Zero phase shift between the drive units in an active loudspeaker unit has been the goal of virtually all designers and constructors ever since the first multiple speaker unit was conceived. For a long time, it has been like trying to achieve a *perpetuum mobile*, but now it has become reality!

phase-corrected cross-over filter

by T Scherer

It is well known that the loudspeaker is the weakest link in an hi-fi chain: it is the final factor that determines how the hi-fi installation sounds. The most serious problem is that presented by the processing of the wide range of frequencies. As long ago as the thirties, designers have tried to solve this problem by sub-dividing the audible frequency range and using a separate drive unit for each of the resulting bands. In the simplest case, this means that a bass speaker (woofer) is used for the low audio frequencies, and a so-called tweeter for the high audio frequencies.

Right up to the 1960s, the network that divides the frequency ranges consisted of a passive filter constructed from chokes and capacitors. When semiconductors became less expensive, designers began to use active filters and to provide each separate drive unit with its own power amplifier that is fitted inside the speaker enclosure. Such active systems are generally better than passive ones, but they are also more expensive. But whether active or passive, filters create problems of their own.

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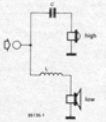

Fig. 1. The simplest form of cross-over network. Its simplicity causes problems, however.

Problems with filters

The simplest two-way dividing filter consists of a choke in series with the bass speaker and a capacitor in series with the tweeter — see Fig. 1. At the cross-over between low and high frequencies, both drive units are fed with the same signal, the level of which is about 3 dB below that of the nominal output at the input to the filter. Moreover, the signal at the bass speaker lags that of the input signal by 45°, while that at the tweeter leads that of the input signal by 45°. Because of the phase difference of 90° between the signals at the two speakers, the air pressures produced by them are added geometrically, so that the overall sound is as if caused by a signal that is identical to the original input signal to the filter. Provided, that is, that everything is ideal.

Tolerances in the components, differences in the drive units, and effects of the enclosure prevent such an ideal state being attained. Even tolerances of 10 per cent can alter the situation quite a lot. If, for instance, the capacitance is 10 per cent smaller, and the choke 10 per cent larger, the levels at the two drive units are almost 0.5 dB lower than in the ideal case: —3.444 dB. The phase difference is also larger: 95.5°. The result is that the overall signal is almost 0.9 dB lower than the original signal. This may not seem serious, until the considerations concern higher-order filters that give a Bessel, Butterworth, or Chebishev response. Such filters have a much steeper cut-off profile. Even small component tolerances then cause a reduction of a few dB in the available gain. The phase characteristic in these filters also has a steeper roll-off. Component tolerances may cause such a large phase shift that the gain is reduced by another few dB. Finally, the loudspeaker characteristics themselves should also be taken into account. The (larger) bass unit is inherently somewhat slower in action than the (smaller) tweeter: they have different rise times. The difference between these times manifests itself at the cross-over frequency as an additional phase shift. Odd-order filters have a phase difference of about 90°. In a two-way system with a cross-over frequency of 1 kHz and a difference in rise times of 100 μs (a typical, practical value), there is an additional phase shift of 36°, resulting in a total of 126°. Even if all other parameters of the network are one hundred per cent correct, such a phase shift results in a 2 dB loss. In even-order filters, the situation is somewhat better: here an additional phase shift of 36° causes only 0.5 dB loss.

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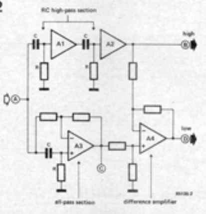

Fig. 2. New techniques of filter design enable the construction of this type of filter that in spite of having only two *RC* sections provides a 12 dB/octave profile and obviates a number of problems inherent in other types of filter.

Solution

The requirement is, therefore, for a filter that produces no phase shift between the loudspeakers, is not affected by component tolerances, is easily reproducible, and, in spite of being a unit for home construction, is absolutely linear.

A critically damped second-order filter (i.e., one consisting of two cascaded simple RC high- or low-pass sections) has the same phase response as an all-pass filter. The filter in Fig. 2 consists of two high-pass sections, A_1 and A_2. The phase shift, Φ, is calculated from

$$\Phi = \arctan \frac{1}{R\omega C}\ [^\circ] \qquad (1)$$

where $\omega = 2\pi f$ (and f is in Hertz); R is in ohms; and C is in farads.

The phase shift in all-pass filter A_3 in Fig. 2 is determined by

$$\Phi = 2 \arctan \frac{1}{\omega RC}\ [^\circ] \qquad (2)$$

The phase shift vs frequency and gain, G, ($= U_o/U_i$) vs frequency characteristics at B, C and D in Fig. 2 are given in Fig. 3. They show that the phase shift at B (output of the two high-pass sections) is identical to that at C (output of the all-pass section). The only difference between the signals at B and C is that the gain of the former is frequency dependent

$$U_{o(B)}/U_i = \frac{1}{\sqrt{1+(1/\omega RC)^2}} \qquad (3)$$

The amplification of the signal at C is unity, i.e.,

$$U_{o(C)}/U_i = 1 \qquad (4)$$

The output at B only contains high audio frequencies, whereas that at C includes all audio frequencies at the same level. If then the signal at B is deducted from that at C, the gain, G_D, at D, i.e., Eq. (4) — Eq. (3) becomes

$$G_D = U_{o(D)}/U_i = 1 - \frac{1}{\sqrt{1+(1/\omega RC)^2}} \qquad (5)$$

Plotting Eq. (5) shows that the output voltage $U_{o(D)}$ is very small at high frequencies and is equal to U_i at low frequencies. The signal at D is, therefore, the low-frequency output of

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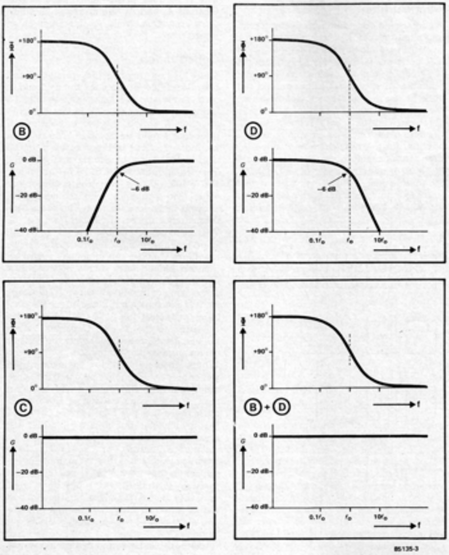

Fig. 3. Box B shows the phase vs frequency and gain vs frequency response at point B in Fig. 2. Boxes C and D show the same for points C and D respectively. Box B + D gives the phase and gain vs frequency response of the acoustically summed B and C outputs of Fig. 2.

the filter. Note, however, that its phase characteristic has the form of that of a high-pass section; that is, the phase of $U_{o(D)}$ leads that of the input signal. This means that there is no phase difference between the high-pass and low-pass branches over the entire frequency range.

The dividing filter of Fig. 2 is a two-way version. It has a 12 dB/octave cut-off profile. Normally, such second-order filters have two *RC* networks in both the low- and high-pass branches. Problems may arise then if, owing to component tolerances, the time constants in the two branches are not the same. These problems are negligible in the set-up of Fig. 2, because, due to difference amplifier A_4, the sum of outputs B and D is always the same as input A, irrespective of component tolerances.

The cross-over frequency, f_o, is defined as that frequency where both the low- and the high-pass output are attenuated by 3 dB. This happens when $\omega_o RC = 1$, whence

$$f_o = 1/2\pi RC \quad \text{[Hz]} \qquad (6)$$

where R is in ohms and C in farads.

Practical filter

The foregoing considerations lead to a practical filter, the block schematic of which is given in Fig. 4 and the circuit diagram in Fig. 5. It concerns a three-way version with a 24 dB/octave cut-off profile: its response resembles that of a critically coupled network, i.e., there is no tendency of overshoot. The filter has two low-frequency outputs, which are inverted with respect to one another. This offers a simple push-pull amplifier for the bass drive unit, since this often requires more power than the middle- and high-frequency speakers.

The input signal — Fig. 4 — is applied via a buffer to a four-stage *RC* high-pass section, and is then available at the high-frequency output. Two of the *RC* stages are designated a; the other two, b. This is done to clarify that the phase shift in all-pass section AP_a is identical to that in the HP_a sections; the phase shift in AP_b is the same as that in sections HP_b — more about this later.

The difference amplifier forms from its two input signals a low-frequency output, whose half power frequency, f_o — see Eq. (6), is the dividing frequency between the high- and middle-frequency branches. This signal is fed to another four-stage RC high-pass section, and is then available as the middle-frequency output. The low-frequency output is obtained in the same way as the middle-frequency output.

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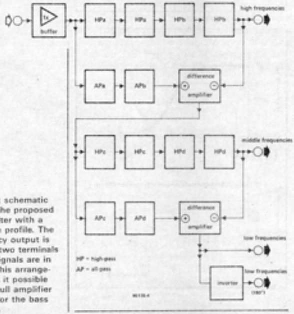

Fig. 4. Block schematic diagram of the proposed three-way filter with a 24 dB/octave profile. The low-frequency output is provided at two terminals where the signals are in antiphase. This arrangement makes it possible for a push-pull amplifier to be used for the bass loudspeaker.

Circuit description

Opamp A_1 is the input amplifier, whose low-impedance output provides the audio signals for the remainder of the circuit. Opamps $A_2 \ldots A_5$ and $A_9 \ldots A_{12}$ are buffers that decouple successive high-pass sections from one another. Opamps A_6, A_8, A_{13}, and A_{14} are connected as all-pass sections with a leading phase shift characteristic. Opamps A_8 and A_{15} are arranged as difference amplifiers, while A_{16} functions as an inverter.

Construction

There is no ready-made printed-circuit board available for the filter, but it should fit on a vero board, or similar, the size of half a Eurocard, i.e., 80 x 100 mm. Before the construction is started, the two cross-over frequencies should be decided. The relevant component values in Fig. 6 result in cross-over frequencies of 570 Hz and 3800 Hz. That is a frequency ratio of 1:6.7 — about two and a half octaves, which is a convenient value. The frequency ratio should not be allowed to be less than 1:4. The cross-over frequencies, f_o, are calculated from Eq. (6). The next aspect to be looked at is the impedance of the *RC* networks. To ensure low thermal noise and minimum delays, all resistors should have values between 10 k and 27 k. As the opamps also contribute to noise (see, for instance *Intuitive IC Opamps* by T M Frederiksen, published by National Semiconductor), the TL 074 should be preferred to the TL 084. Capacitor values are calculated from Eq. (6) once the resistor values have been determined.

Where absolute accuracy is desired, one per cent resistors should be used: these are much cheaper and more easily obtained than close-tolerance capacitors. However, in most cases five per cent resistors are perfectly all right, but it is preferable that resistors with the same letter indices, for instance, R_{3a} and R_{5a}, or R_{13g} and R_{14g}, have identical values. It is, therefore, more economical to buy, say, fifty 5 per cent 18 k resistors than thirty-two 1 per cent ones, and, with the aid of a digital multimeter, sort out equal-value ones: four groups of three and eight sets of two identical resistors are required.

Capacitors can be sorted in a similar way — see Fig. 6. Connect one of the capacitors to

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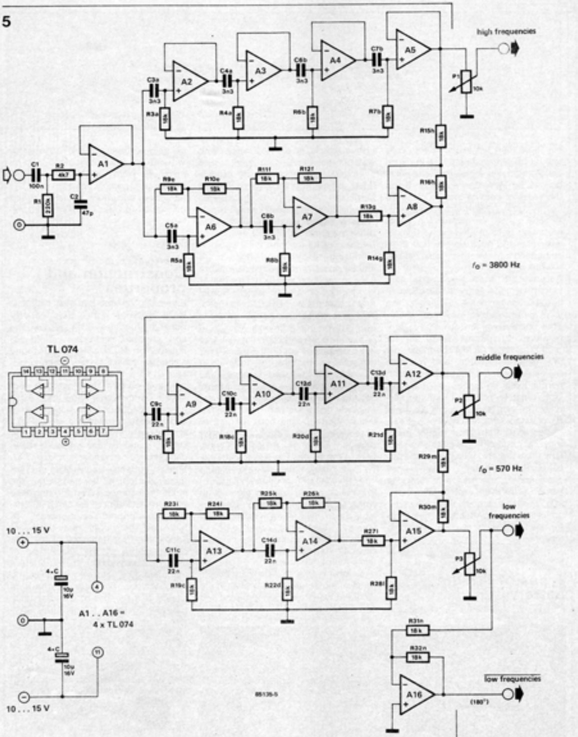

Fig. 5. Circuit diagram of the 24 dB/octave fourth-order filter.

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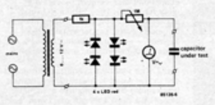

the output terminals of this test circuit and adjust the potentiometer for a reading of 1.8 V on its 2.0 V AC range. Without resetting the potentiometer, check all capacitors and group them in identical sets.

The overall result of this grouping should be sets of resistors within 1 per cent, and sets of capacitors within 5 per cent. These tolerances will ensure a cross-over filter with an accuracy that exceeds even that found in expensive, professionally made loudspeaker units. ◼

Fig. 6. Test circuit to enable the sorting of capacitors in groups of near-identical values.