electronic loudspeaker

It is widely accepted that the loudspeaker is the weakest link in the high-quality audio chain. This is particularly the case at the lowest working frequencies due to the difficulty of providing a useful airload for a radiating diaphragm that has dimensions small compared to the sound wavelength. This compels the manufacturer to adopt clever but more or less expensive constructions for the loudspeaker unit and its enclosure.
The manufacturer has the resources and facilities to tackle the problems at the mechanical-acoustical stage. This article explains that the do-it-yourself approach that provides the best results at the lowest price is invariably the "electronic loudspeaker".

Methods of electronically compensating for the weaknesses of loudspeakers are by no means new. As Harwood recently pointed out, a patent granted in the early 20's already describes a "motional feedback" system.
The basic idea is to somehow derive a signal that depends on the loudspeaker's actual movement and to compare this with the original input signal. The resulting 'error' signal is used to modify the drive to the loudspeaker. One way of obtaining a feedback signal is to extract the voltage that is induced in the loudspeaker's drive-coil when the cone moves.
This extraction of the back-voltage has to be done with great care if the system is to remain stable. Also, not every loudspeaker is suitable for the technique.
The design described in this article has, however, behaved itself properly during many demonstrations.
Apart from the fact that the electronic loudspeaker does not need a specially-mounted pickup-device, which makes it simple to build up, it can be compared to normal applications of the same driver as follows:

a) the lower limit of 'flat' amplitude response is independent of the fundamental resonance-frequency of the driver itself (or of the driver in its enclosure).
b) distortion due to certain mechanical non-linearities in the driver can be considerably reduced.
c) although the frequency response remains 'flat' below the fundamental resonance frequency of the driver in its enclosure, the maximum acoustical power output falls off below this frequency.
It turns out however – as will be explained later – that a 20-watt amplifier produces more than enough sound level for domestic listening situations.
d) a loudspeaker operating in this kind of feedback system can produce good sound at higher as well as lower frequencies, although optimum results can only be obtained when an extended circuit is carefully matched to the individual loudspeaker. On the other hand, the greater cone excursions associated with extended bass response will aggravate the high-range (Doppler) distortion problem, so that it is desirable to use the electronic loudspeaker only for the woofer-range.

The electronic woofer

The behaviour of a moving-coil woofer in a closed box can be fairly accurately predicted from simple theory (see 'loudspeaker diagnosis'). This theory can be used to find a way to improve the bass response.
If one 'looks into' the loudspeaker terminals one 'sees' a series-connection of two impedances – i.e. a voltage divider. One of these, called the static or 'blocked impedance', is the value measured when the voice-coil is prevented from moving (e.g. fixed with glue). The other impedance arises because of the movement of the coil in the permanent magnetic field and is called the dynamic or 'motional impedance'. We will refer to them as Z_s and Z_d respectively. The radiated sound energy corresponds to the dissipation in a 'radiation resistance' which forms part of Z_d.
The objective in operating the loudspeaker is to arrange that this dissipation will be frequency-independently controlled by the input signal applied to the driving amplifier.
The problem is that both Z_s and Z_d vary with frequency, that these variations are by no means the same, and that furthermore the radiation resistance has neither a constant value nor is it a constant proportion of Z_d. Pity the loudspeaker designer! Let us see what can be done about this state of affairs.
The approach adopted for the electronic loudspeaker is to:
a) note that the static impedance Z_s consists essentially of the voice-coil resistance and self-inductance in series and that it is sufficiently well-behaved for elimination by means of an equivalent negative output impedance of the driving amplifier.
b) use this technique to deal with Z_d, and then apply a compensation to the driving signal, to take care of the frequency-dependence of the radiation resistance. This is not too difficult for a loudspeaker acting as a piston in one wall of a closed box: it turns out (see 'loudspeaker diagnosis' elsewhere in this issue) that a 'flat' frequency response is obtained when the voltage across the radiation resistance is made inversely proportional to the frequency. This can easily be done using a 6dB/octave low-pass network inserted ahead of the amplifier in the bass channel. This network, together with the negative output impedance of the amplifier, forms the basis of the 'electronic loudspeaker'.
Summing it all up it can be stated that the radiated sound energy corresponds to the dissipation in the radiation resistance; that for a constant voltage across this resistance the dissipation will increase in proportion to the square of the frequency; that for a flat frequency response this voltage must therefore be inversely proportional to the frequency – this calls for a 6dB/octave low-pass network; that this voltage can be forced to the required value once the series impedance Z_s has been eliminated by means of a negative amplifier output impedance. The driving amplifier will then automatically deliver the required drive current.

Negative output impedance

A negative output impedance can be achieved by means of the arrangement shown as a block-diagram in figure 1. 'A' in this diagram represents the gain of the driving power-amplifier. The loudspeaker is represented as Z_L, consisting of the impedances Z_s and Z_d in series. Z_f is a feedback current-sensing impedance, connected between the 'cold' loudspeaker terminal and amplifier earth return.
The voltage drop across Z_f is found from:

$$\frac{v_z}{Z_f} = i_o = \frac{v_o}{Z_L}$$

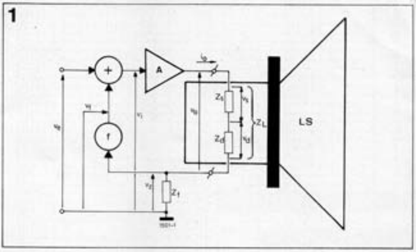
(since the current through feedback network f is negligible) so that:

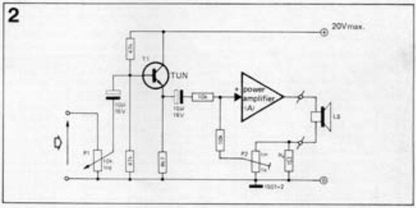
$$v_z = \frac{Z_f}{Z_L} \cdot v_o$$

The output impedance is worked out as follows:

Figure 1. Block diagram of the arrangement for achieving a negative output impedance.

Figure 2. Practical realisation of the 'electronic loudspeaker'. Adjustment is carried out by turning P_2 up from minimum setting (slider to chassis) until the point at which the system starts to 'howl' – and then backing off until the oscillation just ceases. (What was that remark about old-fashioned TRF receivers with 'reaction'?)





$$v_o = A \cdot v_i - v_z = A(v_e + f \cdot v_z) - v_z =$$

$$A \cdot v_e + (Af-1)v_z = A \cdot v_e + (Af-1) \cdot \frac{Z_f}{Z_L} \cdot v_o$$

After some tidying up:

$$v_o = A \cdot v_e \cdot \frac{Z_L}{Z_L - (Af-1)Z_f} = A \cdot v_e \cdot \frac{Z_L}{Z_L + Z_o}$$

in which the output impedance has been introduced as

$$Z_o = -(Af-1) \cdot Z_f$$

This is negative provided that $Af > 1$.
To compensate the static impedance of the loudspeaker we require:

$$Z_o = -Z_s.$$

Assuming that this is successfully done we find:

$$v_d = v_o - v_s = \frac{Z_d}{Z_s + Z_d} \cdot v_o =$$

$$\frac{Z_d}{Z_s + Z_d} \cdot A \cdot v_e \cdot \frac{Z_L}{Z_L + Z_o} =$$

$$\frac{Z_d}{Z_s + Z_d} \cdot A \cdot v_e \cdot \frac{Z_s + Z_d}{Z_s + Z_d - Z_s} = A \cdot v_e!$$

The voltage drop across the dynamic impedance (v_d) is directly proportional to the incoming signal voltage (v_e). This achieves the first objective.

Practical aspects

For many moving coil loudspeakers the impedance Z_s at low frequencies is predominantly a resistance: the resistance of the driving coil (R_s). It is therefore sufficient to use a resistor (R_f in figure 2) as the sensing element for the current-feedback (Z_f). The compensation in this range is set up by adjusting the feedback attenuator (f) so that:

$$R_s = (Af-1) \cdot R_f$$

This can conveniently be done using the circuit of figure 2. The amount of (positive) current feedback is adjusted by P_2. Starting with the slider of P_2 at the earth end, without any input signal, slowly turn up P_2 until a 'howl' from the loudspeaker heralds the onset of oscillation. A slightly lower setting, for which the system just remains stable, is optimal.

One or two more practical aspects appear from the circuit diagram. The buffer stage (T_1) has been included to prevent adjustment of the volume control P_1 from upsetting the calibration by means of P_2. Whether this stage is necessary or not will depend on where the volume control was placed in the original amplifier.

The one place where the volume control may not be located is in the power amplifier itself! The gain factor A must remain constant. On the other hand, if the volume control is in one of the preamplifier circuits the buffer stage will usually not be needed.

Low-pass network

We already indicated that a 6dB/octave low-pass network is required ahead of the power amplifier. The choice of rolloff point is a compromise.

The rolloff point of the network determines the lower limit of compensated response. If this rolloff point is placed at 40 Hz, for example, the response curve of the electronic loudspeaker will be essentially flat from 40 Hz to at least 300 Hz.

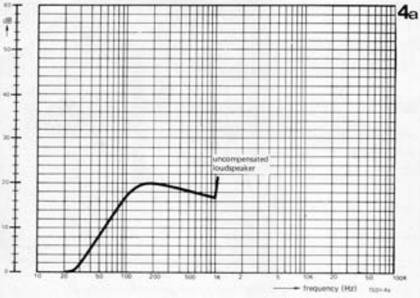
On the other hand it is undesirable to place this lower limit unnecessarily far down the frequency range. This is because the extension of bass response has to be 'paid for'. If we assume that the maximum current which the power amplifier can pass through the loudspeaker is 'matched' to the amount of force which the drive-unit can handle without damage, then the 'price' for an extension of flat bass frequency response is reduced full-drive sound level throughout the *whole* working range of the woofer.

As the lowest working frequency is reduced past the 'normal' loudspeaker-in-box cutoff, compensation of the response requires rapidly increasing amounts of drive-power for a given sound level. Since the drive-power is limited, the power response must fall off. This is not so dramatic as it may sound, however, since the maximum power level in any normal music spectrum (including organ pedal!) rolls off at approximately 6dB/octave below about 100 Hz, so that the maximum power that the loudspeaker can deliver matches the maximum power that is required over the whole frequency range.

How many watts?

What is the desirable loudness level – and therefore how much power is necessary – is probably the 'cause célèbre' of hifi reproduction. The physical situation is sufficiently flexible to provide grounds for 'objective' justification of almost any subjective opinion, while opinions vary between the extremes of 'shatteringly loud and the devil take the neighbours' and 'the loudest passages should not impede normal conversation.'

We will try to steer a middle-course – based on the requirement that the maxi-



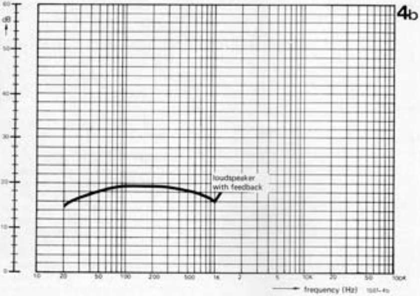


Figure 3. Block diagram of a multi-way system which uses the 'electronic loudspeaker' in the bass channel. Such a system will be described in a further article.

Figure 4. The frequency response of a 5" loudspeaker in a 5" X 5" X 6" (!) closed cabinet, with and without compensation.

mum sound level should be 'reasonable' and 'acceptable' in the 'normal domestic listening situation' (whatever that may be). The very words indicate that this will be pure conjecture – yet it would surprise us if we found ourselves very far off the mark.

For reproduction of 'serious' music (symphony concert, baroque recital etc.) a strong case exists for playback at the same apparent loudness level as that of the original performance. For a typical concert hall the peak loudness level during fortissimo passages varies from about 95 dB at the rear of the hall to about 105 dB near the front. (The reference level for these decibels is the normal threshold of hearing – an intensity of 10^{-12} watts per metre^2.)

The average level of a fortissimo passage is much lower. At the other end of the range, the pianissimo peak level is typically 35 to 45 dB (just far enough above the noise level due to the air-conditioning!) This 60 dB dynamic range can only be tolerated in a large hall, where the 'indirect' or 'reverberant' sound field behaves quite differently to that in a domestic listening room. A similar apparent loudness range appears to be achieved in the latter situation when the reproduced dynamic range is about 40 dB – with fortissimo peaks at 90 dB. Most recording companies produce material with a 40 dB dynamic range, which was monitored at this 90 dB fortissimo-peak level. And they should know.

Let us therefore assume that our 'electronic loudspeaker' must be able to produce momentary loudness peaks of 90 dB in typical domestic surroundings. Since the indirect field takes time to build up intensity, it will be the loudspeaker's direct radiation intensity which must be able to reach 90 dB. Assume further that the listener is 3 metres from the loudspeaker, which radiates evenly in all directions (a fair assumption up to about 400 Hz). The required acoustical power is:

$P_O = 4\pi r^2 \times I_d = 4\pi \cdot 9 \cdot 10^{-3} \cong 100$ milliwatts, where we have inserted 3 metres for distance (r) and 10^{-3} watts per metre^2 for the direct intensity (I_d), i.e. 90 dB. A loudspeaker with 1% efficiency will do this on 10 watts of electrical input – and only 'acoustic suspension' woofers with heavy moving systems are less efficient than this! A 20-watt amplifier for each of two stereo woofer-channels is clearly sufficient.

The driving amplifier

The driving amplifier used in this system must reach a very high standard of performance. Not every 'high fidelity amplifier' automatically satisfies the requirements.

The most important requirement is that the amplifier be unconditionally stable, with any load.

In the compensated system, after all, the apparent amplifier load is the loudspeaker's motional impedance. This appears as a parallel tuned circuit: inductance, capacitance and resistance all in parallel! Worse still, this apparent load is the result of applying positive current feedback around the whole system . . .

We previously described the 'Equa-amplifier', which meets the requirements with an ample margin. It was indeed designed with the electronic loudspeaker in mind. This amplifier, like most 'six-transistor' circuits, has its input and output voltages in-phase. If an amplifier which reverses the signal phase is to be used, it will be necessary to insert a phase reversal in the feedback path. This can be simply achieved by replacing the figure 2 buffer stage by a so-called 'virtual earth' mixer.

The loudspeaker

In principle the loudspeaker and its enclosure do not have to meet any severe requirements. If the best results are to be obtained, attention must nonetheless be paid to one or two details.

The volume of the enclosure will determine the fundamental resonance frequency of the compensated system – and this is the point at which the power response starts to roll off. For normal dom-

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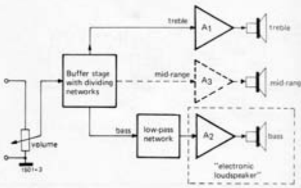

estic listening a volume of 15 litres is adequate. (15 litres = 15 cubic decimetres = 0.5297200050 . . . cubic feet . . . if you must!) If only background music is to be reproduced, the enclosure will do as soon as the driver fits inside it!

The enclosure should also be almost airtight. One way of achieving this is to start with a completely-sealed box, then to drill a small hole (about 2 mm ϕ) in the rear panel. This will enable variations of atmospheric pressure to equalise themselves. The amount of leakage is correct when the cone of the mounted driver takes several seconds to recover position after it has been gently pushed a small amount inwards, momentarily held stationary and then released. (N.B. Amplifier switched off!)

Finally, the walls of the box must be sufficiently 'solid'. They must not vibrate – and therefore contribute to the radiation – under the influence of the strong pressure changes in the driven box. Stiffening ribs may be applied if necessary. Damping material is not strictly necessary; but a single pad of glass-wool or similar material, lath-mounted in the middle of the enclosed volume, will control standing waves in the box. The latter can give audible trouble, particularly if the enclosure is fairly large.

The drive-unit itself should in principle meet three requirements: it must be able to handle sufficient power input; the magnet must be large enough to guarantee an unvarying flux through the entire coil during large excursions of the cone; the cone itself and the front-surround must be reasonably stiff. It must behave as a piston!

Special high-compliance woofers using a rubber front-surround are less suitable for this application, particularly when in a small enclosure. When the cone is driven outwards at high input levels there is a tendency for the surround to be sucked inwards!

The electronic multi-way system

It is best to use the electronic loudspeaker as the woofer in a multi-way system. Figure 3 shows the block diagram of such an arrangement.

The amplifier A_1 is a small high-quality amplifier (6-10 watts) which drives only the treble loudspeaker(s). If desired the reproduction of mid-range and tweeter-range may be separated. This can be done by means of a dividing network after A_1 or by the use of a separate mid-range power-amplifier A_3 (dotted).

The bass drive-unit and amplifier A_2 together form the 'electronic loudspeaker'. The low-pass step-network described earlier is installed ahead of this amplifier. The combination must meet the requirements mentioned above.

The block diagram finally includes a buffer stage with dividing networks for the bass and treble paths. These networks, like the low-pass step network, are built up from RC sections and buffer circuits.

In a further article we will describe complete two- and three-way systems based on the use of 'equa-amplifiers'. Details will be given of the dividing circuits and measurement results.

(to be continued)

In the text, figures and unavoidable formulae the following symbols have been used:

- Z_s = static ('blocked') impedance of the drive unit
- Z_d = dynamic ('motional') impedance of the drive unit
- Z_L = total impedance of the loudspeaker drive unit
- $-Z_s$ = negative (driving-) impedance
- Z_o = output impedance of the amplifier
- Z_f = feedback sensing impedance
- P_o = radiated acoustical power
- v_o = voltage across the speech coil
- v_e = incoming signal voltage
- v_i = modified amplifier input voltage
- v_z = current-dependent voltage across Z_f
- v_f = feedback voltage
- v_d = voltage across the motional impedance
- v_s = voltage across the static impedance
- R_s = copper resistance of the driving ('voice') coil
- R_f = feedback sensing resistor
- f = feedback factor
- A = gain of the driving amplifier proper
- i_o = output current
- I_d = intensity of the 'direct' loudspeaker radiation