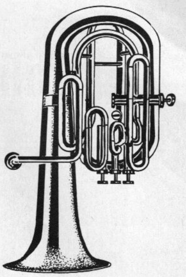


sonant

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'.

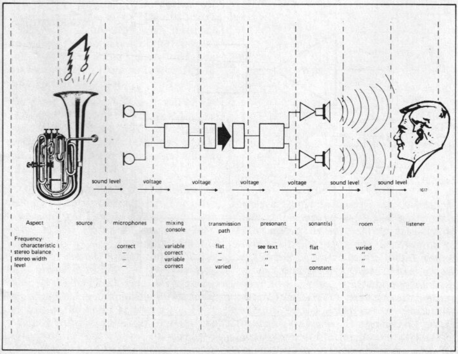
This approach is sufficiently new to merit a new name. We suggest the name 'Sonant'. An example of such a design is the 22RH532 Electronic, recently introduced by Philips (see the article 'Philips Sonant' elsewhere in this issue). The introduction of a low-distortion voltage-to-sound convertor would 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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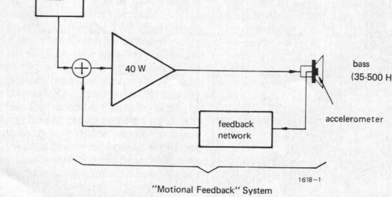

Figure 1. Block diagram of the Philips sonant. Frequencies up to 500 Hz are reproduced via a motional-feedback woofer system.

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!

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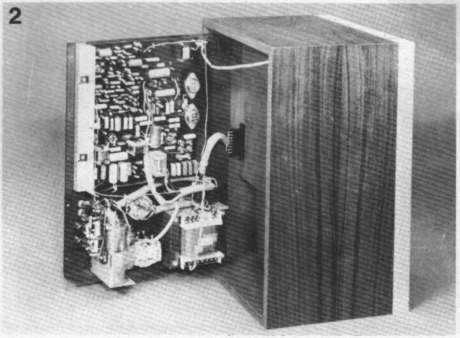

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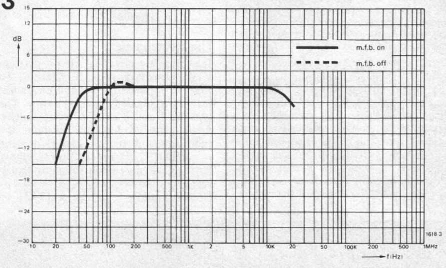

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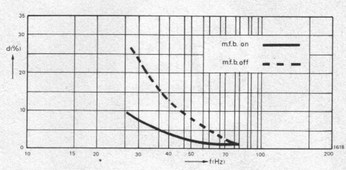


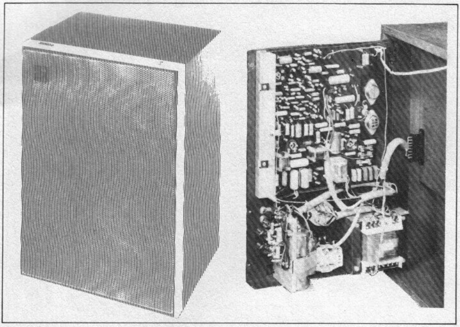
overdriven, when it suddenly starts to produce sharp-edged waveforms containing musically-unpleasant high order components. Loudspeaker distortion at low frequencies, on the other hand, consists mainly of third harmonics. This low-order distortion merely disturbs the balance between fundamental and naturally-present 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.

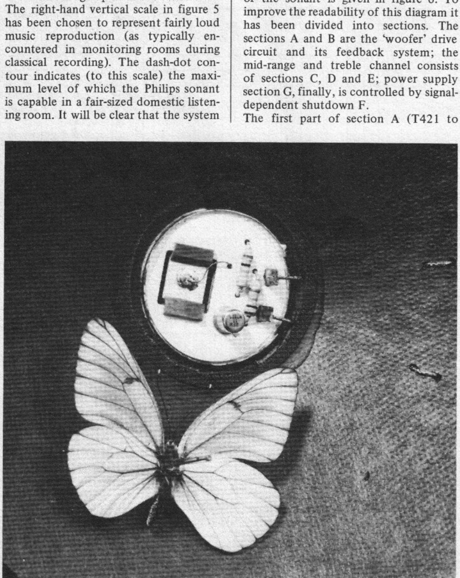
The first part of section A (T421 to T423) is an active bandpass filter with cutoff frequencies of approximately 35 Hz and 500 Hz. The motional feedback signal is injected via C 506. The operating principles of such filters are (or should be) well enough known. The remainder of section A is a normal 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. ◾

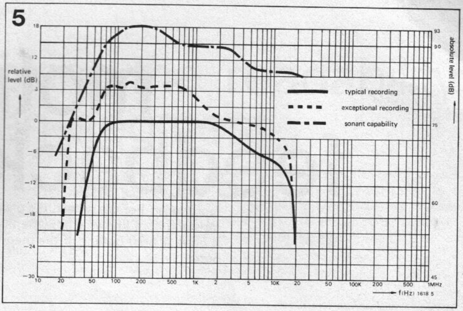


Figure 5. Contours of maximum level against frequency. The solid curve indicates the levels encountered in typical music recordings; the dashed curve shows higher levels occasionally encountered. The right-hand scale is chosen to represent fairly loud reproduction (classical monitoring); the dash-dot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