PROJECT NOTES/ENGINEERING BRIEFS

A NOVEL APPROACH TO LINEAR PHASE LOUDSPEAKERS USING PASSIVE CROSSOVER NETWORKS

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Assuming equal electroacoustic transducer efficiencies in a multiway loudspeaker system, the vector sum of the voltages to the individual drivers must be equal to the input voltage for the accurate transfer of amplitude and phase characteristics of the signal. It has been shown that this is achievable using 6-dB per octave passive filters or higher slope active filters at a low signal stage to avoid unacceptable loss of power. Passive filters of 6-dB per octave slope require transducers with a power bandwidth not achievable with today's technology, while active filters requiring multiple amplifier systems are still too expensive to be provided commercially, especially in multichannel sound systems.

Passive filters are described using one or more auxiliary loudspeakers, "the missing link," in 12-dB per octave and higher degree crossover networks, which at the same time fulfill the requirement that the vector sum of the part voltages be equal to the input, theoretically without loss of power in the crossover network. Linearity of phase and amplitude are thus assured in the electrical circuit. Other requirements for linear phase loudspeakers are also described. Comparisons between computer calculated transfer functions and results from a practical high-performance low-distortion loudspeaker are shown.

INTRODUCTION: Over the last few years, dictated by increasing knowledge of the physiology of hearing and the recognition of sound signals by the human ear, development of high-fidelity loudspeakers and loudspeaker systems with good transient response has had increasing interest for researchers. Further, one of the goals of audio products must be to reproduce sound signals as correctly and accurately as possible, at least to the extent that all forms of audible distortion are removed.

Hansen and Madsen [1], [2] have shown that one form of distortion that is audible is phase distortion, and that this affects transient performance. Thus it is not sufficient, as was previously believed, to be content with a flat amplitude–frequency characteristic and low harmonic and intermodulation distortion, which is achieved by the better designs with steep cut crossover networks, but also low phase distortion. All these qualities can be included in a common test specification, which will be called "voltage transfer error function."

This paper describes multiway loudspeaker systems, the drivers of which will, for theoretical analysis, be assumed to be perfect within the frequency range in which they are used. Thus the description will mainly cover crossover networks and, to limit its length, two-way systems consisting of bass and treble drivers only. The theory can obviously be extended to multiway systems, with each crossover point being regarded as a two-way combination of high- and low-pass filter.

STATEMENT OF ASSUMPTIONS AND DEFINITIONS

To permit the comparison of the amplitude, phase, and impulse characteristics of loudspeakers with various crossover networks, it is useful to record the following assumptions made in the development of the argument.

1) Each driver is assumed to be perfect, that is, it gives a sound pressure level proportional to the input voltage at the voice coil terminals throughout the frequency range in which it is used.

2) All drivers used in a particular loudspeaker system give the same sound pressure level for the same input voltage, independent of its impedance.

3) All drivers have a pure resistive load characteristic.

4) All drivers are mounted so that there is no difference in the path length of sound signals from any driver to the listener.

Only when these requirements are met, can one assume that the voltage received at the voice coil terminals of a driver is also an expression for the sound pressure output from that driver.

We now define a new function, the voltage transfer error function VTE(s):

$$\mathrm{VTE}(s) = \frac{V_{\mathrm{in}}(s) - V_0(s)}{V_{\mathrm{in}}(s)} = 1 - \frac{V_0(s)}{V_{\mathrm{in}}(s)}$$

$$= 1 - \sum_{i=1}^{n} F_i(s) \tag{1}$$

where $V_{\mathrm{in}}(s)$ is the voltage delivered to the terminals of the loudspeaker system, $F_i(s)$ the transfer function of each driver's crossover filter, and $V_0(s)$ the vector-sum of voltages to the individual drivers and thus an expression for the total sound pressure generated. In the case where there is constant total voltage transfer, this function becomes a constant.

For a two-way loudspeaker system consisting of bass and treble drivers which are driven from the crossover with transfer functions $F_l(s)$ and $F_h(s)$, respectively, Eq. (1) can be simplified to

$$VTE(s) = 1 - F_l(s) F_h(s) \tag{2}$$

where l and h represent the low- and high-pass sections, respectively. If the crossover network consists solely of loss-free passive LC filters, one can put down the condition for constant total voltage transfer, which is

$$\mathrm{VTE}(s) = 0. \tag{3}$$

In a practical case the preceding assumptions will not be valid, and this will be discussed later.

Transfer functions are denoted as functions in the complex frequency domain operating normalized, for example, $s_n = s/\omega_0$ where $\omega_0 = 2\pi f_0, f_0$ being the nominal crossover frequency. Butterworth filters are used in all examples, as they give maximally flat amplitude characteristic for each driver.

SECOND ORDER CROSSOVER (12 dB PER OCTAVE) IN PHASE

A typical circuit diagram for a second-order crossover network is shown in Fig. 1*a*.

$$F_l(s) = \frac{V_{\mathrm{ol}}(s)}{V_{\mathrm{in}}(s)} = \frac{1}{s_n^2 + \sqrt{2}\, s_n + 1} \tag{4}$$

$$F_h(s) = \frac{V_{\mathrm{oh}}(s)}{V_{\mathrm{in}}(s)} = \frac{s_n^2}{s_n^2 + \sqrt{2}\, s_n + 1} \tag{5}$$

$$R_l = R_h = R \tag{6}$$

$$C = \frac{1}{\sqrt{2}\, R\omega_0} \tag{7}$$

$$L = \frac{\sqrt{2}\, R}{\omega_0} \tag{8}$$

$$VTE(s) = 1 - F_l(s) - F_h(s)$$

$$= 1 - \frac{1}{s_n^2 + \sqrt{2}\, s_n + 1} - \frac{s_n^2}{s_n^2 + \sqrt{2}\, s_n + 1}$$

$$= \frac{\sqrt{2}\, s_n}{s_n^2 + \sqrt{2}\, s_n + 1}. \tag{9}$$

The analysis shows that a loudspeaker with two drivers operating in phase, fed from a second crossover network, does not fulfill the condition for constant voltage transfer.

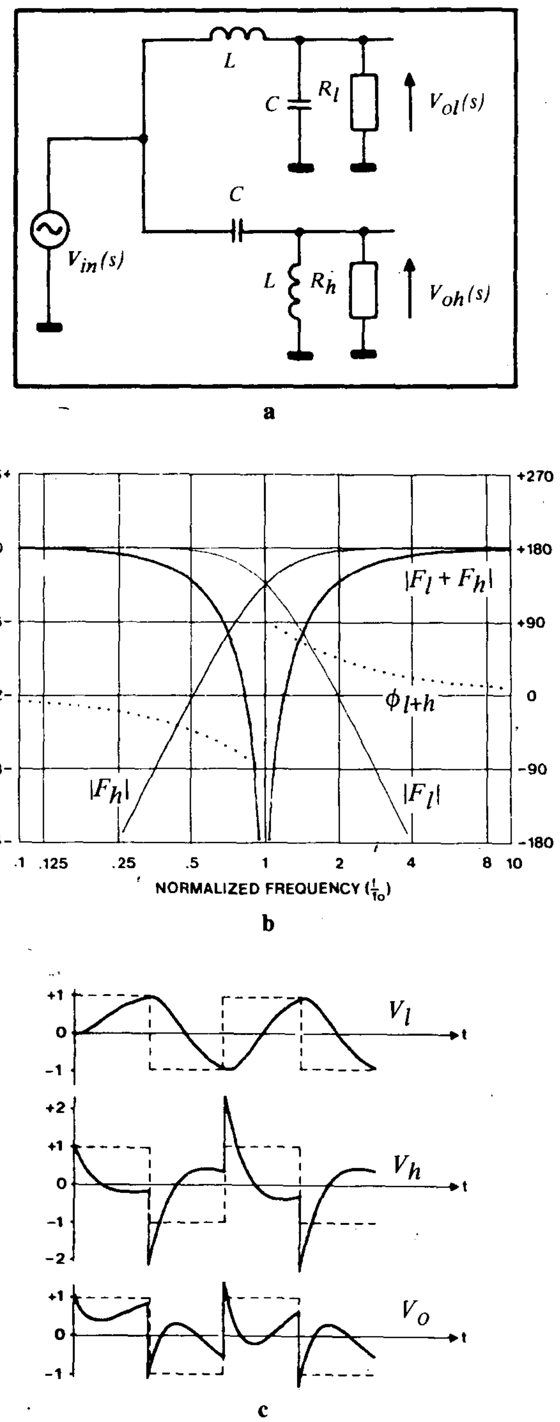

Fig. 1. Typical circuit diagram for second-order crossover network.

This can easily be seen in its amplitude, phase, and square-wave characteristics (Fig. 1*b* and *c*). The amplitude response has a null at the crossover frequency f_0, which can also be seen from the total transfer function in the frequency plane:

$$\frac{V_0(j\omega)}{V_{in}(j\omega)} = F_l(j\omega) + F_h(j\omega) = \frac{1 - \omega_n^2}{1 - \omega_n^2 + j\sqrt{2}\,\omega_n} \qquad (10)$$

where

$$\omega_n = \frac{\omega}{\omega_0}. \qquad (11)$$

SECOND-ORDER CROSSOVER (12 dB PER OCTAVE) REVERSED PHASE

The circuit diagram for this case is the same as that of Fig. 1*a*, but with one of the drivers connected with reserved polarity:

$$VTE(s) = 1 - F_l(s) - (-F_h(s)) = \frac{2\,s_n^2 + \sqrt{2}\,s_n}{s_n^2 + \sqrt{2}\,s_{n+1}}. \qquad (12)$$

Thus a loudspeaker with two drivers operating out of phase also does not fulfill the condition for constant voltage transfer, as can be seen from the amplitude, phase, and square-wave characteristics (Fig. 2*a* and *b*). In this case there is a 3-dB peak at the crossover point, besides the phase shift in the region of the crossover frequency.

ADDING THE MISSING LINK

We can now look at an alternative method for achieving constant voltage transfer. The theory is based on the principle[1] that instead of attempting to correct the electrical transfer function in the crossover networks for one or both existing drivers, the error from the ideal transfer characteristic can be corrected by adding a compensating electroacoustic signal with a transfer characteristic such that the total transfer characteristic is a constant.

In practice a straightforward method of achieving this goal is to add one, or more, filler drivers to the normal bass and treble drivers. These, when fed from an altered crossover network to reproduce the transfer characteristic, $F_c(s)$ = VTE (s), will together with the transfer functions for the bass and treble drivers, $F_l(s)$ and $F_h(s)$, give the required constant total transfer function.

SECOND-ORDER CROSSOVER WITH FILLER DRIVER IN PHASE

For this crossover network with Butterworth response, as shown earlier,

$$F_c(s) = VTE(s) = \frac{\sqrt{2}\,s_n}{s_n^2 + \sqrt{2}\,s_{n+1}}. \qquad (13)$$

In this case the required transfer characteristic is easily achieved by a single filler driver R_c connected in series through a series LC circuit. The complete crossover network is shown in Fig. 3*a*, with amplitude and phase characteristics shown in Fig. 3**b**. Fig. 3**c** shows that the amplitude, phase, and transient characeristics are correct, as expected.

$$R_l = R_h = R \qquad (14)$$

$$L = \frac{\sqrt{2}\,R}{\omega_0} \qquad (15)$$

$$L_c = \frac{R_c}{\sqrt{2}\,\omega_0} \qquad (16)$$

$$C = \frac{1}{\sqrt{2}\,R\,\omega_0} \qquad (17)$$

$$C_c = \frac{\sqrt{2}}{R_c\,\omega_0}. \qquad (18)$$

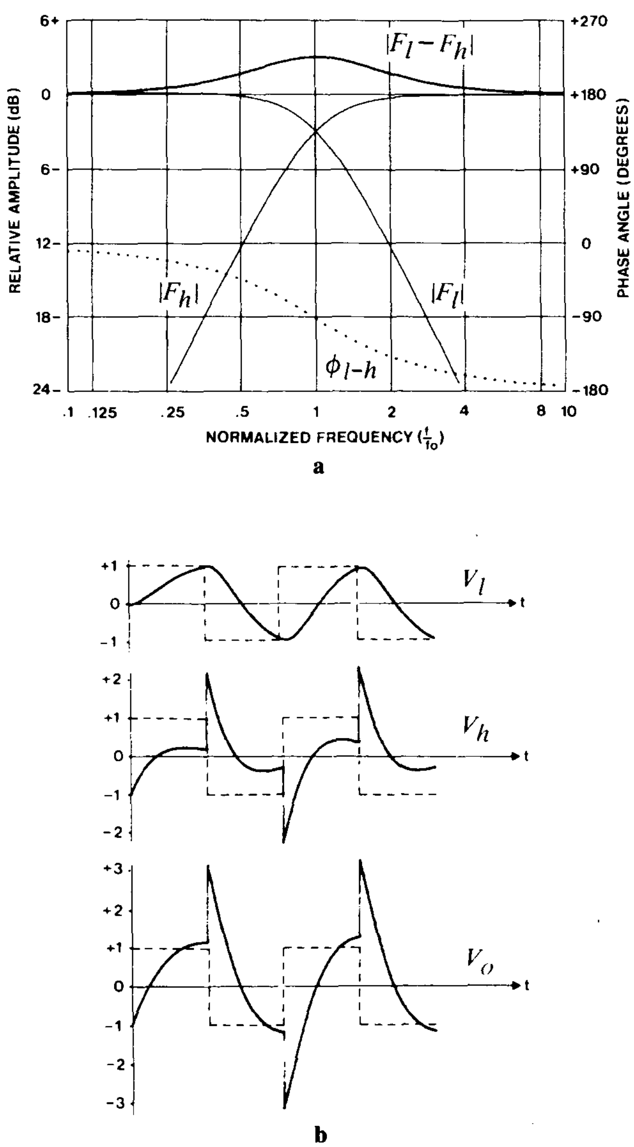

Fig. 2. Same as Fig. 1, but with one driver connected with reverse polarity.

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[1] Patented

$F_c(s)$ is a first-order bandpass function for a second-order crossover network, with a slope of 6 dB per octave and a −3-dB bandwidth of two octaves, that is, the signal is attenuated to both sides of a single peak frequency, to a −12-dB bandwidth of five octaves. Because of this, the design of a filler driver, with the required bandwidth, is within easily achievable limits.

The preceding loudspeaker system would be near perfect but for the fact that the total input impedance is no longer constant. It will vary from

$$R_{in} = R \quad \text{for } f >> f_0 \text{ and } f << f_0 \qquad \text{to}$$
$$R_{in} = R \| R_c \quad \text{for } f = f_0. \tag{19}$$

Conditions for constant input impedance are shown in Appendix A.

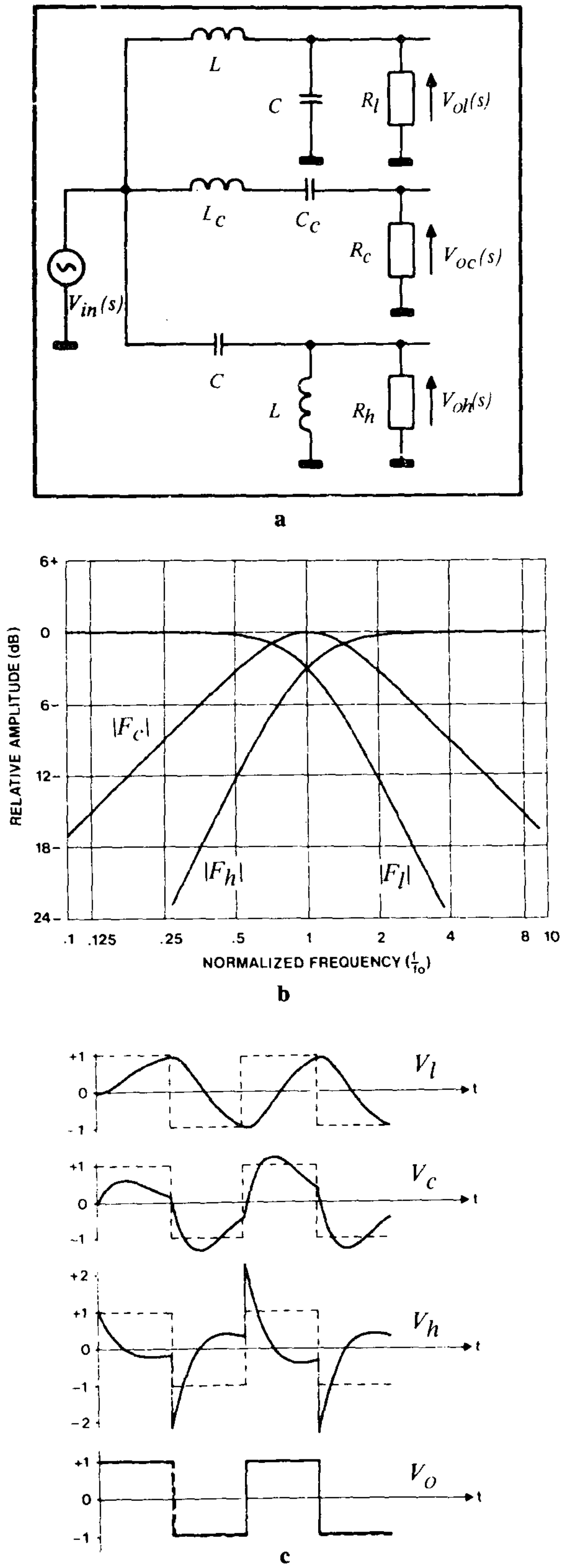


Fig. 3. Second-order crossover with filler drivers.

A PRACTICAL LOUDSPEAKER SYSTEM

A loudspeaker was constructed to verify the theoretical analysis presented in the preceding and to see the extent to which a practical loudspeaker could be made to reproduce square waves in an anechoic room. The setup consisted of a cabinet fitted with the necessary drivers, and external second-order crossover networks operating at a crossover frequency of 500 Hz. The crossovers could be switched to reproduce either with bass and treble drivers only, in phase or opposed phase, or with bass, treble, and filler drivers.

The response to a square-wave input, and the theoretical responses in all three cases, are shown in Figs. 4, 5, and 6. The close resemblance of the theoretical and actual curves leaves no doubt that practical loudspeaker systems can be made which reduce phase distortion to levels below audibility, using the methods explained.

The input level in all cases was 8 volts into 4 ohms, that is, 16 watts, at the crossover frequency.

PRACTICAL CONSIDERATIONS

It may be useful to note some of the deviations between theory and practice, and to point out some of the practical problems that must be solved in the design of a loudspeaker system. As far as drivers are concerned, these are seldom as perfect as assumed for the theoretical analysis, even within their nominal frequency limits. They tend to show limitations in their power-handling ability, amplitude response, and dispersion characteristic, but these, to some extent, may be compensated or corrected in the crossover network. These limitations have been described in various papers [3], [4], and we will not discuss them here any further.

Another problem is to ensure the same acoustic path length [5] from the acoustic axis of each driver to the listener. Differences in this respect will destroy otherwise correct transient response, even though it must be stressed that phase errors of this kind are small compared to those generated in incorrect crossovers. Drivers should be mounted as close to each other as possible, but differences in depth of the individual drivers make this insufficient by itself. One method of compensating for differences in depth is to position the loudspeakers on an angled mounting panel, as shown in Fig. 7. One can take advantage of the fact that a bass driver has a comparatively large dispersion angle, so that angling its axis has no effect on the sound received by the listener. The loudspeakers are thus effectively staggered a distance A, and the path length from the acoustic axis of each driver to the listener is the same. It should be noted, that this method is a satisfactory solution for the horizontal plane, but is ineffective in the vertical.

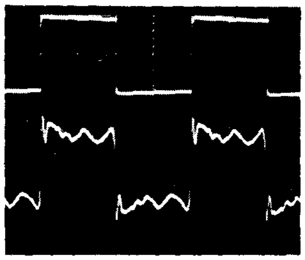


Fig. 4. With filler driver.

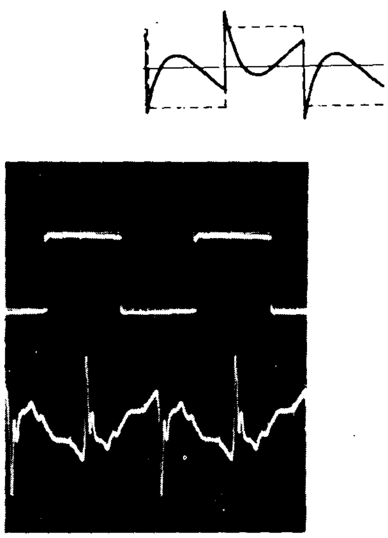


Fig. 5. In phase.

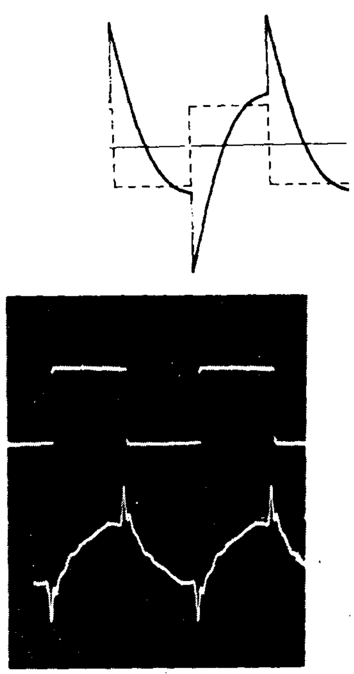


Fig. 6. Out of phase.

Fig. 8 shows the on-axis response and the effect of moving 15 and 30 degrees, respectively, on a horizontal plane in front of a prototype loudspeaker system built on the proposed principles. The input in all the examples is a square wave at the crossover frequency. It can be seen that there is almost no effect on transient performance.

Fig. 9*a* shows the calculated and actual responses, which are altered due to the time delay introduced, as would be the case if all the drivers were mounted on a flat panel. The delay introduced is 200μs (6.8 cm) for the bass driver and 100 μs (3.4 cm) for the filler driver.

For the purpose of this test, the microphone is positioned 15 degrees above the intended listening height to give the same results. There is now an effect on the performance, but this is still better than a conventional second-order crossover. It will be noticed that the loudspeaker system accurately follows the theoretically calculated response.

The effect of moving below the intended listening height is shown in Fig. 9*b*. Again the similarity between the predicted and actual responses is seen.

CONCLUSION

This paper has shown a way to design a multiway loudspeaker system free from amplitude and phase distortion, using inexpensive passive LC filters with crossover slopes of more than 6 dB per octave. Examples shown use the best known types of crossover, but the principles can obviously be used in conjunction with any other type of crossover. It has also been shown that loudspeaker systems do, in practice, closely follow theoretically predictable responses.

APPENDIX A

Input Impedance for Second-Order Crossover with Filler Driver

The conditions necessary for constant input impedance, for this crossover, can be calculated as follows. The general transfer functions for VTE(s) = 0 are

$$F_1(s) = \frac{1}{s_n^2 + as_n + 1} \tag{20}$$

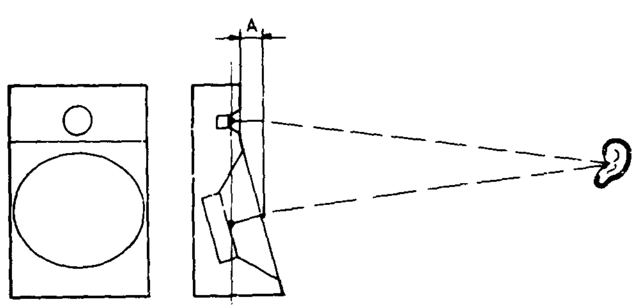


Fig. 7.. Loudspeakers on angled mounting panel.

$$F_c(s) = \frac{as_n}{s_n^2 + as_n + 1} \tag{21}$$

$$F_h(s) = \frac{s_n^2}{s_n^2 + as_n + 1}. \tag{22}$$

The input impedances will be

$$Z_l(s) = R \frac{s_n^2 + as_n + 1}{1 + (1/a)\, s_n} \tag{23}$$

$$Z_c(s) = \frac{s_n^2 + as_n + 1}{(a/R_c)\, s_n} \tag{24}$$

$$Z_h(s) = R \frac{s_n^2 + as_n + 1}{s_n^2 + (1/a)\, s_n} \tag{25}$$

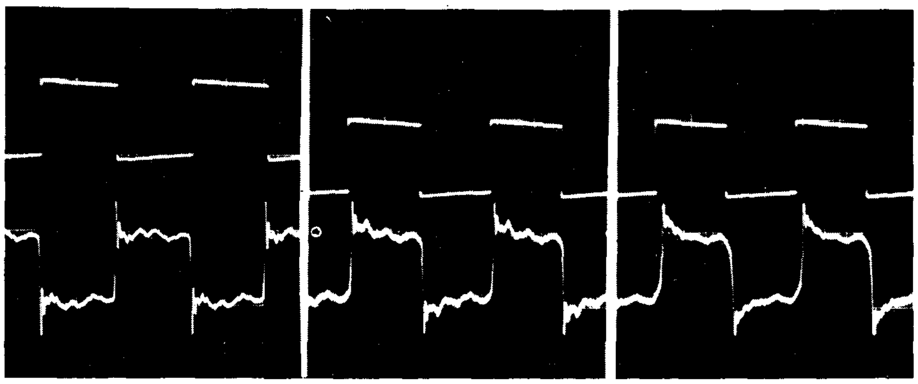
$$\Sigma\, Y_{in}(s) = \frac{s_n^2 + (2/a + a \cdot R/R_c)\, s_n + 1}{R(s_n^2 + as_n + 1)} \tag{26}$$

$$Z_{in}(s) = R\, \frac{s_n^2 + as_n + 1}{s_n^2 + (2/a + a \cdot R/R_c)\, s_{n+1}} \tag{27}$$

If the input impedance may be allowed to vary between limits, say R and $X \cdot R$, one gets the following conditions:

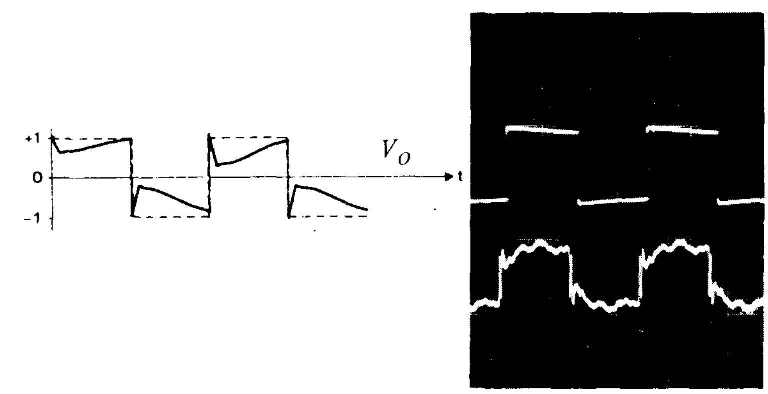
$$X = \frac{a}{2/a + a \cdot R/R_c} \quad \text{or} \quad a = \sqrt{\frac{2X}{1 - (R/R_c)\, X}}. \tag{28}$$

For constant input impedance, $X = 1$. It should be noted that when $R_c 7R$, the efficiency of the filler driver must be higher than that of the basic drivers. Component values for

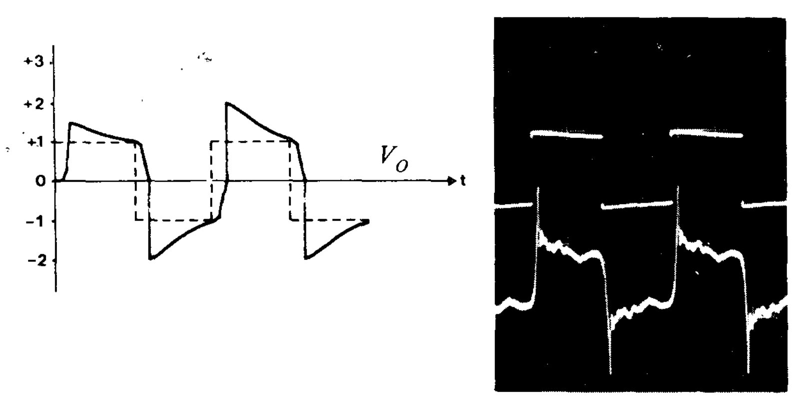


a b c

Fig. 8. Square-wave response. *a*. On axis. *b*. 15 degrees off axis. *c*. 30 degrees off axis.



a



b

Fig. 9. Calculated and actual responses. *a*. Effect of time delay. *b*. Effect of moving below intended listening height.

this filter will then be

$$L = \frac{aR}{\omega_0} \quad (29)$$

$$L_c = \frac{R_c}{a\omega_0} \quad (30)$$

$$C = \frac{1}{aR\,\omega_0} \quad (31)$$

$$C_c = \frac{a}{R_c\,\omega_0}\,. \quad (32)$$

Fig. 10 shows the resultant amplitude characteristics for various values of a. Choice of any particular value in a particular application will depend on the bandwidth characteristics of the drivers available and the input impedance requirements for the loudspeaker system.

APPENDIX B

Third-Order Crossover with Two Filler Drivers

For a third-order crossover network, as shown earlier,

$$F_c(s) = VTE(s) = \frac{2\,s_n^2 + 2\,s_n}{s_n^3 + 2\,s_n^2 + 2\,s_n + 1}\,. \quad (33)$$

This function is easily achieved directly with the help of two filler drivers with the following transfer characteristics:

$$F_{c1}(s) = \frac{2\,s_n^2}{s_n^3 + 2\,s_n^2 + 2\,s_n + 1} \quad (34)$$

$$F_{c2}(s) = \frac{2\,s_n}{s_n^3 + 2\,s_n^2 + 2\,s_n + 1}\,. \quad (35)$$

Third-Order Crossover with Single Filler Driver

The third-order transfer function derived in the preceding belongs to a type that can be achieved easily with a single filler driver. Reducing the expression for $F_c(s)$ we find

$$F_c(s) = \frac{2\,s_n^2 + 2\,s_n}{s_n^3 + 2\,s_n^2 + 2\,s_n + 1} = \frac{2\,s_n\,(s_n + 1)}{(s_n + 1)\,(s_n^2 + s_n + 1)} = \frac{2\,s_n}{s_n^2 + s_n + 1} \quad (36)$$

$$R_l = R_c = R_h = R \quad (37)$$

$$\frac{\eta_c}{\eta_l} = \frac{\eta_c}{\eta_h} = 2 \quad (38)$$

$$L_{1l} = \frac{3R}{2\omega_0} \quad (39)$$

$$L_{2l} = \frac{R}{2\omega_0} \quad (40)$$

$$L_c = \frac{R}{\omega_0} \quad (41)$$

$$L_h = \frac{3R}{4\omega_0} \quad (42)$$

$$C_l = \frac{4}{3R\omega_0} \quad (43)$$

$$C_c = \frac{1}{R\omega_0} \quad (44)$$

$$C_{1h} = \frac{2}{3R\omega_0} \quad (45)$$

$$C_{2h} = \frac{2}{R\omega_0}\,. \quad (46)$$

This function can be achieved with a series LC network connected in series to the driver, provided the efficiency of the filler driver η_c is twice that of the basic drivers η. The complete crossover network is shown in Fig. 11a.

Higher Order Crossovers

In general it may be said that for a crossover network with filters of higher order where

$$F_l(s) = \frac{a_0}{N(s)} \quad (47)$$

$$F_h(s) = \frac{a_n s^n}{N(s)} \quad (48)$$

where

$$N(s) = a_0 + a_1 s + a_2 s^2 + \cdots + a_n s^n \quad (49)$$

the sum of signals to the basic loudspeakers will be

$$\frac{a_0 + a_n s^n}{N(s)}\,. \quad (50)$$

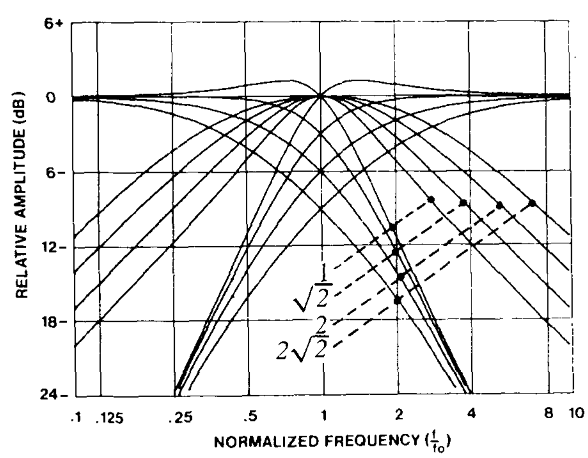


Fig. 10. Resultant amplitude characteristics for various values of a.

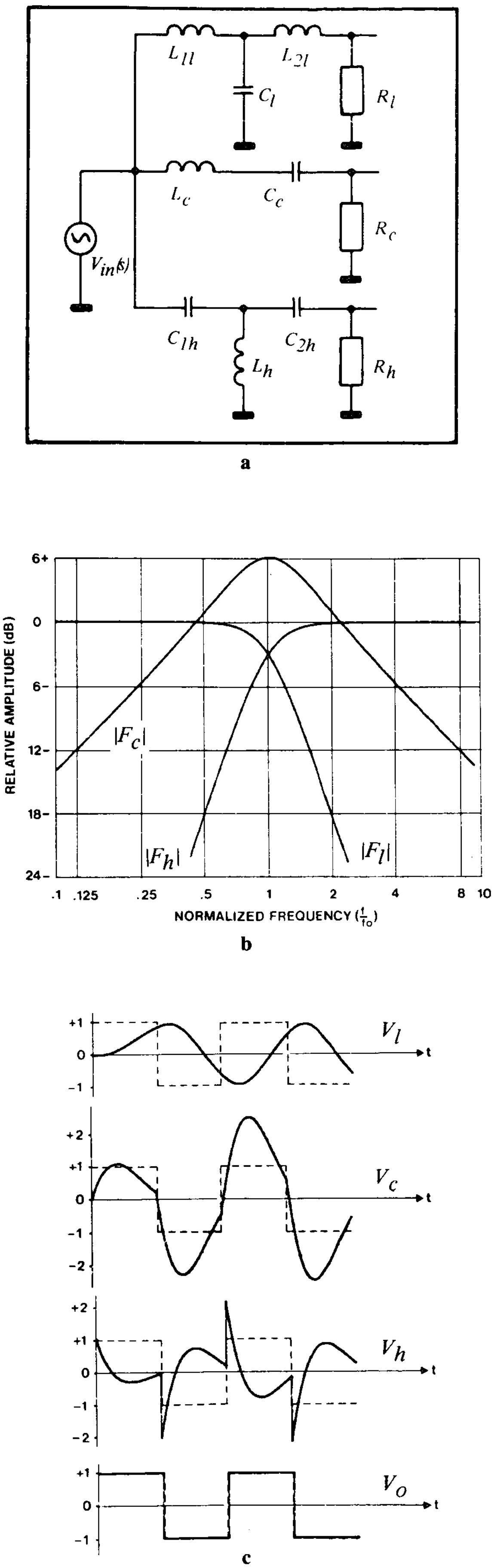


Fig. 11. Third-order crossover with single filler driver.

To achieve VTE(s) = 0, it will be necessary to add an electroacoustic signal with a transfer characteristic

$$F_c(s) = \frac{a_1 s + a_2 s^2 + \cdots + a_{n-1} s^{n-1}}{N(s)} . \quad (51)$$

This signal can be obtained with $(n - 1)$ filler drivers with their respective transfer characteristics,

$$\frac{a_1 s}{N(s)}, \frac{a_2 s^2}{N(s)}, \ldots, \quad \frac{a_{n-1} s^{n-1}}{N(s)} . \quad (52)$$

APPENDIX C

ANALYSIS OF CONVENTIONAL CROSSOVERS

First-Order Crossover (6 dB per octave), (Fig.12)

$$F_l(s) = \frac{V_{ol}(s)}{V_{in}(s)} = \frac{1}{s_n + 1} \quad (53)$$

$$F_h(s) = \frac{V_{oh}(s)}{V_{in}(s)} = \frac{s_n}{s_n + 1} \quad (54)$$

$$L = \frac{R_l}{\omega_0} \quad (55)$$

$$C = \frac{1}{\omega_0 R_h} \quad (56)$$

$$\mathrm{VTE}(s) = 1 - \sum_{i=1}^{n} F_i(s) = 1 - F_l(s) - F_h(s)$$

$$= 1 - \frac{1}{s_n + 1} - \frac{s_n}{s_{n+1}} = 0. \quad (57)$$

Quasi Second-Order Crossover (6 dB per octave), (Fig. 13)

$$F_l(s) = \frac{V_{ol}(s)}{V_{in}(s)} = \frac{a s_n + 1}{s_n^2 + 2 a s_n + 1} \quad (58)$$

$$F_h(s) = \frac{V_{oh}(s)}{V_{in}(s)} = \frac{s_n^2 + a s_n}{s_n^2 + 2 a s_n + 1} \quad (59)$$

$$R_l = R_h = R \quad (60)$$

$$C = \frac{a}{R\omega_0} \quad (61)$$

$$L = \frac{R}{a\omega_0} \quad (62)$$

$$\mathrm{VTE}(s) = 1 - F_l(s) - F_h(s)$$

$$= 1 - \frac{a s_n + 1}{s_n^2 + 2 a s_n + 1} - \frac{s_n^2 + a s_n}{s_n^2 + 2 a s_n + 1}$$

$$= 0. \quad (63)$$

The value of a selected will depend on the limitations imposed in a practical application. If one requires maximum flat response for each driver, it can be shown [6] that this occurs at $a = \sqrt{2/3}$. Amplitude and square-wave characteristics are shown in Figs. 13**b** and **c**. Curves for other values of a are shown in Ashley and Kaminsky [7]. The crossover network will have constant input impedance at $a = 1$.

Third-Order Crossover (18 dB per octave) (Fig. 14)

$$F_l(s) = \frac{V_{ol}(s)}{V_{in}(s)} = \frac{1}{s_n^3 + s_n^2 + 2\,s_n + 1} \quad (64)$$

$$F_h(s) = \frac{V_{oh}(s)}{V_{in}(s)} = \frac{s_n^3}{s_n^3 + 2s_n^2 + 2s_n + 1} \quad (65)$$

$$L_{1l} = \frac{3R_l}{2\omega_0} \quad (66)$$

$$L_{2l} = \frac{R_l}{2\omega_0} \quad (67)$$

$$L_h = \frac{3R_l}{4\omega_0} \quad (68)$$

$$C_{1h} = \frac{2}{3R_h\,\omega_0} \quad (69)$$

$$C_{2h} = \frac{2}{R_h\,\omega_0} \quad (70)$$

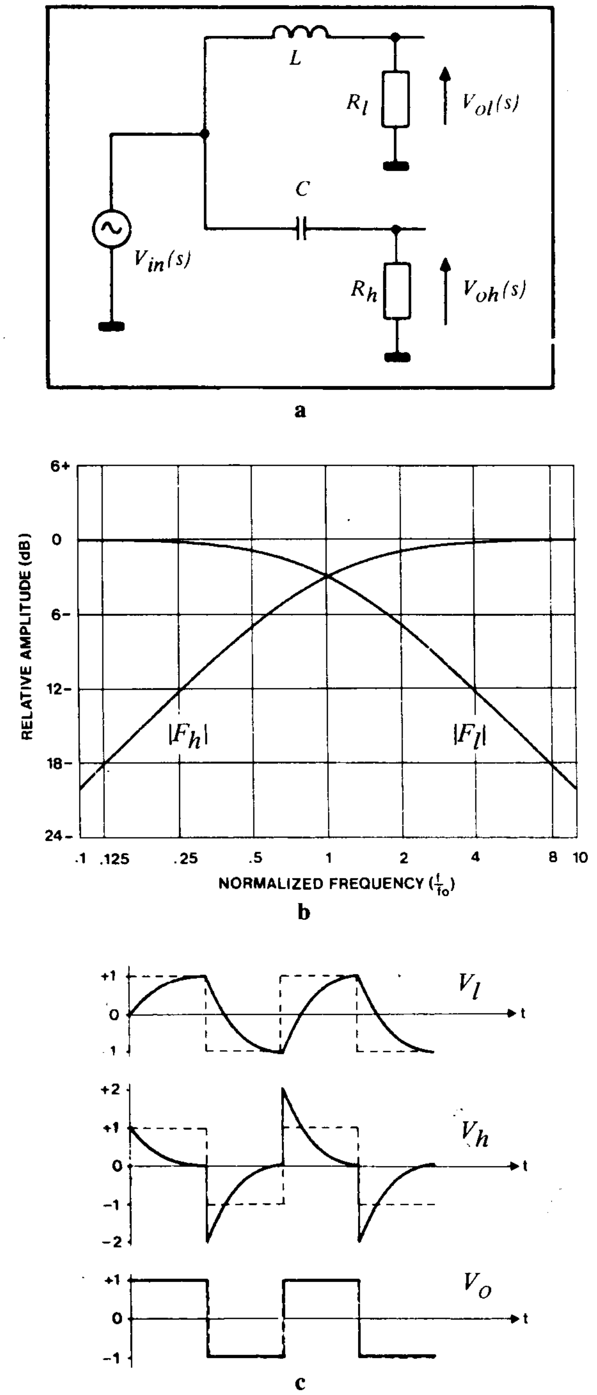


Fig. 12. First-order crossover.

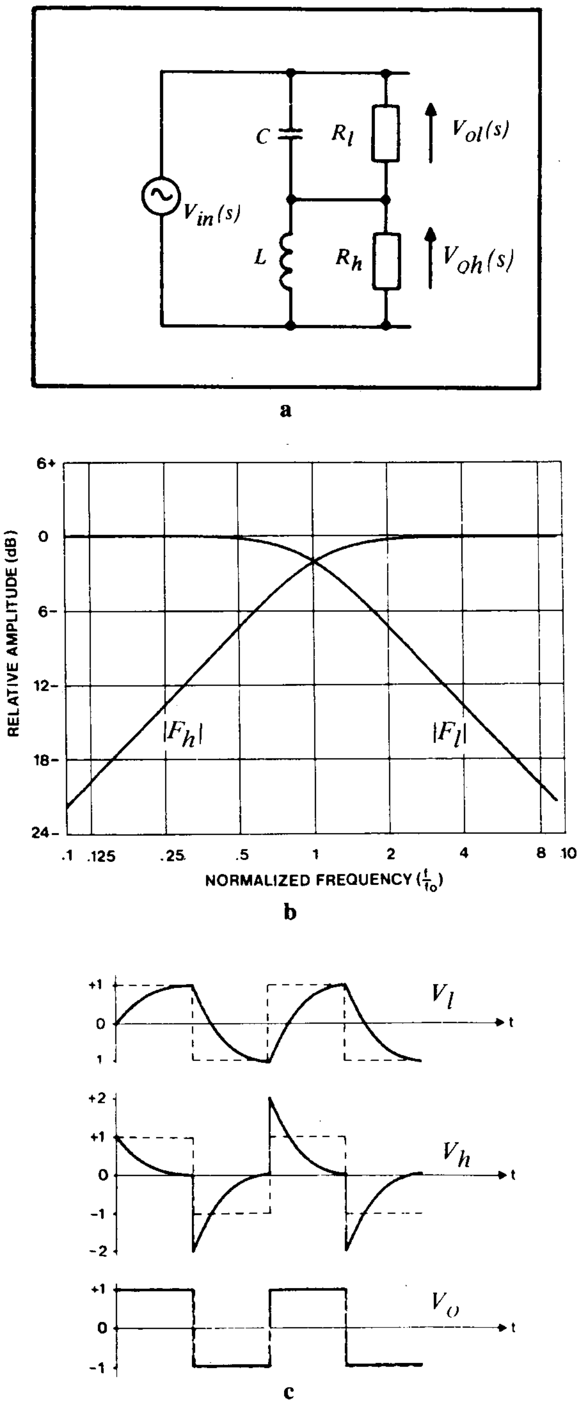


Fig. 13. Quasi second-order crossover.

$$C_l = \frac{4}{3R_l\,\omega_0} \tag{71}$$

$$\mathrm{VTE}(s) = 1 - F_l(s) - F_h(s)$$

$$= \frac{2s_n^2 + 2s_n}{s_n^3 + 2s_n^2 + 2s_n + 1}. \tag{72}$$

Other Types of Crossover Networks

We have now analyzed the best known crossover networks. Their widespread use is no doubt due to their straightforward design principles and suitability for realization with passive LC components.

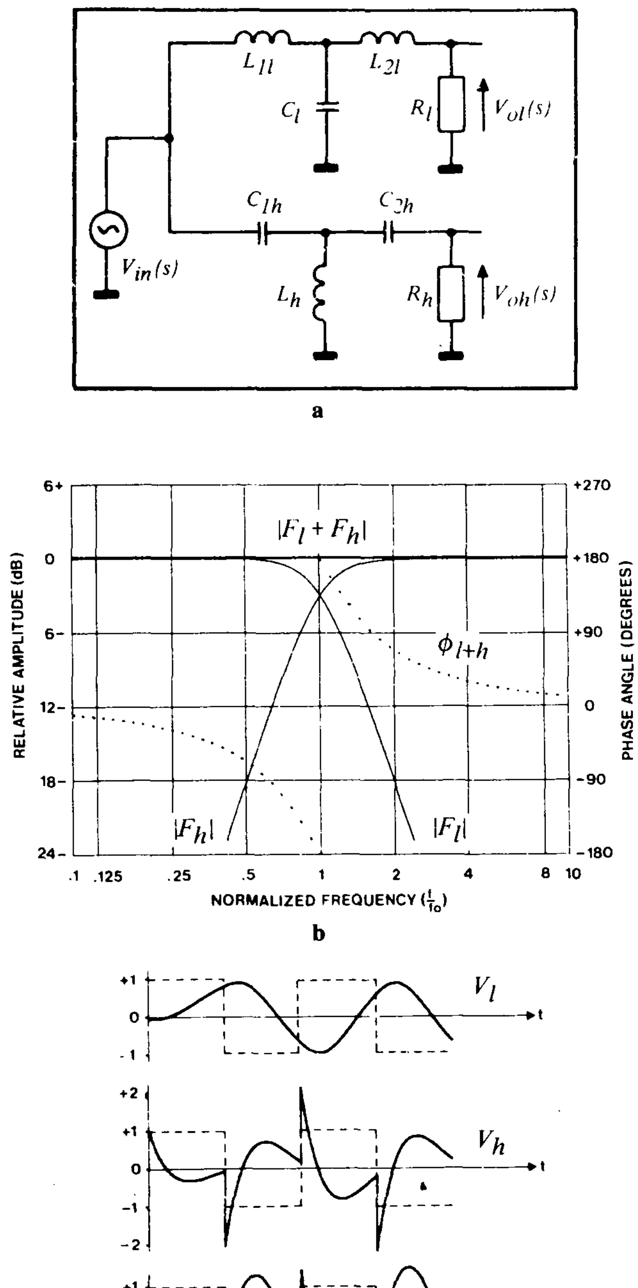


Fig. 14. Third-order crossover.

Some work has been done on the design of more complicated crossover networks [3], [7], [8] in order to achieve constant total voltage transfer, and at the same time have steep attenuation outside the driver's bandwidth. All of these have the drawback that they are based on active filters with individual amplifiers for each driver in order to avoid a large amplifier power loss. Further, attenuation in the crossover region is relatively gradual, and therefore they still require drivers with good performance over a large bandwidth. For example, in the case of one network giving slopes of 18 dB per octave, the overlap for 12-dB attenuation is over four octaves, which is not better than that for a first-order network, and considerably worse than that for second-order networks which have an overlap of two octaves.

This may account for the disappointing results with so-called phase linear loudspeakers observed by some researchers in listening tests, when compared with conventional loudspeakers.

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DESIGN PROBLEMS OF HIGH-LEVEL CONE LOUDSPEAKERS*

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To produce large acoustic power output with acceptable quality, a cone loudspeaker must combine high efficiency, high power capacity, and freedom from distortion and spurious noises. Analysis of loudspeaker operation shows that this set of performance goals imposes many difficult and often conflicting requirements upon the designer.

INTRODUCTION: The music business has demanded loudspeakers in recent years that are capable of withstanding higher and higher power input. The assumption seems to have been made that putting more power into the voice coil inevitably produces more power in the sound field. In any case, the "more is better" competitive pressure among system manufacturers has forced loudspeaker designers to solve some formidable reliability problems and to design loudspeakers that can absorb hundreds of watts from monster amplifiers and survive for hundreds or thousands of hours.

Unfortunately, high power capacity has usually been achieved through the use of heavy voice coils, cones, and suspension parts and through large air-gap clearances. These measures have resulted in reduced efficiency, and the benefit of increased power input has been only partially realized in terms of acoustic power.

The most promising strategy for getting more acoustic watts from high-power loudspeakers is improving efficiency. We will examine relationships among small-signal efficiency, bandwidth, and loudspeaker physical parameters; we will look at problems caused by the presence of high electrical power in the voice coil; we will discuss problems caused by large cone displacements, velocities, and accelerations; and we will consider how these problems could be avoided in the design of higher performance high-power loudspeakers.

SMALL-SIGNAL PERFORMANCE

Small [1] has derived the reference efficiency η_0 of a direct-radiator loudspeaker radiating into half-space as

$$\eta_0 = \frac{\rho_0 S_D^2}{2\pi c R_E} \cdot \left(\frac{Bl}{M_{MS}}^2\right) \quad (1)$$

where

ρ_0 = density of air (1.18 kg/m³)
c = velocity of sound (345 m/s)
R_E = resistance of voice coil (ohms)
B = average magnetic flux density through coil (Teslas)
l = length of voice coil conducter (meters)
M_{MS} = mechanical mass of diaphragm assembly plus air mass load (kilograms).

The reference efficiency is the mid-frequency acoustic output power P_A, divided by the nominal electrical input power P_E, where

$$P_E = \frac{e_g^2}{Z_N} \approx \frac{e_g^2}{R_E} . \quad (2)$$

Here e_g is the open-circuit source voltage and Z_N the nominal or rating impedance (ohms).

At the suspension resonance frequency f_S the acoustic output is usually different than at mid-band, and is determined by the details of the loudspeaker and the enclosure. In order to compare the effect on efficiency and bandwidth of various parameter changes, we will assume that the response shape can be made constant as f_S varies. For convenience, we will also assume that the response shape in the vicinity of f_S will remain constant if the electrical damping ratio of the loudspeaker (that is the ratio of R_E to reflected motional reactance) Q_{ES} is held constant. But,

$$Q_{ES} = \frac{2\pi f_S R_E M_{MS}}{B^2 l^2} \quad (3)$$

or

$$f_S = \frac{Q_{ES}}{2\pi R_E} \cdot \frac{B^2 l^2}{M_{MS}} = \frac{Q_{ES}}{2\pi R_E} \left(\frac{Bl}{M_{MS}}\right)^2 M_{MS}. \quad (4)$$

Note that if η_0 is forced to remain constant as M_{MS} is varied, then from Eq. (1) the ratio Bl/M_{MS} remains constant and f_S is proportional to M_{MS}. (This is the result of making the suspension compliance C_{MS} proportional to

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