system is built up – the amplifiers, drivers and the enclosure – have to meet certain specific requirements.

The amplifier used with the electronic woofer must be completely and unconditionally stable, even when its output is short-circuited. The prototype systems used the 'Equa amplifier' previously described (Elektor no. 1). The amplifiers used in the treble channel(s) may be small high-quality units. They are not normally called upon to deliver as much power as the woofer's drive-amplifier, but they must of course be free from audible distortion. The bass driver should preferably have a high-quality electromagnetic motor system. This ultimately determines the system performance and the maximum obtainable output. It is also desirable that the cone-suspension be moderately stiff; the supercompliant rubber surrounds on some woofers can misbehave quite seriously at high drive-levels in a small enclosure. One will then need a larger – possibly damped – enclosure.

During measurements on various makes and types of bass drivers it became clear that the Philips 9710 (M) is a particularly suitable unit. The same maker's AD 3701 also did well in the tests, but it has in the meantime been replaced by the nominally almost identical (according to Philips) AD 7061 M which we have not yet extensively tested.

The treble reproduction in these systems is 'standard' – without use of the electronic loudspeaker principle – so that the dirvers have to meet normal hifi-requirements. Among the several units that seemed to give good results are the mid-range drivers Kef B 110 and the Philips AD 5060/Sq, along with the 'dome' tweeters Kef T 27 and Heco PCH 24.

The woofer enclosure can be a fairly simple design. What is required is a totally closed box having fairly solid walls – we suggest chipboard of 15 or 18 mm thickness. The volume is not critical since it really only affects the low-end power-response. For typical domestic listening 15 litres is usually sufficient (e.g. 12″ x 9″ x 8″). A little damping is desirable, particularly if the box is made consider-



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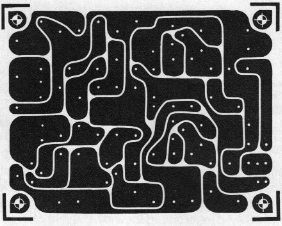

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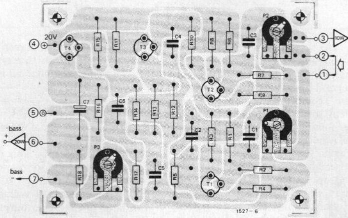

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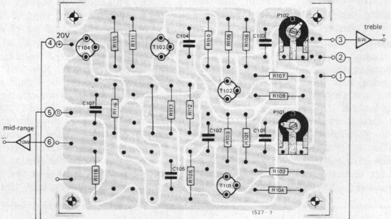

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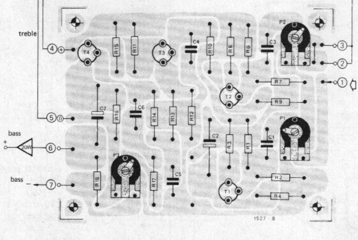

Figure 4. Circuit diagram of the three-way system. P_3 is once again used to correctly set up the bass-channel motional feedback; loudness-balancing is done by adjusting P_2, P_{101} and P_{102}. P_1 provides an additional total-level adjustment, which can be convenient when setting up a stereo pair.

Figure 5. The universal printed circuit layout for all filter circuits.

Figure 6. Components layout for the single p.c. board used with the two-way system according to figure 3.

Figure 7. Component layout for the mid-range and tweeter channels of the three-way system (figure 4).

Figure 8. Component layout for the bass-channel and buffer stage as used in the circuit of figure 4.

Figure 9. Result of measurements (total radiated power) on a typical three-way system.

ably larger than 15 litres. This will usually arise if one wants an extended power-response. Damping may in any case be needed with drivers having limited magnetflux (to help eliminate the 'after-pong' effect). A 20 mm thick pad of glasswool, mounted by means of laths at a random angle through the centre of the enclosed volume – not parallel or close to the walls – will do the trick.

Conclusion

Application of the electronic loudspeaker-compensation described here – actually a form of 'motional feedback' – can enable excellent results to be obtained with a relatively small woofer-enclosure. The curves in figure 9 are the results of measurements made on a typical system, with and without the compensation operative.

The only real objection to this approach is the fact that it requires fairly critical adjustment for best results. This difficulty would be considerably lessened if a bass-driver fitted with a properly-designed and reliable feedback-transducer were to become generally available. The transducer could deliver a signal proportional to the cone's displacement, to its velocity, or – preferably – to its acceleration. One recently-introduced commercial system (Philips) is designed around just such a woofer-accelerometer combination.

As a final remark we note that a loudspeaker operating with motional feedback forms part of a control-loop which includes the power amplifier. It is obviously desirable to install this amplifier in the loudspeaker cabinet, or the fact that the feedback signal is carried by the same leads which deliver the drive power means that the system will require re-adjustment every time the lead length is changed! ◾