Loudspeaker system design

Three-enclosure system with active delay and crossovers

by Siegfried Linkwitz

This detailed description of a multiple-driver loudspeaker design is in two parts and covers driver selection, enclosure design, the active crossover, equalization and positioning. Sufficient information is given to duplicate the system or to improve existing systems by equalizing the low-frequency response or adding a separate woofer box.

THE SYSTEM DESCRIBED has evolved over years and out of experimentation with many different configurations and types of drivers and enclosures. Many people have contributed their ideas. It is not "the ultimate loudspeaker", but it reveals enough about microphone placement and recording practices to suggest, that the recording studio is the next weak link in the chain between original and reproduction. The few recordings with good spatial definition are proof that the full potential of stereo has not been exploited. Possibly this potential has gone unnoticed because hardly any commercial loudspeaker reproduces the depth perspective adequately, giving either a diffuse or thin-walled stereo image.

Every driver becomes more directional as frequency increases. The radiation pattern of a rigid piston mounted at the end of a long tube[1] is omnidirectional at frequencies where the ratio of piston diamter d to the wavelength λ of radiated sound is small, Fig. 1. As d/λ increases the on-axis pressure increases but the pressure at 45° off-axis decreases relative to it. Experience shows that wide dispersion of sound is desirable for natural reproduction. Allowing for a maximum 6dB drop-off at 45° off-axis requires that a driver be only used over a frequency range where its equivalent piston diameter is less than one wavelength. This is an idealized assumption because real drivers do not behave exactly like rigid pistons but the general principle still holds that uniform, wide dispersion can only be expected for frequencies where $d/\lambda \leq 1$.

In all loudspeaker designs the physical dimensions of driver, box and room have to be compared to the wavelength of the radiated sound to determine whether a dimension is acoustically small, as when $d/\lambda < 0.5$, or large, Fig, 2. A 200mm diameter driver for example should only be used up to 1.5kHz to maintain wide dispersion.

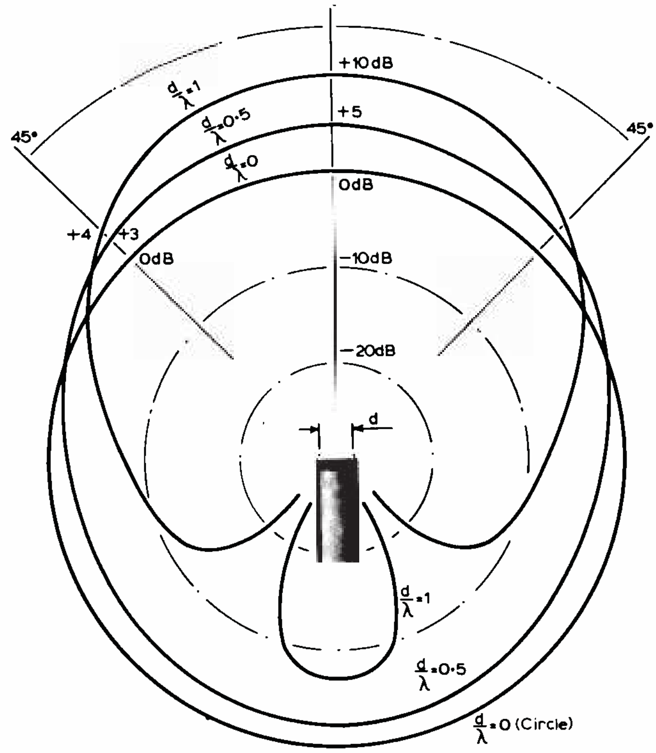


Fig. 1. *Directivity pattern for a rigid circular piston in the end of a long tube as function of d/λ (d is piston diameter, λ is radiated sound wavelength). Wide dispersion can only be obtained for frequencies where d/λ ≤ 1.*

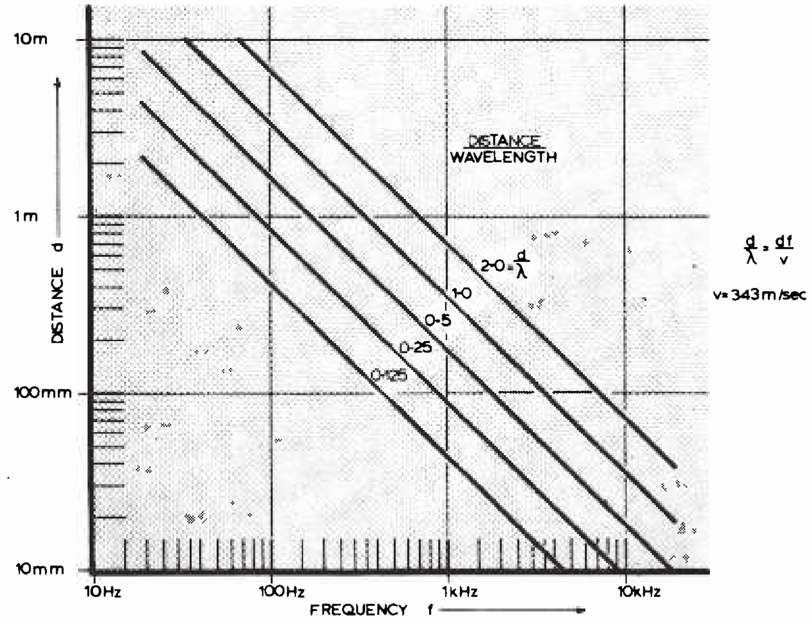


Fig. 2. *Dimension (d) of driver, box or room must be compared to wavelength of sound to determine whether a dimension is acoustically small.*

This is indeed a popular crossover frequency but it is also well within the critical frequency range of fundamentals and lower harmonics of many musical instruments. It is unavoidable that some change in the radiation pattern is introduced around the transition frequency from a larger to a smaller driver. In the design described a 100mm diameter driver is chosen which crosses over to a 25mm diameter unit at 3kHz. The radiation pattern change occurs therefore an octave higher in a relatively less critical frequency range, but still care has been taken in the design of the crossover circuitry to minimize irregularities in the transition region.

Some designers try to obtain wide dispersion or "omni-directionality" by using multiple drivers, covering the same frequency range. The fallacy in this approach can be seen by comparing the radiation pattern of a single driver to the resulting dispersion when two of these units radiate together, Fig. 3. If the distance d between the drivers is greater than half a wavelength signal cancellation can occur. The two outputs will be 180° out of phase whenever the path lengths from each of the drivers to the listener differ by an odd multiple of a half wavelength. As frequency increases the two units move relatively further apart (Fig. 2) and the locations for which the outputs cancel become more frequent. Such a system can only be described as multi-directional. Additional drive units further destroy the phase coherence of the direct sound output from the speaker system. This imparts the illusion of wide dispersion to all program material but lacks the accuracy in sound perspective which can be obtained from a single drive unit.

After establishing from the cone diameter the highest frequency up to which a driver can be used with good dispersion, the lower frequency limit will be determined from the cone excursion capability of the drive unit and the desired sound pressure level.

The radiation from the piston in a long tube was found to be omnidirectional for low frequencies where $d/\lambda \ll 1$, Fig. 1. If the piston moves with a peak-to-peak excursion a_{pp} at frequency f and radiates into free space, then the pressure p at a distance r from the source is

$$p = \frac{\pi^2 \rho_0}{8\sqrt{2}} \; \frac{a_{pp} f^2 d^2}{r}$$

Normalizing the pressure with respect to the reference pressure $p = 2 \times 10^{-4} \mu$ bar yields an expression for the more familiar sound pressure level (d and a_{pp} increased in mm)

$$20\log(p/p_0) = -86 + 40\log f - 20\log r + 40\log d + 20\log a_{pp}.$$

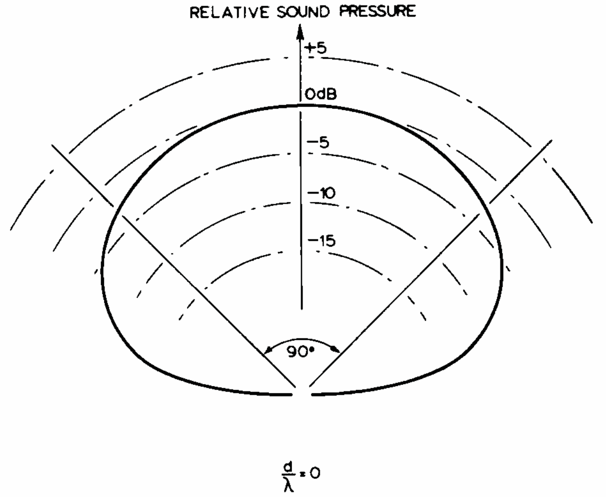
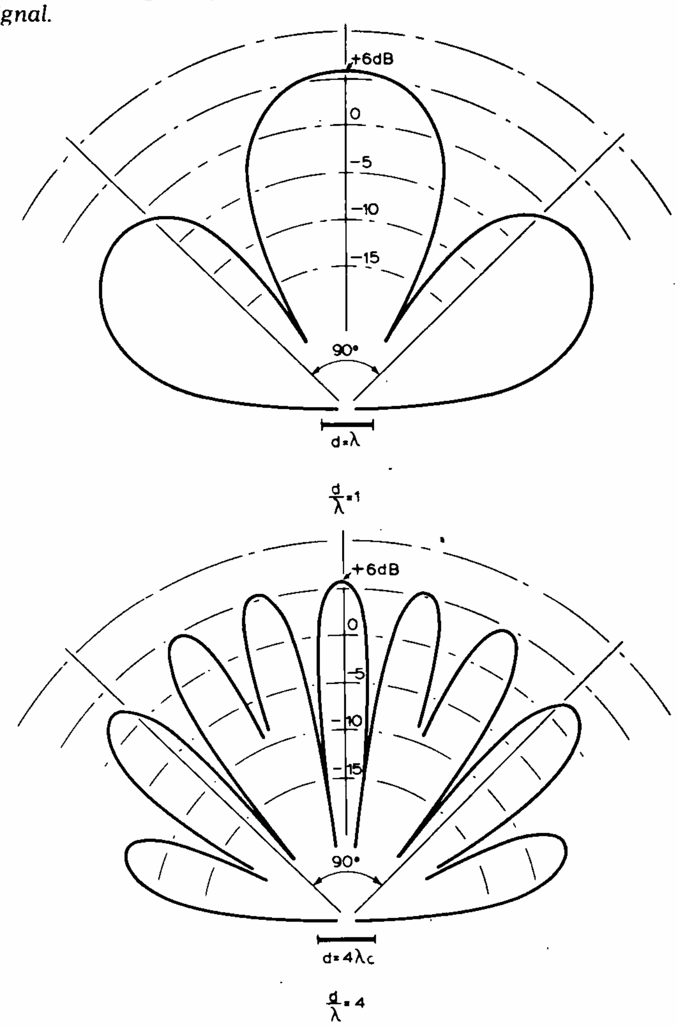


Fig. 3. *Use of multiple drivers to give wide dispersion produces interference effects for d>0 (a) single driver, (b) two drivers one wavelength apart, (c) two drivers four wavelengths apart, all with same signal.*



Assuming a_{pp} is 6mm and f 70Hz a direct pressure level of 83dB at 1m can be obtained from the 100mm driver and 85dB from a 200mm unit.

These s.p.ls may not seem very high but crossing over to a woofer at 70Hz will double the maximally obtainable sound pressure (+6dB) and because signals from the left and right channels of a stereo system are predominantly in phase at such low frequency a further increase of approximately 6dB can be expected. Therefore from a stereo system with 100mm drivers a direct free-field s.p.l. of about 95dB can be expected. Furthermore the normal listening environment is a semi-reverberant room where sound is reinforced by reflections from walls and objects.

Practical experience confirms that a 100mm unit can handle program material down to 70Hz at adequate levels and low distortion. This moves the crossover to the woofer to a less critical frequency range and a single large woofer can be used to cover the remaining frequency range below 70Hz. The large woofer enclosure can be placed separately from the relatively small midrange and tweeter enclosures and still be acoustically close because d/λ is small. Further consideration is given to this aspect of the system design later.

The frequency range below 70Hz could be covered by two 200mm units which will generate 90dB of direct s.p.l. at 35Hz and 1m or two 250mm diameter drivers with 94dB s.p.l. assuming 6mm peak-to-peak excursion capability.

The particular drivers chosen for this design are the 100mm KEF B110 low frequency/midrange unit, the 25mm KEF T27 tweeter and the KEF B139 woofer. A different unit like the KEF B200 or some other make with adequate excursion capability and linearity could be substituted for the B139.

There are of course considerations other than dispersion and cone excursion which must enter into the selection of a drive unit, such as smoothness of frequency response, freedom from high Q resonances, minimum phase behaviour, and low non-linear distortion. Unfortunately few meaningful data are published by many manufacturers. Knowing the magnet weight and flux density is of little help. With some training though the ear can sort out those drive units that seem worth further investigation and the units chosen for this design proved to be very satisfactory.

The selection of drivers was primarily guided by the desite for wide, uniform dispersion and crossover frequencies as high and low as possible. Had emphasis been placed on high power output capability or lower non-linear and Doppler distortion then larger diameter drivers would have to be chosen, or the crossover frequencies shifted to a more critical frequency range. Wide dispersion can only be obtained from a small drive unit which will also have higher distortion than a larger unit. It appears though that psychoacoustically the increased distortion is outweighed by an improved sound perspective which gives a greater sense of realism. Some further investigation of this subject is needed.

Speaker enclosures

Usually the size of a loudspeaker enclosure is dictated by the required low frequency response and efficiency. A different approach is taken here where the enclosure is optimized for minimum secondary radiation over as wide a frequency range as possible. The low frequency output capability is treated as a separate problem.

The purpose of the enclousre is to control the radiation from the back of the cone. A closed box design is chosen as the simplest form of enclosure. If the largest box dimension is less than a quarter wavelength at the highest frequency from the driver then the box is acoustically small and the air volume inside the enclosure will act like a uniform soring. The box has to be made sufficiently stiff so that the internal air pressure changes will not deflect the walls and cause secondary radiation. The woofer enclosure can be made small relative to the 70Hz maximum frequency. It will therefore have no internal air volume resonances and resonances of the box panels can be pushed above 70Hz by crossbracing of the walls.

The enclosure for the B110 presents greater difficulties because of the wider frequency range covered. The volume inside the enclosure will exhibit cavity resonances which have to be eliminated. Acoustic energy is stored whenever one of these resonances is excited and gradually released after the excitation has been removed. Most of this acoustic energy exits through the cone: the speaker regurgitates its own characteristic box sound. Fortunately it is not difficult to dampen cavity resonances. The technique of filling the enclosure with long-fibre wool is well established and very effective.[2,3].

Another form of undesired secondary radiation comes from the enclosure walls themselves. The walls can be excited to vibrate by the internal air volume pressure changes, but more serious is the direct transmission of the mechanical vibration of the driver's cone to the enclosure. The walls then radiate the transmitted mechanical energy as sound, particularly when its frequency coincides with a panel resonance.

It is not unusual that more energy is radiated directly from the enclosure walls than from the cone at resonance frequencies. If for example the vibrating enclosure surface has ten times the area of the cone then its acoustic output will already equal that of the cone if it has only one tenth of the cone excursion. The output of most loudspeakers is coloured by radiation from the enclosure walls.

It has been verified experimentally that vibration coupling between the driver and the walls occurs primarily through the rigid mounting of the driver to the enclosure. Vibration-mounting the driver to the enclosure with some form of complaint suspension will significantly reduce the wall excitation, but it poses some difficult mechanical design problems. The natural frequency of the driver mounting has to be well below the acoustical output frequency. The mount has to seal the enclosure air tight and provide sufficient mechanical support for the driver. Another approach would be to enclose the box

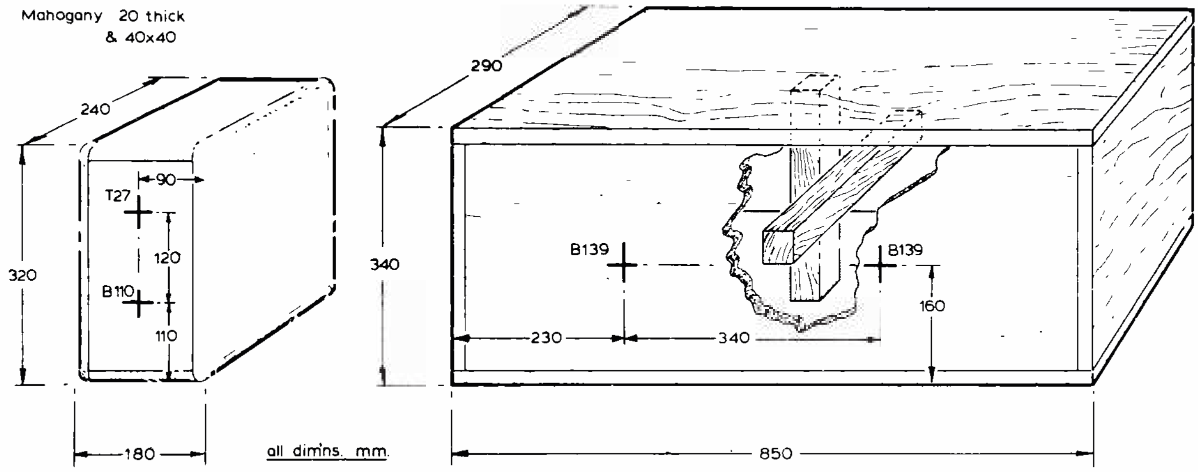


Fig. 4. *Loudspeaker enclosure dimensions. As there is little stereo information below the 70Hz limit of the enclosures (left), a centre woofer covers the remaining range down to 25Hz (right).*

to which the driver is mounted by a second box avoiding all rigid coupling between the two.

For the design described here a single, totally enclosed box was chosen, Fig. 4. The B110 driver was attached to it with soft rubber grommets in the four mounting holes of the basket and the sealing foam gasket barely compressed. Comparing this to a directly mounted driver by tapping on either basket indicates a significant reduction in coupling to the box. Some further investigation of this subject is in progress.

The relatively small size for the B110/T27 assembly has the advantage that the internal air volume resonances occur at high frequencies where they can be damped effectively with wool filling. The lowest cavity resonance occurs at 600Hz, the next ones at 800, 1000, 1200Hz etc. The resonances are measured easily with a small omnidirectional electret microphone protruding into the box and applying a sweep signal to the B110. Filling the boxes rather tightly with long-fibre wool attenuates all the resonances to a smooth frequency response inside the box and at the outside cone surface.

The boxes are constructed out of 20mm mohagony boards. The panels are quite stiff. The lowest panel resonance was observed at 430Hz using a magnetic phono pickup in a makeshift tonearm as vibration transducer. A 430Hz tone was measured to decay 40dB in 120msec after the electrical signal was removed. This indicates that the resonance was a Q of 36 according to $0.7f_R\tau_{40dB}$.

The Q is quite high and the decay time is long. By applying approximately two litres of roof patching tar to the inside of the box the resonant frequency was lowered to 300Hz due to the added mass to the panels. The decay time decreased to 40ms, corresponding to a Q of 8.4. While this treatment proved effective it does point out the problem that a small panel can have a high Q which is difficult to dampen because of its high stiffness and large mass. Better results might be expected from a thin plywood construction, with thick layers of damping material to attenuate resonances and to reduce the direct transmission of sound from the inside of the box.[4]. Ideally of course the panels should not be excited in their resonances at all, neither from the air pressure changes inside the box nor from the mechanical coupling to the driver.

A small box presents a small obstacle to omnidirectional sound propagation. This is a clearly audible advantage when properly placed in the room. As the box is only marginally wider than the B110 driver it can be assumed that the radiation pattern for a piston at the end of a long tube is an adequate first-order approximation to its sound dispersion, Fig. 1. The T27 tweeter is mounted as closed as possible to the B110. At the crossover frequency of 3kHz the spacing corresponds to a distance of one wavelength. In the vertical plane therefore the radiation pattern at the crossover frequency should follow the previously discussed behaviour of two drivers contributing equally, Fig. 3b.

Enclosure design objectives

- Narrow frontal area for optimum horizontal dispersion; tweeter mounted directly above the mid-range unit.
- Box edges rounded to reduce scattering.
- Drivers mounted to minimize direct transmission of vibration.
- Air cavity resonances attenuated with filling materials to eliminate delayed re-transmission through cone.

Ideally the sound from the T27 should be able to disperse freely in all directions, but because of the large width of the front panel relative to the cone diameter a wave emanating from the cone will initially be blocked by the panel and then encounter an abrupt transition where it ends, Fig. 5. A second wave is generated at the cabinet edge which will interfere with the original wave. If a pulse is radiated from the T27 then a secondary pulse of lower amplitude is generated at a time $t = d/c = 260\mu s$ later; the original pulse is smeared out. This scattering of sound should be avoided by eliminating sharp discontinuities through bevelling of the cabinet edges.[5].

Figure 6 shows the on-axis amplitude response of a small driver mounted in the center of a cube and of a sphere.[6] Clearly the sphere with its surface gradually receding from the source produces a much smoother response than the cube with its sharp edges. Therefore the larger box relative to the driver the more closely should it approach the shape of a sphere. It follows that the midrange/tweeter enclosures might be further improved by reducing their size and a more curved driver mounting area. Papier maché, cardboard or epoxy fibreglass with damping materials applied to it rather than wood might be more suitable

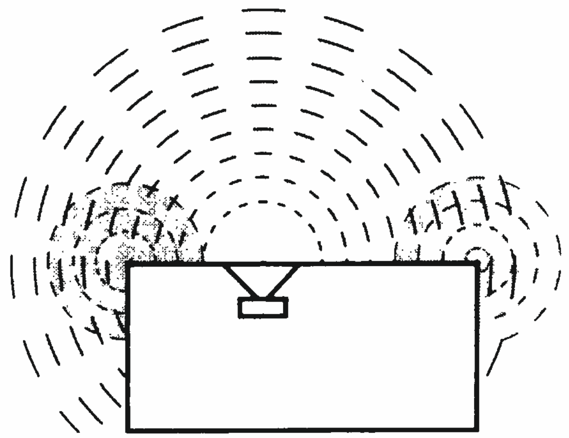


Fig. 5. *Sound scattering from the sharp corners of a loudspeaker enclosure producing a smeared out transient behaviour of the system.*

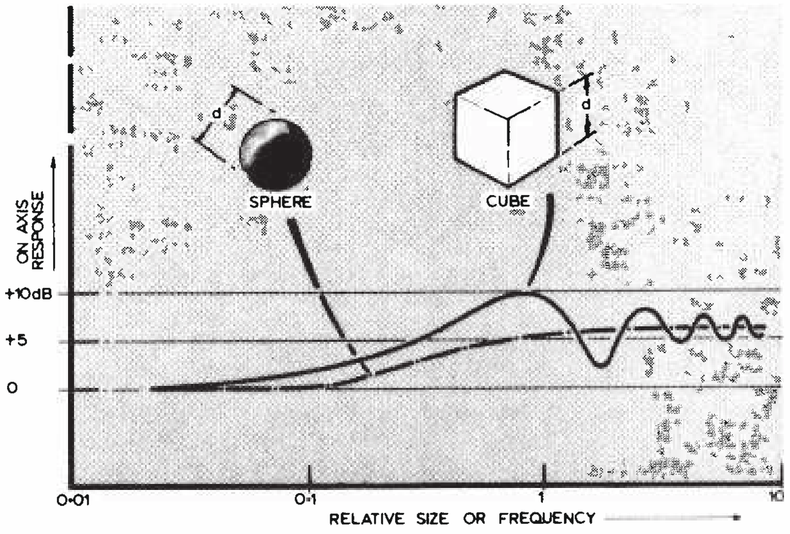


Fig. 6. *On-axis frequency response of a point source mounted in different enclosures shows smoother response of sphere (after ref. 6).*

materials for the unconventional contours of such an enclosure.

For the given design the frequency range extends down to 70Hz where the B110 has its enclosed resonance. As there is little meaningful stereo information below this frequency a single centre-channel woofer box can cover the remaining range down to 25Hz, Fig. 4. This is built with internal bracing to stiffen it and to push panel resonances to frequencies above 70Hz. In addition 25mm- thick heavy felt is glued to all panels to reduce direct transmission. The box is loosely fitted with long-fibre wool.

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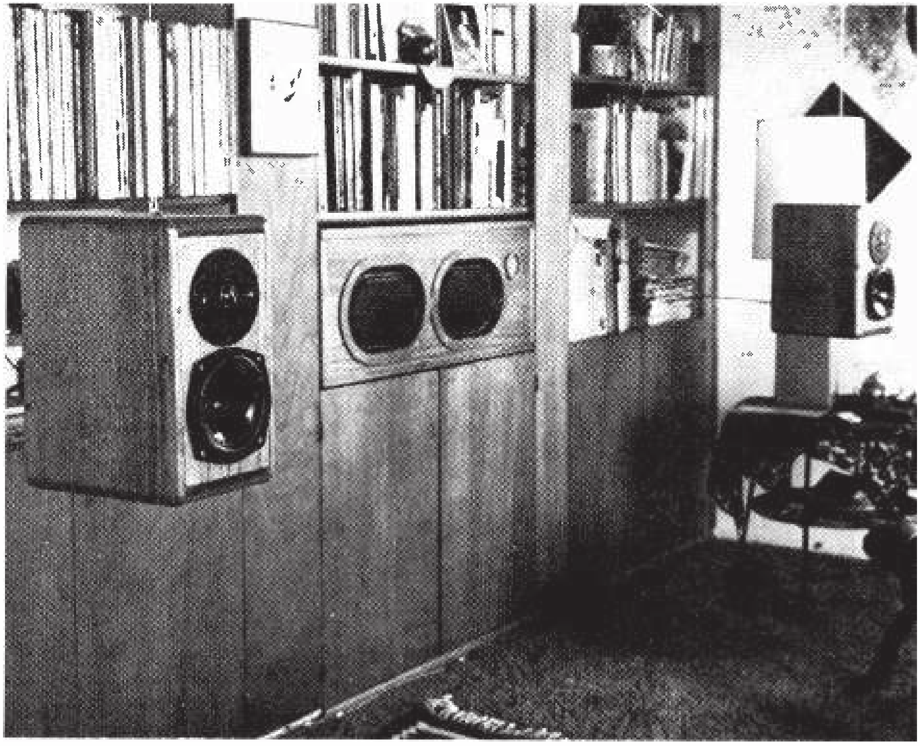


This article will be concluded with active crossover and equalizer designs and a discussion of speaker location.

Current broadcasting techniques "a hybrid rat-bag of improvisation"

TV engineers may propose a 1875-line tv service for Band III when 405 line transmissions are no longer needed, but it seems unlikely that the frequencies will be used for television, according to Dr Boris Townsend, head of the IBA's engineering information service. Speaking at the first of the three IBA lectures on tomorrow's broadcasting, he said, "I expect that we shall lose the use of these frequencies for television, which will be a pity, for band III has excellent propagation characteristics for our purposes. Nevertheless, I expect the engineers to put up a case for a new television service in the band using the latest advance in engineering terms: for example, better definition, 1875 scanning lines, larger pictures, tone gradation scales tailored to human physiology and, I hope, a certain technical flexibility to make television more helpful in teaching the handicapped child with abnormal physiology."

Speaking of the current state of broadcasting technology he said, "We are stuck with old-fashioned television transmissions which are a hybrid rat-bag of improvisation and almost incompatible technologies, which carry a quite unnecesssary amount of repetitive data yet still reproduce inadequate detail, inadequate contrast, inadequate colour gamut, and pictures which lack most of the attributes of real life. It is a system which gives the viewer and listener little choice and to which any graduate tea-boy engineer could now suggest improvement. What are we going to do about it?

"In the next few years, not much. We did something about our equally outmoded radio transmissions, and duplicated our use of precious wavelengths with our stereo f.m. transmissions using circular polarisation. These transmissions are glorious. What happened? Nothing. The public ignored them. Everyone still listens on antiquated amplitude-modulation receivers — statistically, so to speak."

Dr Townsend warned of the problems technology could bring if it were directed solely at increasing efficiency and reducing costs: "Perhaps unemployment *is* a more important matter than television automation. If it is, then should design engineers be devoting their thinking to making this public service of television an under-capitalised, more labour-intensive operation? It is a naive thought. But the basic dilemma is a matter of concern to many engineers." The problem of deciding among the bewildering array of technical possibilities "is not a problem which should be left to engineers." They were moving so fast in some areas, and not at all in others, "that we may be unable to prevent some massive mistakes on their part unless we speak quickly, and precisely, and regularly with them."

In questions after his speech he said of surround sound, "We can do it. Whether people want it is another matter, whether people will like it when they have got it. I think they will like it, but whether they will think it worth the trouble and the cost I'm dubious." There were also problems of production, and engineers had to get used to it to make the best use of the new forms of drama

Dr Townsend predicted that the changes at the transmission end of the broadcasting chain would be radical: "Engineers are making a two-pronged thrust forward. One is a miniaturisation of the analogue techniques which have been used in our studios since Savoy Place and Alexandra Palace, while the other is based . . . on digits." Both analogue and digital methods were dependent on micro-circuits. "The reduction in costs is staggering. An integrated circuit now costs about the same as an apple." Complex functions could be performed with devices that could be cheaply mass-produced, and circuits were getting, and would continue to get, even smaller, and so even more complex. While mechanisms grew more and more expensive, electronics became cheaper: "Compare the £70 monochrome television receiver, of dubious performance, which sold in 1938, with its contemporary 1938 £100 small motor car; and then take today's large, bright, reliable monochrome television receiver, still at the same £70, and its contemporary £2,000 Mini. So," he added, "whenever we can replace mechanisms by electronics we shall."

Programme labelling for sound broadcasts

Electronic "labelling" of sound broadcast programmes is definitely on the way, according to a BBC spokesman, and the only problem is deciding the best way to do it. This technique (described by Duncan MacEwan in "Radio in the '80s" in the May 1977 issue) uses data signals associated with the broadcast signal to identify particular programmes or channels so that they can be automatically pre-selected at the receiver for, say, an evening's listening or recording purposes. An automatic "search tuning" receiver could be arranged to select programmes of a particular type. The labelling code signals could be transmitted on a subcarrier in or out of band or by frequency modulation of the a.m. carrier. ☐

Loudspeaker system design

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

by Siegfried Linkwitz Dipl. Ing.

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.

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.

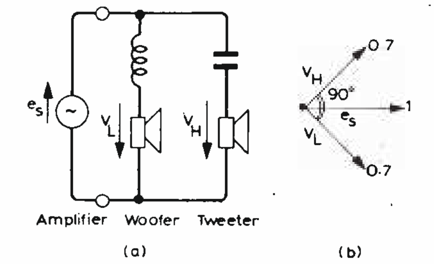


Fig. 7. *Schematic network with 6dB per octave slopes and voltage phasor diagram at the crossover frequency.*

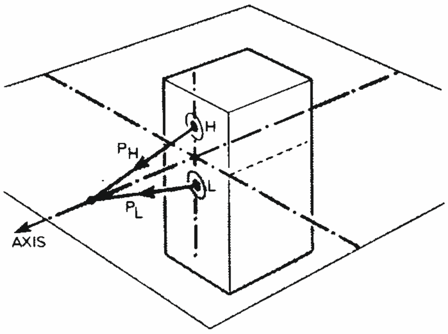


Fig. 8. *Plane of points are equidistant from the high frequency and low frequency drivers. The sum of the sound pressures is proportional to the sum of the electrical drive signals only in this plane.*

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.

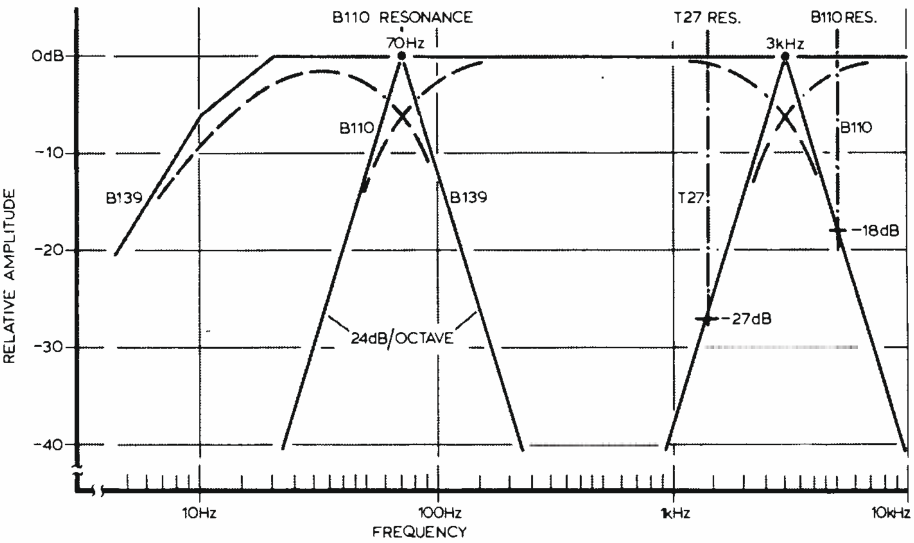


Fig. 10. *Schematic response for crossover points and driver resonances.*

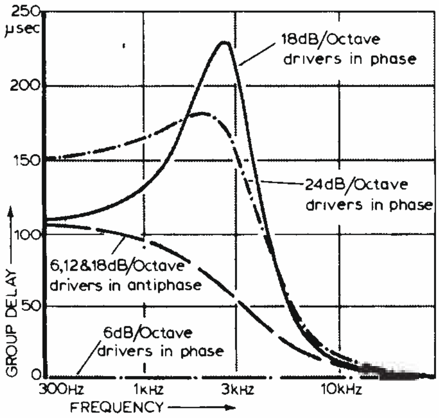


Fig. 11. *Group delay frequency response of a speaker system due to a 3kHz crossover between midrange and tweeter with conventional first and third-order Butterworth networks, and with second and fourth-order cascaded Butterworth sections (ref. 10).*

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.

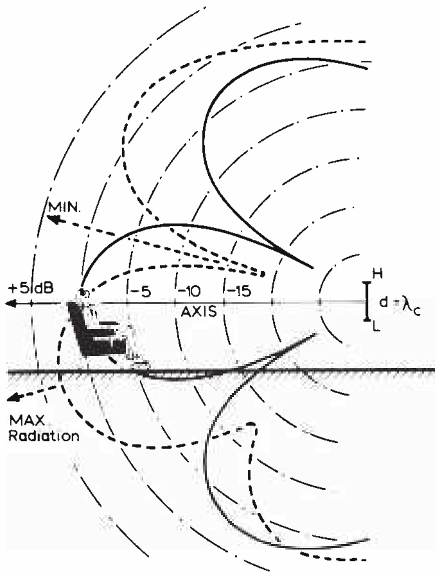


Fig. 9. *Radiation of a 6dB per octave crossover network at the cross-over frequency (3dB peak occurs below the plane of equidistance for non-coincident drivers) and the symmetrical pattern of a 24dB per octave crossover network at the crossover frequency (ref. 10).*

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. 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

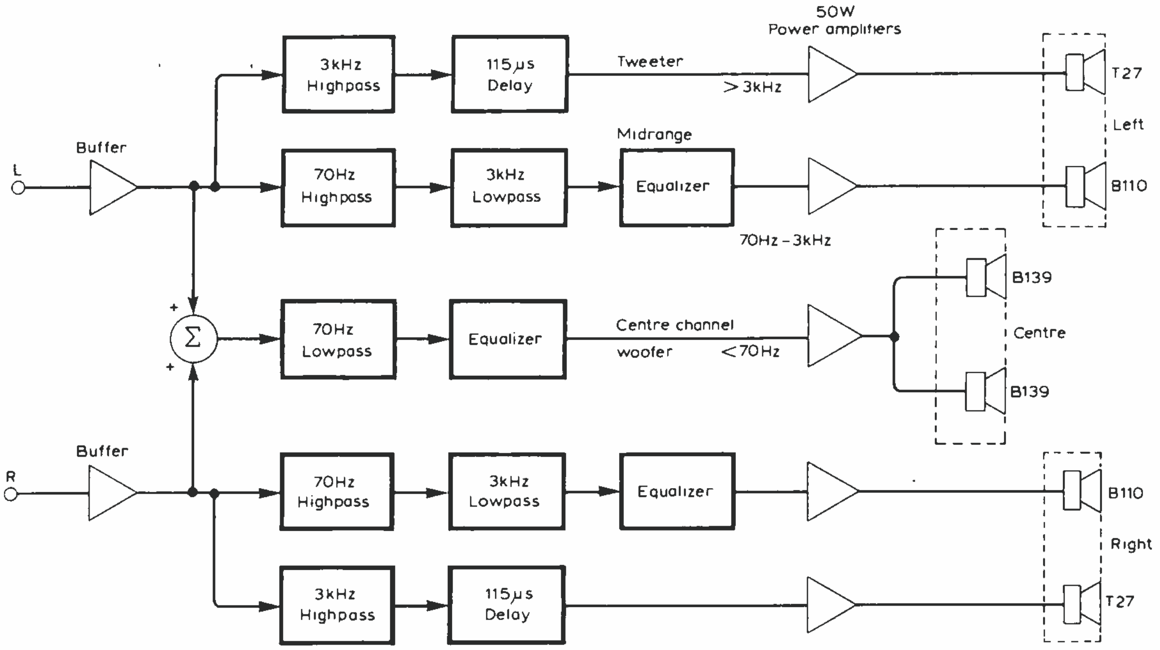
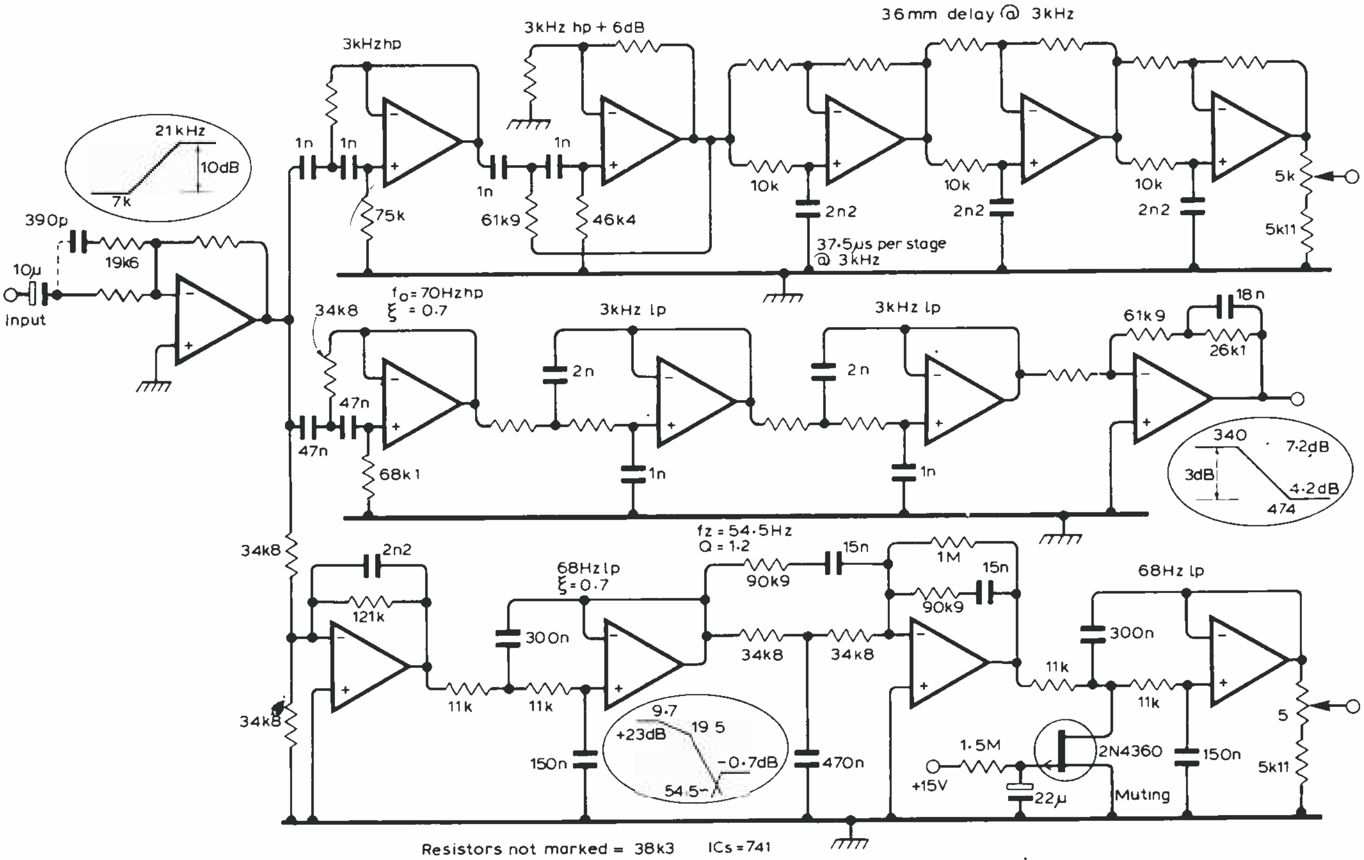


Fig. 12. *System block diagram. Design formulas for each functional block are given to allow adaptation of the circuits of Fig. 13.*

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



Resistors not marked = 38k3 ICs=741

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 $RC \leqslant 1/20f_c$. 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.

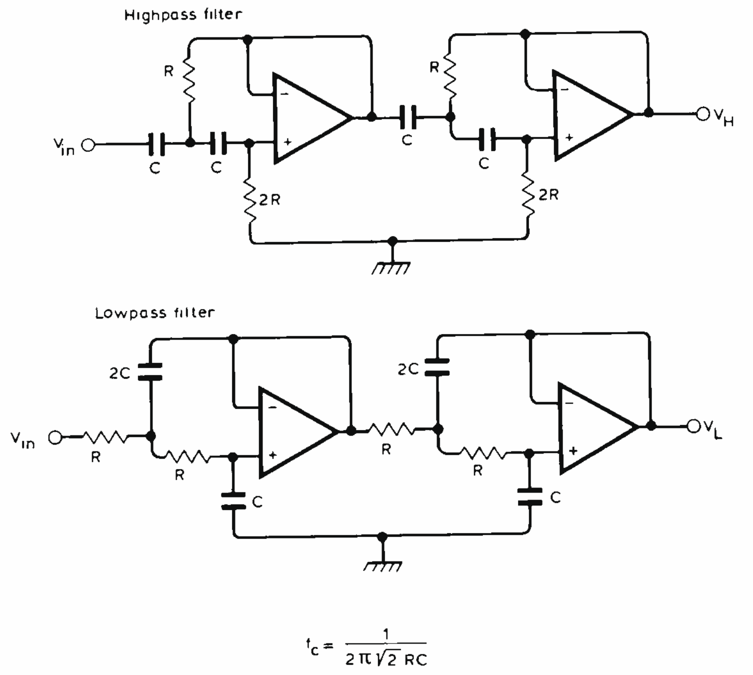
Woofer equalization

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.*

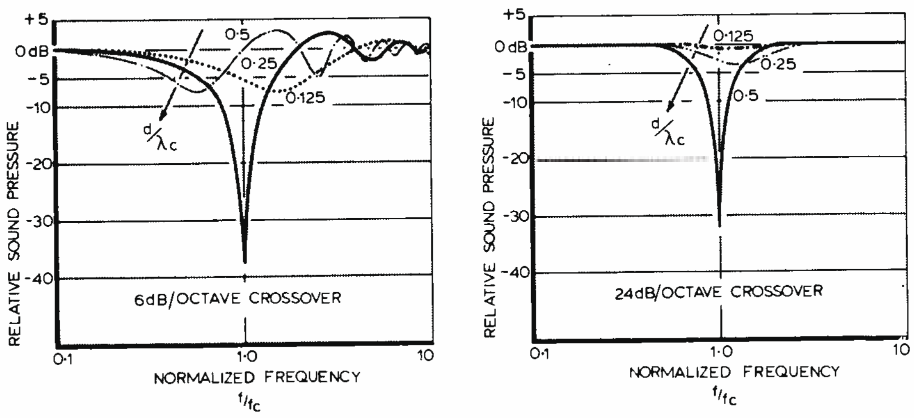
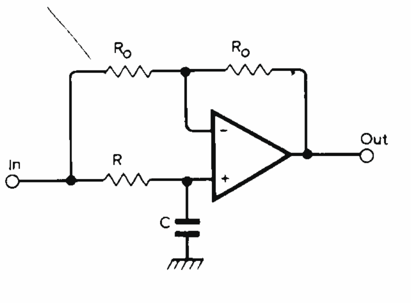


Fig. 15. *On-axis response when the tweeter is positioned acoustically in front of the midrange by* d/λ_c *with 6dB per octave crossover, and 24dB per octave crossover.*



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

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

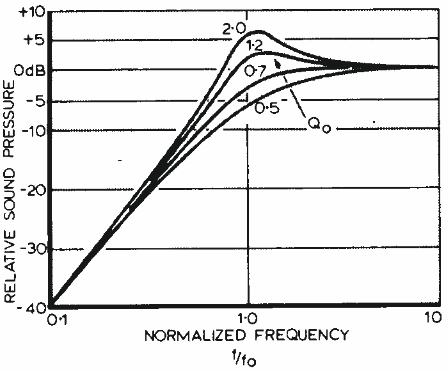


Fig. 17. *Fall-off in response of a rigid piston in a closed box (ref. 11). Box resonance f_o and Q are determined as in Figs. 18 and 19.*

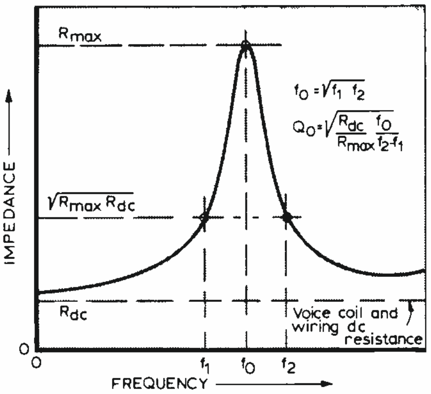


Fig. 18. *Schematic response of the woofer driving point impedance measured as in Fig. 19 from which f_o and Q_o of Fig. 17 are derived (ref. 11).*

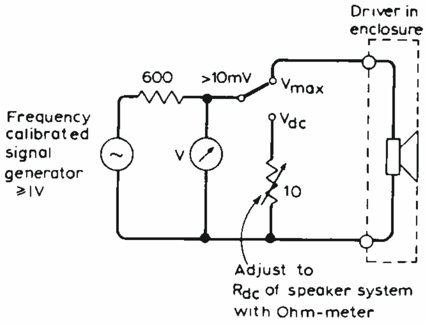


Fig. 19. *Measurement setup for Fig. 18 to determine R_{DC}/R_{MAX} from V_{DC}/V_{MAX} and the frequencies f_1 and f_2 from $V=\sqrt{V_{MAX}\times V_{DC}}$*

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].

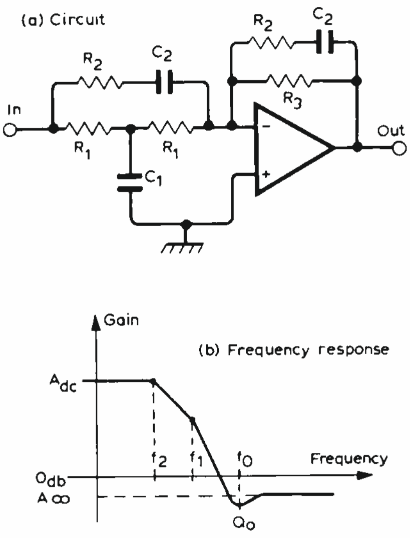
The corrected response of the woofer can be verified by placing a microphone about 1cm away from the cone to determine the near-field sound pressure which for a uniformly moving piston is proportional to the far-field sound pressure[13].

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

$$f_o=\frac{1}{2\pi R_1\sqrt{C_1C_2}}$$

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

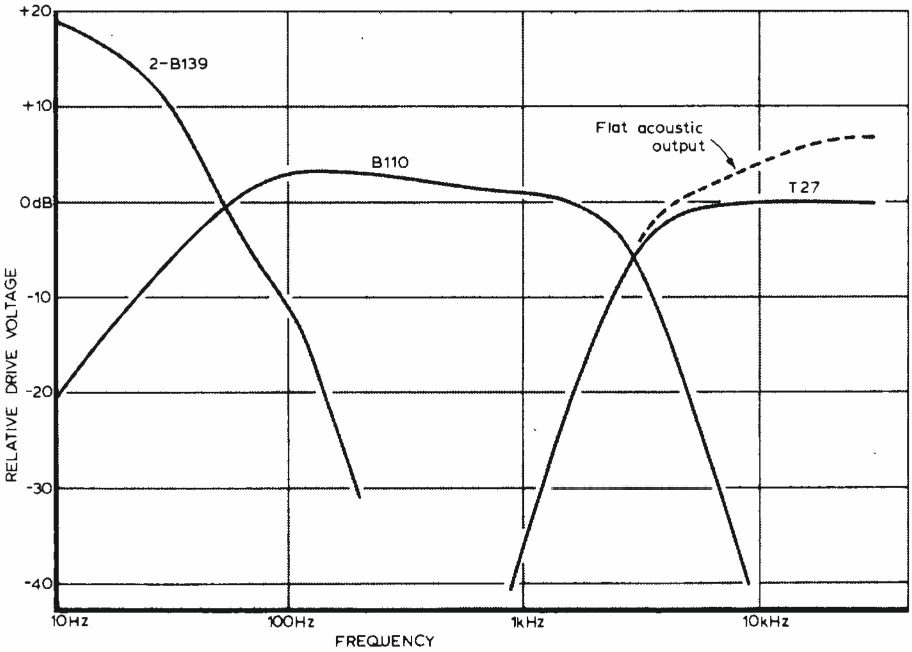
$$\frac{R_2}{R_1}=\frac{1}{Q_o}\sqrt{\frac{C_1}{C_2}}-2 \qquad A_{dc}=\frac{R_3}{2R_1}$$

$$f_1=\frac{1}{\pi R_1C_1} \qquad A\infty=\frac{R_3}{R_2+R_3}<1$$

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

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.

System X speed-up

An Integrated Systems Development (ISDD) has been set up by the Post Office to accelerate the development of System X, the electronic exchange equipment "that will revolutionise the nation's phone system." *(Wireless World, passim)*. The department is to be headed by Mr John Martin, who has worked on electronic exchange equipment for 15 years, and on System X for four.

In a statement the BPO said they intended to place contracts for the first production exchanges before the end of the year. The first exchanges would come into service by the early 80s. The new department is an offshoot of the TSSD (Technical Systems Strategy Department), set up in 1974 which, under Roy Harris, has been responsible for the overall design of the system. ISDD will liaise with Plessey, GEC and STC, who are developing System X with the PO, and other Post Office departments.

The Carter Committee on the running of the Post Office criticised the slow development of System X, which it feared would arrive too late to be competitive in world markets.

The Post Office revealed some of the ideas underlying System X at the Communications 78 exhibition and conference at Birmingham.

System X will use integrated digital transmission and switching, stored programme control and common channel signalling. The devices used would be based on low power Schottky t.t.l., c.m.o.s. and n.m.o.s. □

IERE hits out at "degree obsession"

"The national obsession with education to 'degree standard' has deprived the traditional technician engineer training pattern (ONC/HNC) of both its status and much of its best raw material; this, coupled with the inflexibility of CEI's qualification rules and procedures which has robbed the technician engineer of the facility to proceed by practical achievement to chartered engineer status, has divided the electronic engineering profession into a rigid two-class structure to the detriment of the profession and of the national interest."

This broadside was delivered in the IERE's evidence to the Finniston Committee on the future of the engineering profession. In other respects, however, the IERE seems to agree with the views of the Engineering Professor's Conference that degree courses should be longer and that there should be more co-operation with industry. The IERE recommends that there should be an extensive publicity campaign aimed at raising degree course entry standards to restore the status of ONC, HNC, HND and TEC courses and the associated work levels, to stimulate recruitment and to boost the morale of "this vital element of the engineering workforce."

This, as we said in our September, 1977 leading article, is more like it. □

Nabbed by satellite

American GEC has demonstrated a system for preventing drugs being smuggled over remote parts of the Mexican-US border which uses mobile radio and a geostationary satellite.

Loudspeaker system design

Changes and refinements to the system described in the May and June issues

by **Siegfried Linkwitz,** Dipl. Ing., Hewlett-Packard Co., Santa Rosa, California

It is unfortunate that there are still so few loudspeakers commercially available which achieve a high standard of accuracy, according to Mr Linkwitz. "After all," he says, "the design concepts are rather straightforward and rational." The design of a loudspeaker system has to include a large number of electrical, mechanical and acoustical parameters if optimum results are desired. There is not one single parameter which by itself will turn a poor loudspeaker into a superior one; attention has to be given to all parameters, including the driver, enclosure and crossover.

THESE NOTES are intended to encourage further development of loudspeakers, and bring increased enjoyment to those who want to undertake the task of building their own systems. The changes and refinements made to the original loudspeaker system, described in the May and June issues, are presented to show the completeness of the analytical design approach, and should not be taken as an indication that the previous system is obsolete. The audible effects of the changes are subtle and the added complexity of the circuits would be worthwhile only to someone trying to achieve greatest accuracy of reproduction. But the techniques described should be of general interest to any loudspeaker design.

I feel the weakest link in recreating the illusion of a life source with loudspeakers lies at the microphone pick-up end of the signal chain. It seems likely that more than two loudspeakers are needed, but first a much better understanding for recording and reproducing the appropriate sound field has to be developed and demonstrated. Then it may be possible to transport oneself to Symphony Hall without moving out of the living room chair. Meanwhile the loudspeaker as the necessary electro-acoustic transducer can approach a high state of development.

Any moving coil driver has the general frequency response of Fig. 22 (Fig. 17 ref 16*) when driven from a constant voltage source. This is a second-order filter with an asymptotic slope of 12dB/octave below the resonant frequency f_o and flat sound pressure output above it. The height of the peak near f_o is governed by Q_0. Both parameters f_o and Q_o are easily determined from an impedance measurement of the driver, Fig. 18. This general transfer function between terminal voltage and sound pressure output applies to woofers, mid-range units and tweeters as long as their cone dimensions are small acoustically, Fig. 2, and must be taken into account when designing a crossover network.

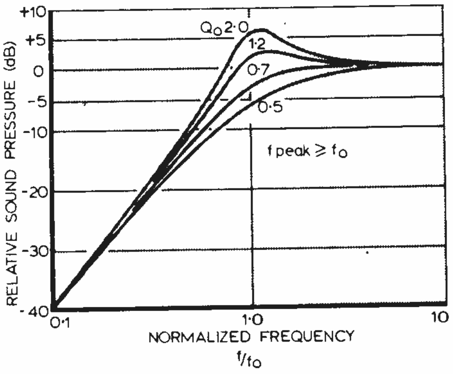

Fig. 22. *Frequency response of a moving-coil driver with dimensions small compared to a wavelength, which must be taken into account when designing crossover networks.*

As an example, consider the high-pass section of a crossover to a 25mm dome tweeter which has a resonance of 800Hz with Q_o of 0.9, Fig. 23(a). The desired acoustic output should follow the fourth-order high-pass characteristic of the 24dB/octave crossover with 1.5kHz as the −6dB crossover frequency (b). At first glance it seems sufficient to shape the driver terminal voltage to follow the 24dB/octave high-pass function of (b) because the

Fig. 23. *To achieve an acoustic or overall high-pass filter response with 24dB/octave slope (b), requires the terminal voltage to follow a 12dB/octave slope below resonance to compensate for the effects of the driver, whose sound pressure and phase response are shown at (a).*

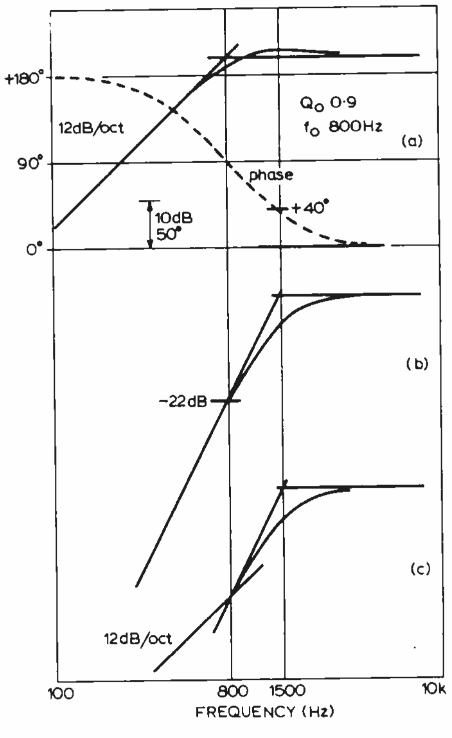

Fig. 24. *Required drive voltage (c) has to be constant below the driver resonance frequency f_o to give the desired acoustic h.p. response (b) (cone excursion shown dashed), as a result of driver response (a).*

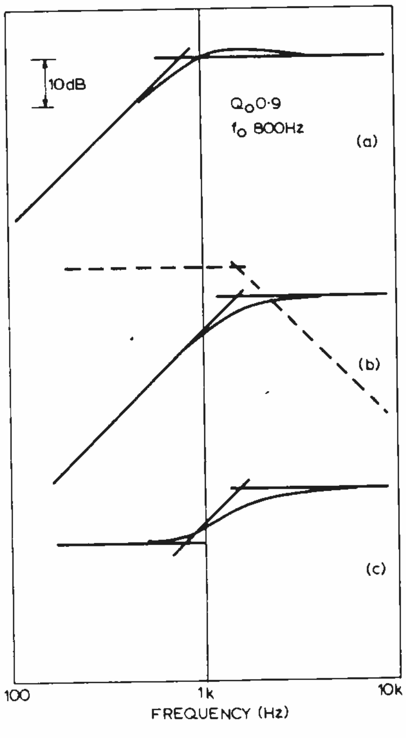

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*Figure numbers prior to 22 refer to the author's previous articles, ref. 16.

filter has 22dB of attenuation at the driver resonance. Indeed, this was the procedure in the original crossover design for the T27 tweeter, Fig. 10. Such terminal voltage, however, causes a 36dB/octave roll-off in acoustic output from the driver for frequencies below resonance f_0. To achieve the exact acoustic frequency response of (b) the terminal voltage must follow a 12dB/octave slope below the 800Hz driver resonance (c). This then compensates exactly for the phase shift and group delay which the driver would otherwise add to the acoustic high-pass function. The additional phase shift would cause a tilting of the radiation pattern as the sound pressures from the tweeter and mid-range unit would add to a maximum at a point off-axis[10]. The amount of the phase shift introduced by a second-order high-pass filter can be calculated for $Q_o \geqslant 0.5$ from

$$\phi = 180^0 - \arctan\left[2Q_0\frac{f}{f_0} + \sqrt{(2Q_0)^2-1}\right]\ldots$$

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

For the above example, the driver contributes 40° of phase shift at 1.5kHz. Sound pressures form the mid-range unit and tweeter are therefore not in phase unless the measures described are taken.

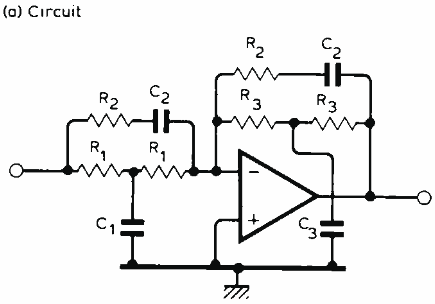
Driver terminal voltage

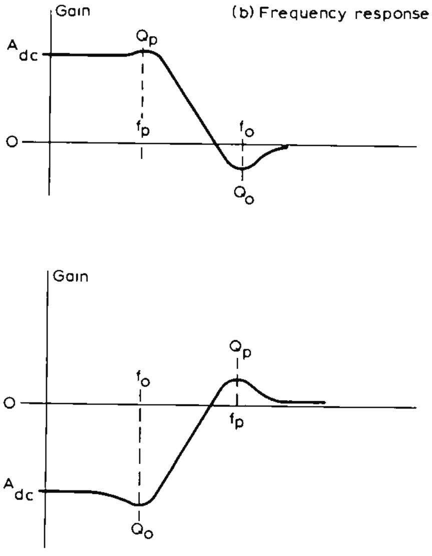
The acoustic high-pass function of the previous example requires an exactly-shaped terminal voltage to compensate for the driver's own frequency response.

A fourth-order high-pass response is equivalent to the cascade of two second-order Butterworth sections[10]. The first step then is to equalize the driver output to follow a second-order Butterworth function by shaping the terminal voltage applied to it, Fig. 24. Design formulas were developed for a very useful network, Fig. 25. It is a modification of Fig. 20 and will later be used also to extend the woofer response. A note to those familiar with the description of transfer functions by poles and zeroes in the complex frequency plane: This network will generate a pair of complex zeros (f_0, Q_0) which are positioned to cancel the complex poles of the driver (f_0,Q_0). In addition, a pair of complex poles (f_p,Q_p) is available which are placed at the crossover frequency in the case of the tweeter highpass or at the lower cut-off point of the woofer in the case of woofer equalization. The factor K in the design formulas is necessary for cancelling a pole-zero pair (f_{pl},f_{zl}) which would otherwise be introduced by the network.

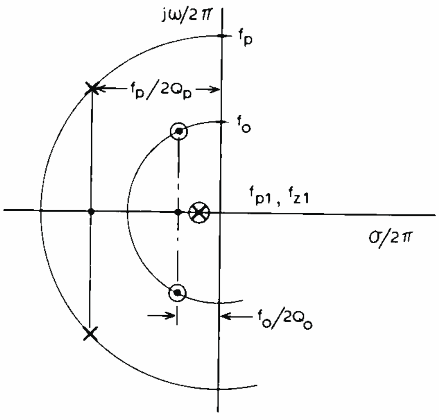
The second step in designing the acoustic high-pass filter is to follow this network with a standard second-order Butterworth section to achieve the overall drive voltage of Fig. 23(d). The complete circuit of Fig. 26 is only slightly more elaborate than Fig. 14 but it achieves the exact fourth-order acoustic output, Fig. 23(b).

(a) Circuit




(b) Frequency response

(c) Pole-zero location



(d) Design formulas

(1) Specify f_o, Q_o, f_p, Q_p

(2) $$k = \frac{\frac{f_o}{f_p} - \frac{Q_o}{Q_p}}{\frac{Q_o}{Q_p} - \frac{f_p}{f_o}} \quad k > 0 \text{ required}$$

(3) Choose C_2 (4) $R_1 = \frac{1}{2\pi f_o C_2[2Q_o(1+k)]}$

(5) $R_2 = 2kR_1$ (6) $C_1 = C_2[2Q_o(1+k]^2$

(7) $C_3 = C_1\left(\frac{f_p}{f_o}\right)^2$ (8) $R_3 = R_1\left(\frac{f_o}{f_p}\right)^2$

(9) $A_{dc} = 40 \log \frac{f_o}{f_p}$ [dB]

(e) Circuit analysis

$$\left.\begin{aligned} f_{p1} &= \frac{1}{\pi C_1 R_1} \\ f_{z1} &= \frac{1}{\pi C_3 R_3} \end{aligned}\right\} f_{p1} \approx f_{z1} \text{ required}$$

$$f_o = \frac{1}{2\pi R_1\sqrt{C_1 C_2}} \qquad Q_o = \frac{R_1}{2R_1 + R_2}\sqrt{\frac{C_1}{C_2}}$$

$$f_p = \frac{1}{2\pi R_3\sqrt{C_2 C_3}} \qquad Q_p = \frac{R_3}{2R_3 + R_2}\sqrt{\frac{C_3}{C_2}}$$

Fig. 25. *Useful network for compensating driver resonance at f_o and extending frequency response to f_p for woofer equalization or providing cut-off at f_p for mid-range or tweeter high-pass, responses. Calculated values should be checked with the circuit analysis equations.*

Crossover frequencies and drivers

The techniquie described could be used to modify the original T27 high-pass filter (f_o 1.2kHz, Q_o 1.1). Instead, I used a Son-Audax HD12×9 D25 soft-dome tweeter with a 1.5kHz crossover frequency to the B110. At 3kHz, the previous crossover point, the B110 cone diameter is about one wavelength, so that a certain amount of directionality can be expected, Fig. 1. Further, the mid-range and tweeter units are separated by one wavelength at 3kHz so that the combined radiation patter begins to narrow in the crossover frequency range, Fig. 3(b). The lower crossover reduces the acoustical dimensions by a factor of two so that a wider and more uniform dispersion is obtained over all frequencies in both the vertical and horizontal planes of radiation, Fig. 2. The loudspeaker then approaches more closely the acoustical point source. While the mid-range unit has to cover one octave less in frequency, the tweeter must now have four times the excursion capability to maintain the same acoustic output. The Son-Audax unit works well in this application and there is no sacrifice in overall smoothness of response compared to the T27. The new unit does not roll off towards the high end. For most commercial recordings a slight droop of about 3dB between 2k and 15kHz seems subjectively preferable and such response can be easily adjusted with properly designed treble controls.

The crossover point between woofer and mid-range units has been raised from 70 to 100Hz, thus reducing the maximum cone excursions for the B110 by a factor of two for constant sound output. Experience has shown that only the mid-range power amplifier is occasionally driven into clipping. If carefully fused a 100W amplifier might be considered for driving each B110. The three-way system is very forgiving to clipping of the mid-range amplifier. It is not audible on short transients because the woofer and tweeter channels still reproduce their undistorted portion of the total signal. The reduced frequency coverage of the B110 at both low and

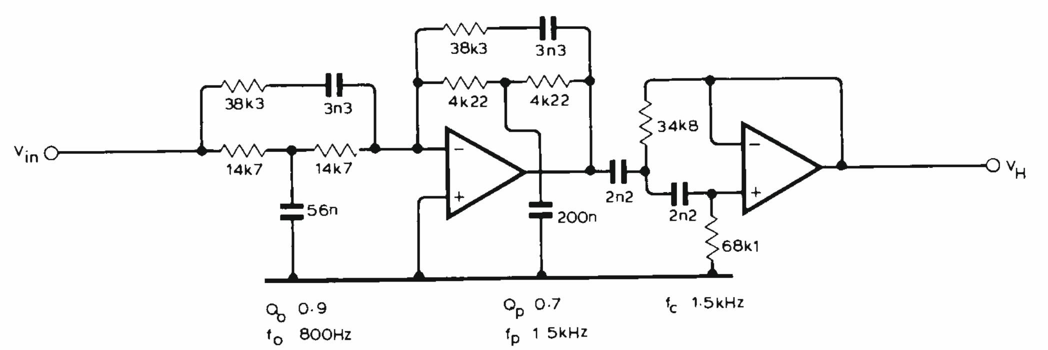


Fig. 26. *Network for a 1.5kHz 24dB/octave acoustic highpass filter for a Son-Audax HD12×9D25 dome tweeter. The first op-amp stage compensates exactly for the driver resonance at 800Hz and gives a 12dB/octave 1.5kHz acoustic high-pass response. The second op-amp stage is a conventional Butterworth section. Design formulas for this network are from Fig. 25 and Fig. 14.*

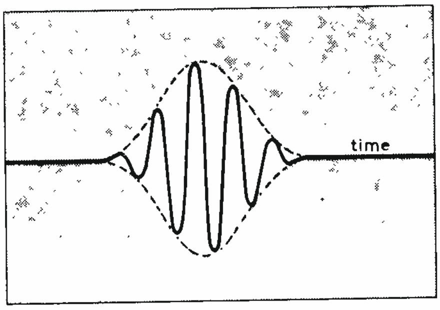


Fig. 27. *Shaped toneburst used to evaluate the audibility of phase distortion.*

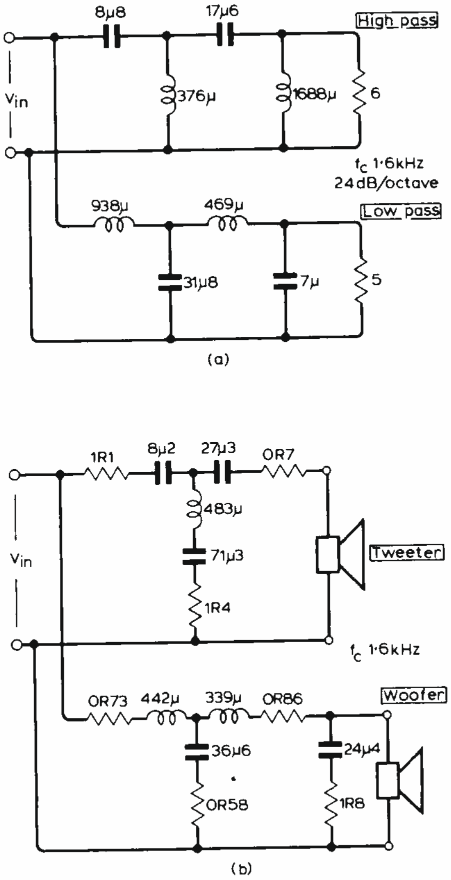


Fig. 28. *Passive network for 24dB/octave acoustic slopes and 1.6kHz crossover frequency. If constant terminal impedance is assumed for the drivers then the network and its element values can be determined analytically[20] (a). Computer optimized network for actual drivers shown at (b).*

high frequencies improves the amplifier power distribution between the drivers.

The crossover frequency between woofer and mid-range units was not raised further because the centre woofer is positioned 0.84m behind the mid-range unit and the phase shift due to this path length would become excessive. Further, the stereo effect might suffer from the blending of left and right-channel information for too high a crossover frequency.

In the future it could become necessary to have truly full range, separate speakers for reproducing an appropriately recorded sound field. Previously the mid-range resonance at 70Hz was used as one section of the 24dB/octave acoustic high-pass function. The second section was provided by an active network. Now, both sections are implemented electronically using the circuit of Fig. 25 to compensate for the B110's resonance in its enclosure, with f_0 and Q_0 determined from Fig. 18 (f_0 73Hz, Q_0 0.6). The complete network has therefore a configuration similar to that of the tweeter, Fig. 26.

Woofer equalization

The centre channel woofer covers a relatively narrow frequency range. Of particular interest is the lower cut-off point and cut-off rate. There is some indication that the low-end phase behaviour of a system can have audible effects. A 5Hz square wave for example, which sounds like a sequence of clicks, will change its tonal character when transmitted through an all-pass network[10]. From network theory it is known that any high-pass filter with a slope of more than 6dB/octave will produce some amount of ringing to a step input[17]. It is impractical to roll off the woofer at a 6dB/octave rate because it would mean that its cone excursion has to continue to increase at 6dB/octave even below the 3dB corner. The only practical way is to use a 12dB/octave rate. If the Q of this high-pass network is kept low at 0.5 then a minimum of overshoot is combined with a minimum of cone excursion.

The original network Fig. 13 is a good approximation. The revised crossover uses the circuit of Fig. 25 with f_p19.3Hz and Q_p0.5 which gives a 30Hz, 3dB corner frequency.

The high-pass nature of the woofer channel introduces phase shift at the 100Hz crossover to the mid-range unit according to the previous formula for Q_0 0.5:

$$\phi = 180^0 - 2\arctan\frac{f}{f_p} = 22^0$$

This amount of the phase shift by itself is insignificant, but combined with the phase shift due to the woofer location of 0.84m behind the mid-range it becomes necessary to add delay to the mid-range channel. It is implemented with the network of Fig. 16 which has a phase shift of

$$\phi = -2\arctan(2\pi fRC)$$

Both the absolute value of the phase shift and the slope of the phase curve, or the group delay, can be made to coincide between woofer and mid-range channel. The specific network component values R and C depend upon the set-up of the loudspeaker system and no compensation is needed when mid-range and woofer radiate from the same plane.

Audibility of crossover networks

Lowering of the tweeter crossover to 1.5kHz raised some concern over the audibility of phase distortion. The combined mid-range of tweeter sound pressure has all-pass characteristic. Sound pressures from the two drivers are in phase at all frequencies relative to each other but the overall sound pressure has a frequency-dependent phase shift relative to the electrical signal at the input to the crossover network. The group delay is not constant with frequency[10].

A new form of test signal was used which consists of a five-cycle tone burst of variable frequency. The tone burst is

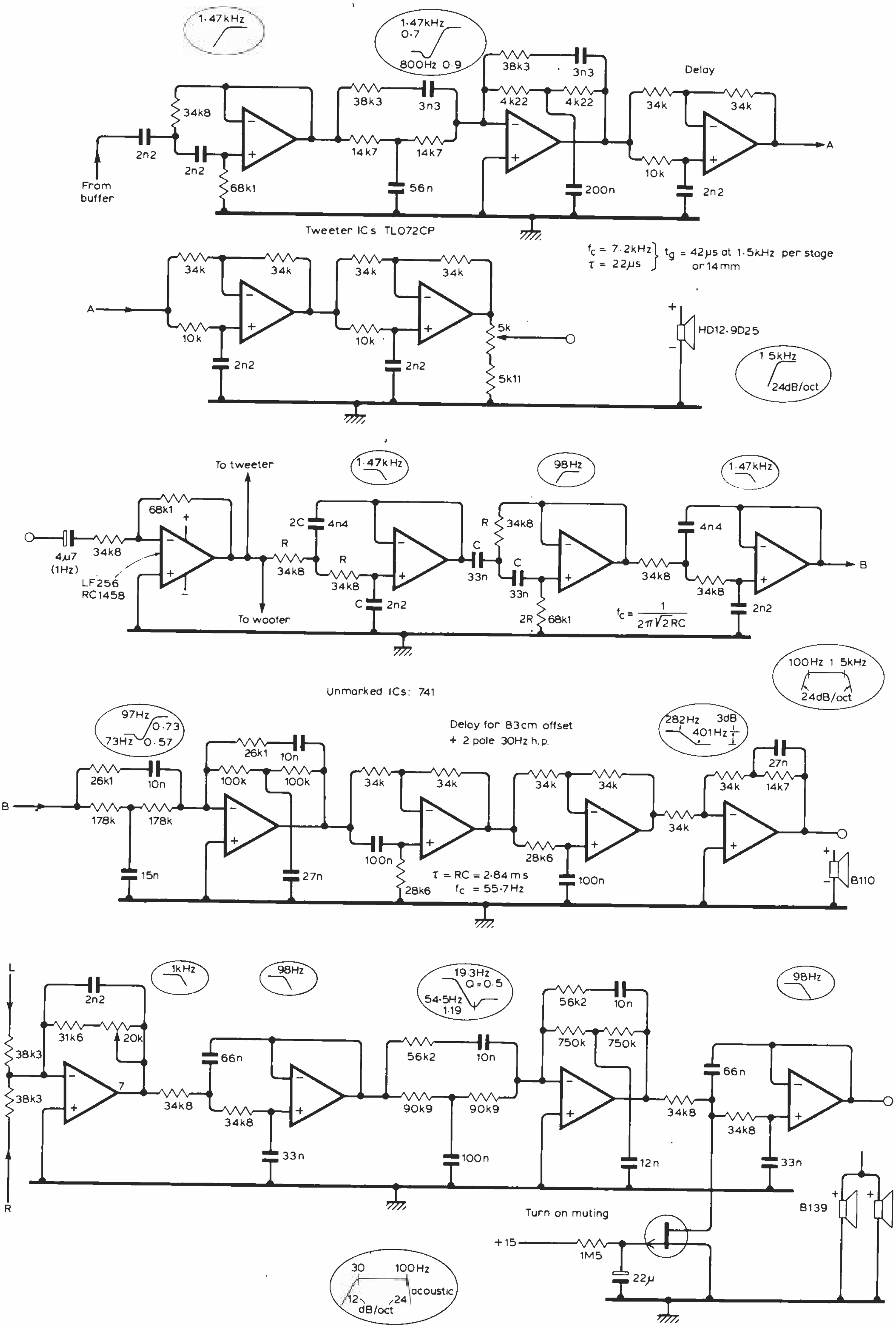


Fig. 29. *Complete schematic of crossover network and equalization of the author's system.*

not turned on and off in the usual abrupt fashion but instead it builds up and decays gradually, Fig. 27. The envelope of the burst follows a raised cosine function[18]. The spectral content of the shaped tone burst is concentrated in a narrow frequency range. The ear appears to be very sensitive to phase distortion of this signal, while a square wave or rectangular envelope burst are almost useless at higher frequencies for such tests. A system with 24dB/octave crossover filters has the phase shift of a second-order all-pass network with complex poles and zeroes of Q = 0.7. No audible change could be noticed on insertion of this network into the test signal path. The Q had to be increased to 2.4 before any effect was noticed with the test signal at 1.5kHz. Observation with an oscilloscope indicated ringing of the trailing edge of the shaped burst which became increasingly more audible as Q was raised above 2.4. It can be concluded safely from these tests and others with program material that the phase distortion of a 24dB/octave crossover is insignificant.

Often, claims are made for the superiority of low-order crossover networks with 6dB/octave slopes. It should be obvious from Fig. 24 that a 6dB/octave acoustic response cannot be realized with a passive network because the driver itself introduces a 12dB/octave slope and the aforementioned associated phase shift. Merely applying a terminal voltage which changes with 6dB/octave would guarantee an 18dB/octave slope below the driver resonance and 6dB/octave above it, but with excessive phase shift which defeats the whole phase argument for this type of network.

Even a 12dB/octave acoustic high-pass filter would be extremely difficult to achieve passively as can be seen from the required terminal voltage of Fig. 24(c).

The lowest-order acoustic high-pass filter which can be realized with a passive network has 18dB/octave slope, sometimes called an acoustic Butterworth[19]. This filter still suffers from the phase quadrature between low and high-frequency driver outputs and the resulting frequency-dependent irregularity in the radiation pattern[10]. Surprisingly then, the 24dB/octave crossover is the lowest-order function for which the all-important radiation pattern has a stable axis. So-called "linear phase" loudspeakers are based on wishful thinking and not on physical realities.

Enclosures

Further investigation into the constructon of a well-damped enclosure for the mid-range and tweeter led to the following conclusion.

A small box with 20mm thick walls is too stiff for tar-based damping layers. The tar has not enough stiffness of its own to control the motion of the panels at resonance. A better match between the two stiffnesses is required[20]. Building the enclosures out of 6mm plywood with a 15mm damping layer consisting of a 3:1 mixture of water-based roof patching tar and sand gave optimum results.

A simple and quite revealing test is to knock on any box to hear how dead acoustically it is.

Passive crossovers

Not everyone is at home with the electronics and the rather elaborate op-amp circuits for this loudspeaker system. A passive crossover seems attractive as it would consist only of inductors, capacitors and resistors in a relatively simple interconnection. Unfortunately it is considerably more difficult for the home constructor to arrive at the correct element values for a passive network than to design active networks with their great flexibility to change transfer functions and gain[19].

To design a passive network for a 24dB/octave acoustic crossover requires a computer optimization routine unless one is satisfied with the trial and error procedure on which most loudspeaker design has been based on up to today. If a driver could be represented by a resistor then exact network values are easily calculated[21], Fig. 28(a). Real drivers have complex terminal impedances, Fig. 18. This not only affects the component values of the theoretical network but also the topology as can be seen by comparing the two networks of Fig. 28. Here a prototype design is shown for a 1.6kHz crossover between a 25mm dome tweeter and a 100mm woofer/midrange. Even the computer-optimized network of Fig. 28(b) has the desired acoustic amplitude and phase characteristic only for about two octaves either side of the crossover frequency.

The active network in contrast to this can be exact because the voltage source at the driver terminals is able to impose any desired acoustic frequency response to the driver, without interaction between the source's frequency response and the driver impedance.

Note. In addition to the points noted on page 91 of the October issue, Mr Linkwitz points out that the horizontal scale for Fig. 6 is d/λ.

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Association of audio consultants

IN AN ATTEMPT to improve the standard of audio equipment reviews, an Association of Professional Audio and Radio Consultants has been formed. Acting secretary is James Moir, 16 Wayside, Chipperfield, Herts WD4 9JJ. The aims are as follows.

Aims of the association

The Association of Professional Audio and Radio Consultants was formed in July, 1978 to improve the standard of services offered by consultants, work towards protecting the interests of their clients and advance the reputation of the profession. It recognised that the work of unqualified consultants sometimes fell below desirable standards and the membership requirements of the association ensure that a high level of professional and technical competence is maintained.

The objects of the Association are:

(a) To maintain and where possible improve the standards of professional conduct and competence of consultants concerned with audio and radio engineering.
(b) To represent and make known the views of its members upon matters relating to, or affecting the profession.
(c) To promote further education and knowledge in audio, radio and acoustic engineering.

Some consultancies tend to specialise in particular aspects and members of the Association will be bound by their Code of Ethics to restrict their activities to areas where their professional expertise is relevant. The range of expertise available is fully comprehensive. For example, the prospective client will be able to choose consultants from within the membership of the Association to give assistance with:

1. The assessment of the performance of audio and radio equipment and the associated software.
2. All aspects of the acoustics of concert halls, studios and theatres, including speech reinforcement, sound recording and associated techniques.
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Membership: (a) *Member:* An organisation or an individual offering consulting services with no significant financial interest in the product of the service. *(b) Conditions of membership:* Applicants for membership shall furnish evidence of professional and technical competence.

The ethic which has been adopted by the Association is basically similar to that of the existing professional associations. It ensures that the advice proferred by its members is not influenced by financial interests in the products and that any commercial involvement is, in any event, limited to a small proportion of the member's activity. Members will always declare the nature of this involvement when appropriate.