Loudspeaker system design

Three-enclosure system with active delay and crossovers — part 2

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This is not the "ultimate loudspeaker", but in the first part of this article in the May issue Mr Linkwitz says that "the recording is the next weak link in the chain." The equalized system incorporates electronic crossovers and delay compensation. Part one describes the enclosure design and this article gives sufficient information for the experienced constructor to duplicate or to adapt the electronic design to other needs.

THIS SPEAKER DESIGN project was started with the idea of mounting the boxes flush with the surrounding wall. Positioning the box in front of a wall causes a severe dip in amplitude response when the distance from the front of the box to the wall equals a quarter wavelength. For a typical 300mm enclosure depth the dip occurs around 250Hz, Fig. 2 (ref. 7).

If one imagines the walls near the speakers to be made of mirrors then one can easily visualize the images of the box in these mirrors. At frequencies where the radiation from the box is omnidirectional, each of these images can be thought of as a separate sound source, whose output will add or subtract from the original source, depending upon how far the image source is removed in terms of wavelengths. For a half wavelength distance to the image source, corresponding to a quarter wavelength distance from the speaker to the mirror, the output from the real and the imagined source will cancel each other.

This description of virtual sound sources is valid provided that the speaker is radiating sound towards the walls and that the walls, floors and ceiling act as acoustically reflecting surfaces, i.e. mirrors. Mounting the boxes flush with the reflecting surface eliminates the virtual image behind the speaker and produces a smooth frequency response. The completed system with flush mounted boxes performed very satisfactorily. In particular it gave a good sense of depth perspective for some stereo material.

It was discovered later that by moving the speaker out into the room and at least 0.5-1m away from walls and floor, a significant improvement in sound perspective was obtainable, see photograph in part 1. On approprate program material it now became quite easy to pinpont the location of individual instruments both laterally and in their distance behind the speaker plane. It might be said that the whole sound stage moved into focus.

Furthermore, tape hiss and record surface noise became spatially separated from the musical material. The noise originated definitely at the speaker boxes while the musical instruments assumed their own space between and behind the boxes. In this sense the noise and ticks from a record surface are comparable to tne coughing and shuffling of people at a live concert where one can concentrate on the performance and not be side-tracked by unrelated acoustical events[8].

It is not clear why the placement only a relatively short distance away from the walls should give such a marked improvement, particularly in light of the just-mentioned frequency response interferences from virtual images. It might be that the ear-brain combination performs a time domain analysis and is able to allocate the wall reflections which occur 4 to 6ms later than the direct sound to the characteristic of the listening room and to the program material.

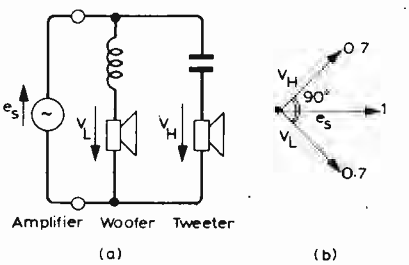


Fig. 7. *Schematic network with 6dB per octave slopes and voltage phasor diagram at the crossover frequency.*

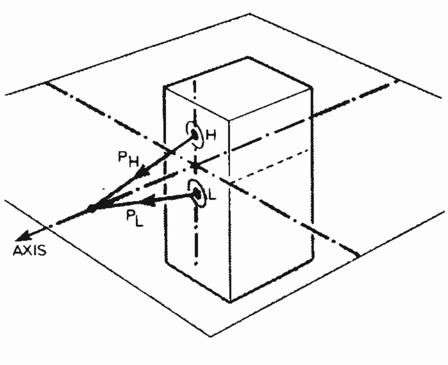


Fig. 8. *Plane of points are equidistant from the high frequency and low frequency drivers. The sum of the sound pressures is proportional to the sum of the electrical drive signals only in this plane.*

Mounting of the speakers away from the walls was accomplished by hanging them from the ceiling with a nylon monofilament. Electrical connections run from the back of the enclosure to the wall behind it and also serve to keep the speaker aiming forward. The small hanging units might be appropriately called satellites to the woofer box. The woofer itself is located halfway between the satellites, which are 2.5m apart.

The listening room is 5×8×4m (w×l×h), with the speakers in front of the narrow walls and the typical listening positions 5 to 6m away from the satellites.

Crossover design

The simplest crossover network is the —6dB per octave slope filter of Fig. 7. Assuming idealized components, the current from the generator will split in such a way that the vector sum of the voltages across the low and high frequency driver terminals is equal to the source voltage at all frequencies $V_L \hat{+} V_H = e_S$.

Correspondingly the vector sum of the sound pressures p_L and p_H generated by the drivers will be directly proportional to the generator voltage $p_L \hat{+} p_H = ke_s$ and independent of frequency, provided that the distance from the listener to each of the drivers is identical. The B110 and T27 drivers though are a wavelength apart, which means that equidistance is obtained only for a plane in space, Fig. 8. For points outside this plane the sum of the two driver outputs will vary with frequency.

Furthermore, because the two drive voltages already have a 90° phase difference the summation will be different for symmetrical points above and below the plane of equidistance. In the crossover frequency region where both drivers contribute equally the system will radiate its maximum pressure at a 14-degree angle below the plane, Fig. 9. This simple dividing network has a wide range of overlap between the two drivers and therefore a tilted radiation pattern over at least two octaves.

A seemingly attractive feature of this network is its complete lack of phase distortion for points which are equidistant from H and L, Fig. 8. At these points perfect square-wave reproduction is achievable under free-field conditions or in an anechoic chamber. In a living room the increased radiation towards the floor and the reduced radiation upwards will produce a coloration in sound due to the frequency-selective change of the reverberant field. The ear is more sensitive to the amplitude response than to phase shift. Therefore this filter and related designs with even greater than 90° phase difference between the drive signals and correspondingly greater off-axis intensity peaks are not used for the satellite system[9].

A 24dB per octave slope filter was chosen which has no off-axis peaks in the radiation pattern[10], Fig. 9. The steep filter cut-off narrows the overlap region where the B110 and T27 interact. The T27 has its fundamental resonance at 1.4 kHz and the highpass provides 27dB of attenuation at this frequency. At 5kHz where the B110 exhibits a cone resonance the filter has reduced the drive voltage by 18dB, Fig. 10. A 6 or even 12dB per octave filter would have insufficient attenuation to minimize exciting these resonances. The 18dB per octave filter was not considered because it tilts the polar pattern.

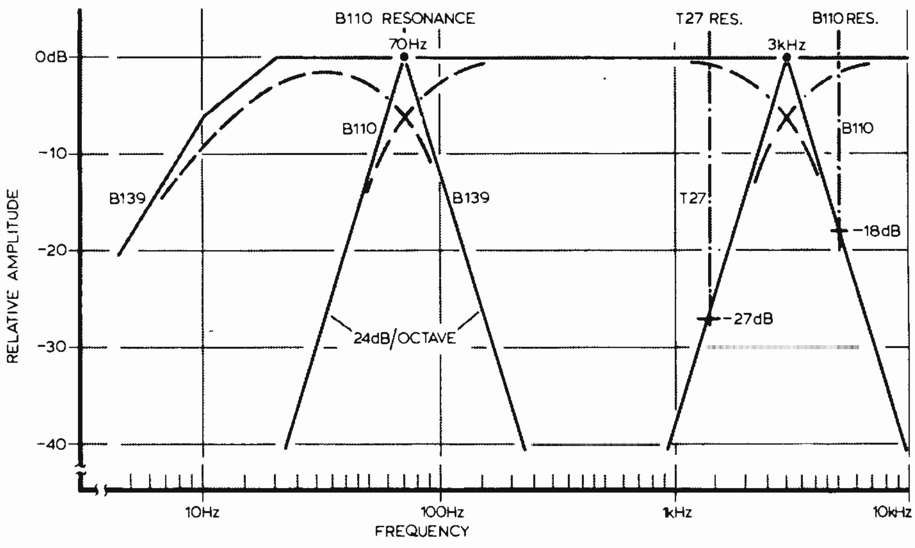


Fig. 10. *Schematic response for crossover points and driver resonances.*

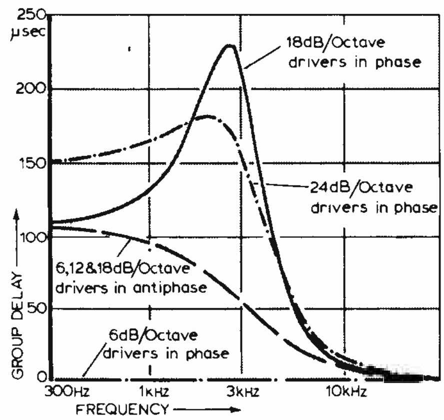


Fig. 11. *Group delay frequency response of a speaker system due to a 3kHz crossover between midrange and tweeter with conventional first and third-order Butterworth networks, and with second and fourth-order cascaded Butterworth sections (ref. 10).*

All these filters, with the exception of the 6dB per octave network, have a frequency-dependent phase shift and consequently some phase distortion. Only a network of linearly increasing phase shift with increasing frequency will have no phase distortion. The slope of the phase curve is constant in this case. Any deviation from the constant slope indicates that some amount of phase distortion is present. The question arises how much slope variation can be tolerated before it becomes audible and not merely visible on an oscilloscope. The slope of the phase curve, usually referred to as envelope delay or group delay, has been plotted for typically used Butterworth crossover networks and the new network function[10], Fig. 11. Merely changing the polarity to one of the drivers drastically changes the group delay for the summed driver outputs as in the case of the first and third-order Butterworth crossovers. Their on-axis amplitude response is unchanged, unless the drivers are separated some distance from each other. Then the polar pattern will tilt either up or down with the change in driver polarity.

To investigate the audibility of phase distortion an all-pass network was built which duplicates the group delay of the new second and fourth-order crossover networks (12 and 24dB per octave curves in Fig. 11). Listening with headphones to stereo and mono program material, no audible difference could be detected with either one of the all-pass networks switched in or out.

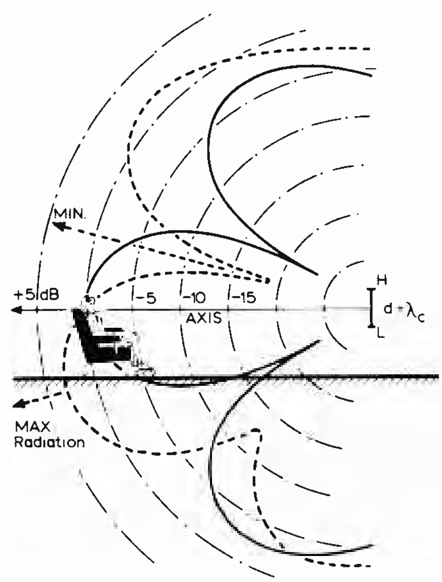


Fig. 9. *Radiation of a 6dB per octave crossover network at the cross-over frequency (3dB peak occurs below the plane of equidistance for non-coincident drivers) and the symmetrical pattern of a 24dB per octave crossover network at the crossover frequency (ref. 10).*

Therefore it seemed safe to use the fourth-order filter with its sharp cut-off behaviour which minimizes the overlap between drivers.

Crossover and equalizer circuits

The crossover networks and equalizers consists of a variety of active filter circuits. The overall block diagram of Fig. 12 gives an indication of the system complexity. Design formulas are presented for each functional block so that the experienced constructor should be able to duplicate the circuits of Fig. 13 or adapt the design to particular needs.

3kHz crossover networks

The fourth-order high and low-pass filters are made up from cascaded second-order Butterworth sections, Fig. 14. The outputs V_H and V_L are in phase with each other at all frequencies and the voltage sum is equal to V_{IN}. At the crossover frequency f_c, therefore, the output from each filter will be $V_{IN}/2$ or 6dB down, which is different from the typical 3dB crossover point for filters where V_H and V_L are in phase quadrature[10].

Delay compensation

The B110 and T27 drivers do not radiate from the same acoustical plane even though they are mounted on the same

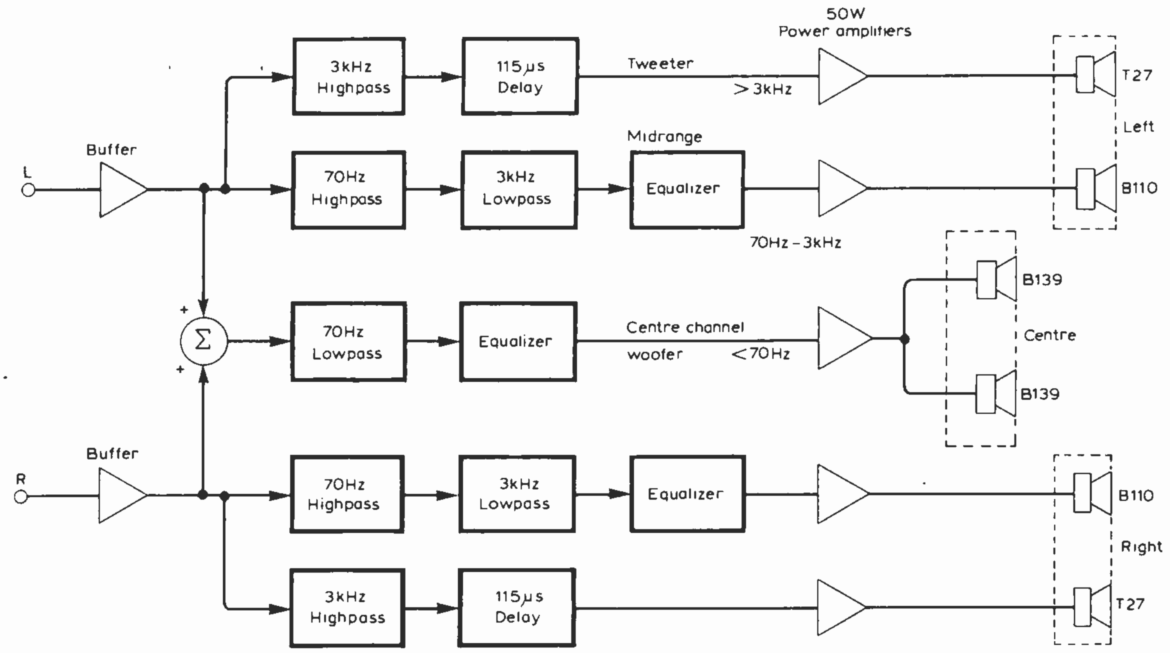
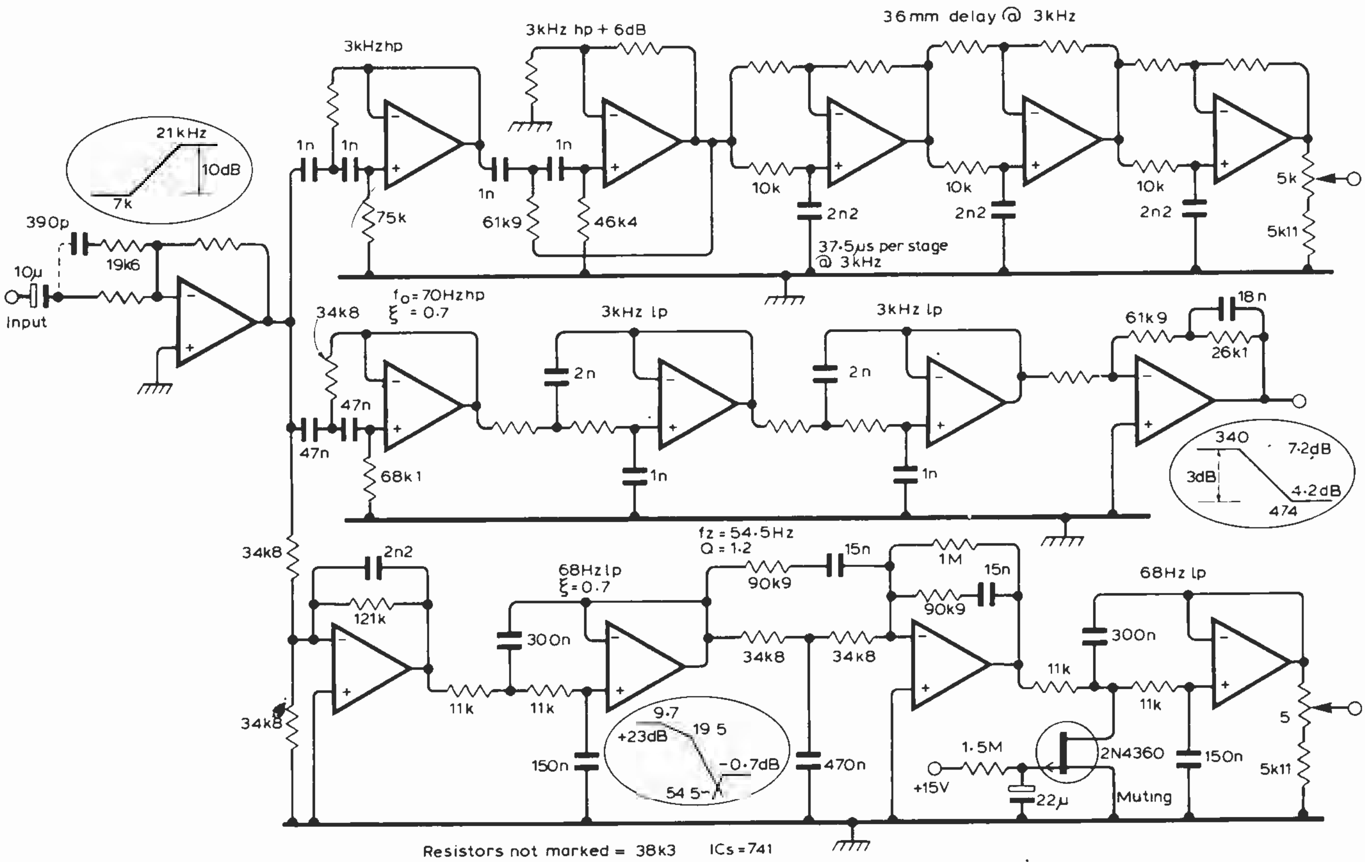


Fig. 12. *System block diagram. Design formulas for each functional block are given to allow adaptation of the circuits of Fig. 13.*

Fig. 13. *Circuit diagram of crossover networks and equalizers incoroporating delay compensation. Broken lines show optional h.f. boost components.*



baffle. The electrical signals arrive at each voice coil at the same time but because the T27 voice coil is located in front of the B110 voice coil the sound pressure wave generated by the T27 will be advanced relative to the B110. The 40mm driver off-set may seem insignificant unless it is related to the 144mm wavelength of a 3kHz tone where it corresponds to a 100° phase difference between the two driver outputs.

The effect of driver off-set on the on-axis frequency response can be quite significant, particularly if both drivers contribute almost equally over a wide frequency range, Fig. 15. The frequency region of overlap is significantly narrower for higher-order filters because of their steeper cut-off.

The driver offset can be compensated for by adding electrical delay to the tweeter drive signal, or by mounting the tweeter in a different plane. Mechanically moving the tweeter back is feasible provided care is taken to avoid sharp cabinet edges and their associated scattering of sound. For electrical delay an all-pass network has been used, Fig. 16. Its delay varies with frequency from $\tau = 2RC$ at low frequencies, approaching zero delay at very high frequencies. To reduce the frequency dependency in the crossover region of around f_c the component values should be chosen such that $RC \leq 1/20f_c$. Several delay stages are cascaded to obtain the required total delay. This delay has to be determined experimentally, but the spatial driver offset gives a reasonable starting point.

70Hz crossover network

The transition between the woofer and the satellite uses 24dB per octave slope filters similar to the 3kHz crossover. A transition frequency of 70Hz was chosen because the B110 output is 3dB down at this frequency due to the small internal volume of the satellite enclosures. The output continues falling off at a 12dB per octave rate below this frequency with approximately second-order Butterworth response shape. Therefore the driver in the closed box can be used as one half section of the required high-pass filter. The other half is implemented with an active second-order Butterworth filter section — the first stage in the centre channel of Fig. 13. The low-pass filter for the B139 is again the two amplifier fourth-order network design — the second and fourth stages of the lower channel in Fig. 13.

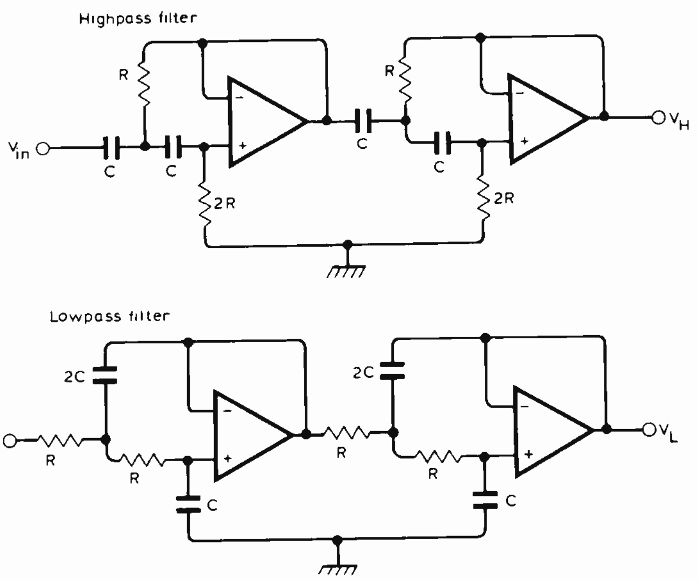
Woofer equalization

The response of the woofer does not extend sufficiently far down in frequency. The fall-off in acoustic output will therefore be compensated with a properly increasing drive signal. Over the frequency range where the driver acts like a rigid piston its frequency response when mounted in a closed box (ref. 11) is

$$F_w = \frac{\left(\frac{f}{f_0}\right)^2}{\sqrt{\left[\left(\frac{f}{f_0}\right)^2 - 1\right]^2 + \left(\frac{1}{Q_0}\frac{f}{f_0}\right)^2}}$$

This is a high-pass function with a corner frequency near the closed box resonance f_0 and some peaking depending on Q_0, Fig. 17. The two parameters f_0 and Q_0 can be conveniently determined from the frequency response of the driving point impedance[11] of the speaker system, Fig. 18. If the system is driven from a generator with an internal impedance much larger than R_{max}, then the terminal voltage becomes proportional to the system impedance and Q_0, f_0 can be determined from the voltage response as in Fig. 19.

For the two B139 woofers in their closed box, the resonance occurs at 54Hz with a Q_0 of 1.2. The response can now be compensated with a network which exactly complements the woofer roll-off and extends it to a lower cut-off frequency, Fig. 20. This design approach can be used to equalize other speaker systems if careful attention is given to



$$f_c = \frac{1}{2\pi\sqrt{2}\,RC}$$

Fig. 14. *Fourth-order 24dB per octave crossover filter sections are made up from cascaded second-order sections in both high and low-pass forms.*

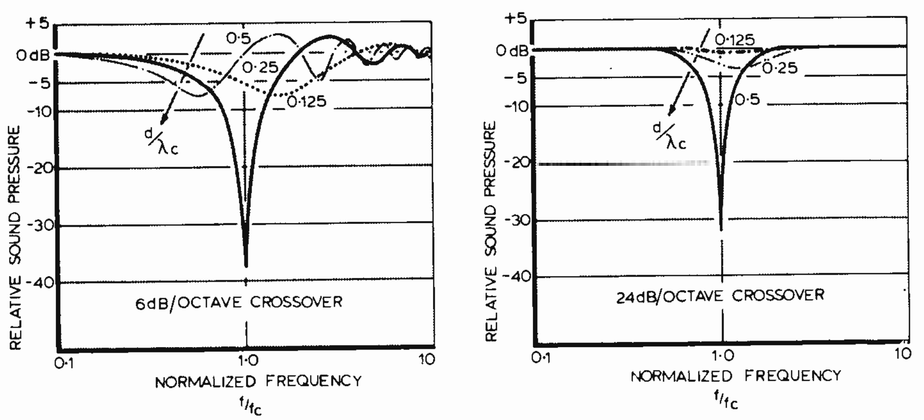
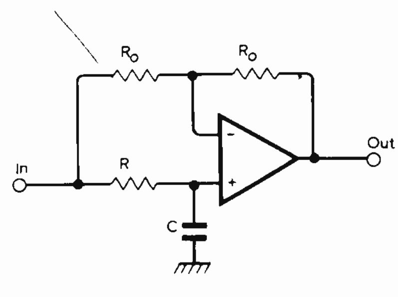


Fig. 15. *On-axis response when the tweeter is positioned acoustically in front of the midrange by d/λ_c with 6dB per octave crossover, and 24dB per octave crossover.*



$$\text{Delay } \tau = \frac{2RC}{1 + (2\pi f RC)^2}$$

Fig. 16. *Several all-pass phase shift networks are cascaded to obtain the required delay compensation.*

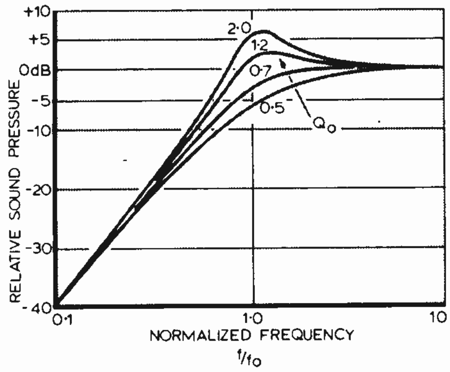


Fig. 17. *Fall-off in response of a rigid piston in a closed box (ref. 11). Box resonance f_o and Q are determined as in Figs. 18 and 19.*

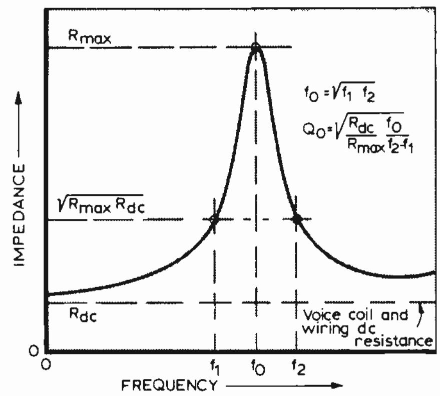


Fig. 18. *Schematic response of the woofer driving point impedance measured as in Fig. 19 from which f_o and Q_o of Fig. 17 are derived (ref. 11).*

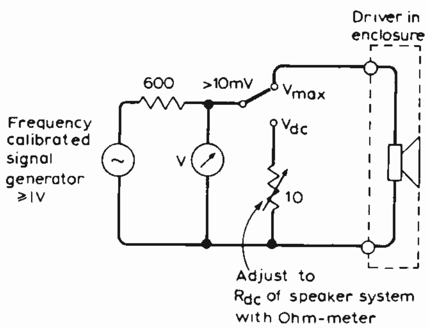


Fig. 19. *Measurement setup for Fig. 18 to determine R_{DC}/R_{MAX} from V_{DC}/V_{MAX} and the frequencies f_1 and f_2 from $V=\sqrt{V_{MAX} \times V_{DC}}$*

the cone excursion capability and the power amplifier output voltage swing limitations. Both increase by a factor of four when the cut-off frequency is lowered by an octave.

For the playback of records much of the linear excursion range of the woofer is used to reproduce the large amplitudes of record rumble. This wastes driver linearity. Fortunately the left and right-channel vertical rumble outputs from the pickup are out of phase and therefore cancel when the left and right channels are summed for a center channel woofer, as in this design. Separate left and right channel woofers can easily be tied together electrically to eliminate the unnecessary movement of air at subsonic frequencies from one speaker box to the other[12].

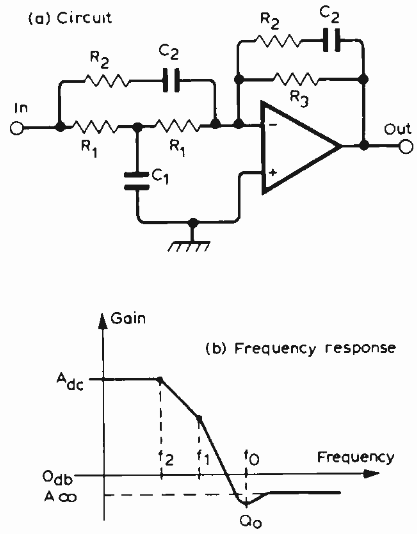
The corrected response of the woofer can be verified by placing a microphone about 1cm away from the cone to determine the near-field sound pressure which for a uniformly moving piston is proportional to the far-field sound pressure[13].

System equalization

As active networks are already used for the crossover filters it seems attractive to also use them to equalize the complete speaker system for a flat amplitude response at the preferred listening location. A microphone at that position will pick up the direct sound coming from the speakers and a large number of reflections from various objects and the walls of the room. The microphone cannot distinguish between the different sources. The microphone output voltage which corresponds to the direct sound from the speakers will be masked by the voltage due to the reverberant sound field. The ear-brain combination seems to be taking its clues for locating the details of the stereo image from the direct sound even when the reverberant sound energy is much larger than the direct sound. This might explain why attempts to equalize for a flat response at the listening location gave unsatisfactory results.

The response at one metre from the speaker measured in the room appears to be a better starting point for equalization. But even for this location a completely flat response does not seem to give the most natural-sounding reproduction. Some form of shelving or sloping response seems necessary[14].

In this design a 3dB low-frequency boost is applied to the B110 signal to obtain flat acoustic output over its range (last stage in the centre channel of Fig. 13). The T27 is allowed to follow its own gradual roll-off, but if a flat high-end response seems desirable then the simple network shown with broken



(c) Design formulas

$$f_o = \frac{1}{2\pi R_1 \sqrt{C_1 C_2}}$$

$$Q_o = \frac{1}{2\xi} = \frac{R_1}{2R_1 + R_2}\sqrt{\frac{C_1}{C_2}}$$

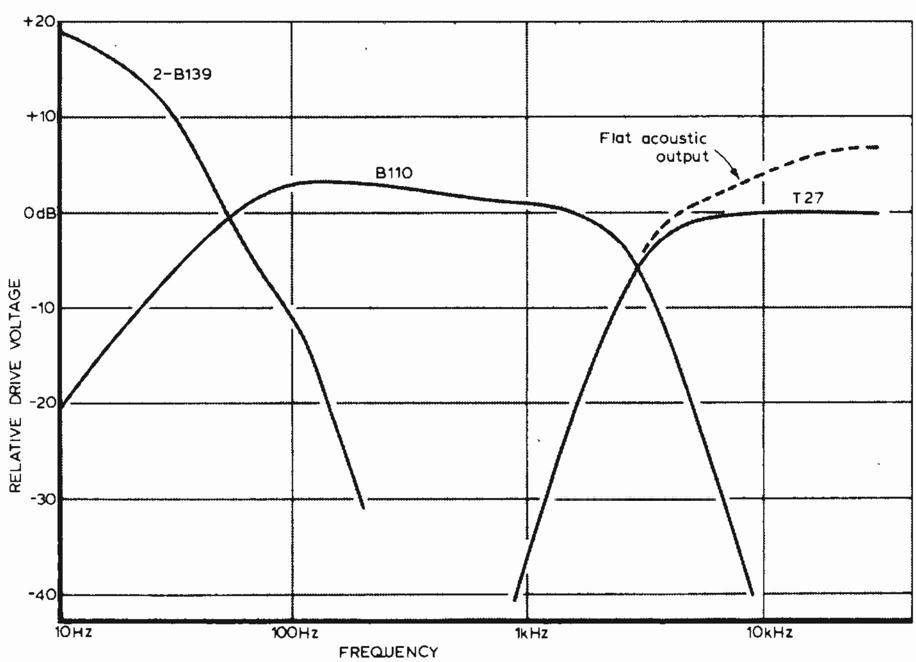
$$\frac{R_2}{R_1} = \frac{1}{Q_o}\sqrt{\frac{C_1}{C_2}} - 2 \qquad A_{dc} = \frac{R_3}{2R_1}$$

$$f_1 = \frac{1}{\pi R_1 C_1} \qquad A_\infty = \frac{R_3}{R_2 + R_3} < 1$$

$$f_2 = \frac{1}{2\pi (R_2 + R_3) C_2} \qquad \frac{A_{dc}}{A_\infty} = \frac{R_2 + R_3}{2R_1}$$

Fig. 20. *Network extends the woofer low frequency response to f_1 by providing exact compensation for Q_o and f_o with schematic amplitude response, and design formulas.*

Fig. 21. *Measured voltages at the driver terminals of the complete system. A flat response does not seem to give natural-sounding reproduction.*



lines at the input stage will give the necessary high-frequency boost.

An analogy might help to describe the subjective impression of a properly designed and equalized system by comparing it to the colour photograph of a familiar scene. A fair sound system might correspond to an out-of-focus picture, possibly with the wrong reds and blue or an overall colour tint. Comparing two such systems to each other is like looking at two blurry pictures of reality, where one might prefer one over the other because of its colour balance but there is no question of either being a realistic reproduction. A high accuracy sound system corresponds to a photograph which is focused and without unnatural emphasis on any colour. When a high standard of reproduction is being approached it becomes possible to hear clearly areas of slight imperfection like in a picture which is not exactly focussed or has just a slight colour tint. For the high-frequency equalization of the speakers this means that too much output shifts the sound image out of focus. The image depth becomes blurred because the high frequency overtones seem to be less distant than the virtual sources which generated them.

The chosen speaker equalization appears to match the greatest variety of program material. A properly functioning treble control in the preamplifier is needed though to correct for differences in recordings. The final response of the drive voltages for the three speaker units, Fig. 21, could have been generated or approximated with passive networks. The practical implementation might prove to be difficult though and no attempt has been made to design a passive crossover/equalizer. The design flexibility of active networks far outweighs the possible cost saving of passive networks when only a single system is being built.

Conclusion

It is hoped that some of the design techniques and ideas expressed here will stimulate a more rational design of loudspeaker systems. Certainly the drivers will be continuously improved for reduced spurious resonances but even more so the enclosure design, materials and shapes will need further study and development[15]. Nevertheless it is possible to design a highly satisfactory system even with today's standard components.

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Siegfried Linkwitz became interested in sound reproduction as a hobby, recognizing the many dualities between the propagation of microwave and acoustic fields. "A 1GHz electrical wave and a 1kHz acoustical wave have about the same wavelength." After modifying and equalizing several commercial loudspeakers, he set about designing his own system — out of frustration with available units.

He is program manager for a high performance microwave spectrum analyzer with Hewlett-Packard, and he's been involved with the design of signal generators, a vector-voltmeter (8450A) and was project leader for a spectrum analyzer (8554L). Joined Hewlett-Packard Company in California as a development engineer for r.f. test equipment following graduation in 1961 from Darmstadt University.