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Speaker Builder

THE LOU JOURNAL

LISTEN WITH US...

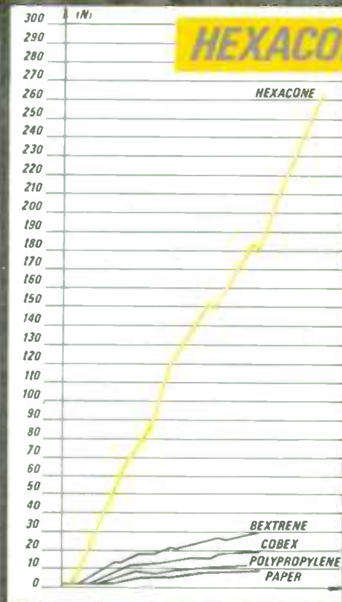
PERFECTION

Most of the high-tech products exported from West-Germany to North-America are more expensive but everyone knows why and accepts that price difference if you talk about cars for instance. The reasonable foundation of ETON DEUTSCHLAND-ELECTRO-ACOUSTIC GMBH is to produce less, but high-end quality drivers. ETON, established in 1983, likes to introduce itself as a manufacturer of sophisticated loudspeaker-drivers. As a designer of speakers ETON has always been working for both industry and kit-market. From the start ETON believed in different ways of developments in accordance to high-tech materials and dynamic measurement procedures. Looking forward to design „the driver" especially for the audiophilists ETON invested in computer aided constructions. With that support ETON succaded in producing the HEXACONE-diaphragm and something more for instance. The advantages of such cone-material are app. 70 – 100 times better stiffness and app. 30 % less weight in comparsion to paper or thermoplastic parts. The result shows no break-up resonances in the recommended frequency-range. HEXACONE drivers are the great step forward compared with highclass common plastic and paper cones – the new epoch of loudspeaker technology. Something more about HEXACONE: The diaphragm is a 3-layer-sandwich component. The inner honeycomb-structure, made of special coated phenolic NOMEX, is laminated by KEVLAR (fiberglass) and also coated but with duroplastic resin. For the typical shape ETON designed special tools suitable to guarantee less tolerances and highest temperature demands. The production is controlled by a CAM-System but cannot run without the attention of trained craftsmen. HEXACONE diaphragms require higher material and tools costs to give only one – and that is to say – the best to the listener. ETON, the small but innovative company, always looks for „the better products" made by us in West Germany.

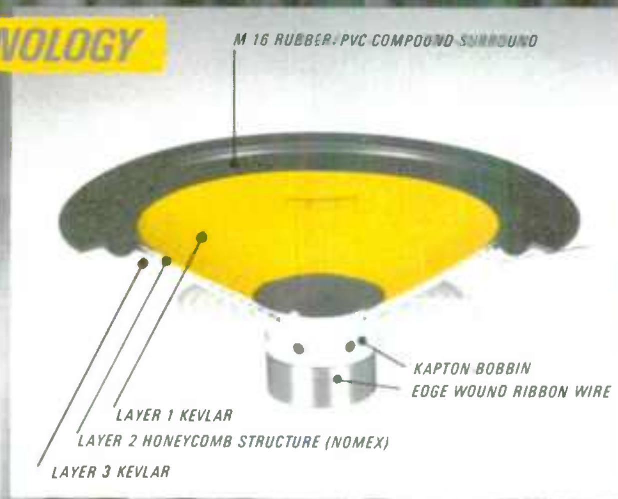
NOMEX and KEVLAR are registered trademarks of DUPONT
ETON and HEXACONE are registered trademarks.
HEXACONE is patented

ETON IS LOOKING

ETON is looking for a sole agent in USA and CANADA. Please contact the General-Manager Mr. Ulrich Ranke by letter or telex for closer informations.



HEXACONOLOGY



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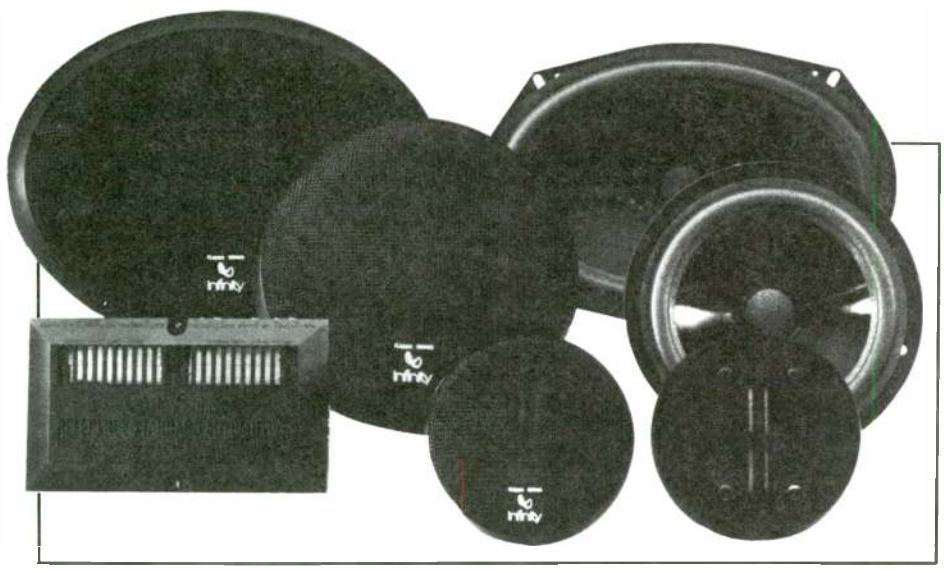
Good News



INFINITY has developed the Reference Standard *Kappa* Automotive Series for optimum performance in demanding car environments, featuring much of the technology first introduced in the IRS Series V system.

The series uses injection-molded graphite/polypropylene cones, neodymium magnets, a lighter diaphragm and new crossover networks. The woofer features a new cone (IMG) which is injection-molded in a high pressure process, forcing graphite fibers into the polypropylene in a radial pattern. The cone's stiffness-to-mass ratio, fast acceleration and high self-damping enable increased bass.

Replacing Infinity's current car speaker line, the new *kappa* models are: the 6 by 9 CS-1 component system; 6½″ CS-2 component system; 6 by 9, three-way RS693; 6 by 9 RS692; 6½″, three-way RS63; 6½″ RS62; 5″ RS52; 4″ RS42; 4 by 6 RS462; 3½″ RS32; and EMIT™ tweeter. Each model is completed by new ABS-reinforced grilles to prevent cracking or warping during temperature extremes. The CS-1 and CS-2 component systems incorporate a new crossover network, designed to improve clarity, spaciousness and linearity.

For more information contact Infinity Systems, Inc., 9409 Owensmouth Ave., Chatsworth, CA 91311, (818) 761-8838.

Fast Reply #HB354

A new Soundfield Imaging loudspeaker system has been introduced by **dbx.** The Model 50 is a vented four-way, six-driver system engineered to produce flat frequency and power response in the forward 180° and balanced stereo and bass/treble throughout the listening room.

The system uses a 10″ woofer, 6½″ midrange, 4½″ upper midrange, and three ultra-wide-dispersion ½″ tweeters. Crossover points (main axis) are 200/800/3150Hz. Frequency response is 34Hz–20kHz ±2.5dB (with a large vertical and horizontal angle in front of the pair); sensitivity of 91dB SPL/2.83V (1W at 8Ω)/1M; and an impedance of 4Ω nominal, 2.5Ω minimum.

Distinguished by new styling, the speaker grilles wrap around the cabinet with a charcoal/black or beige/walnut color choice. Dimensions are 44 by 16 by 13.

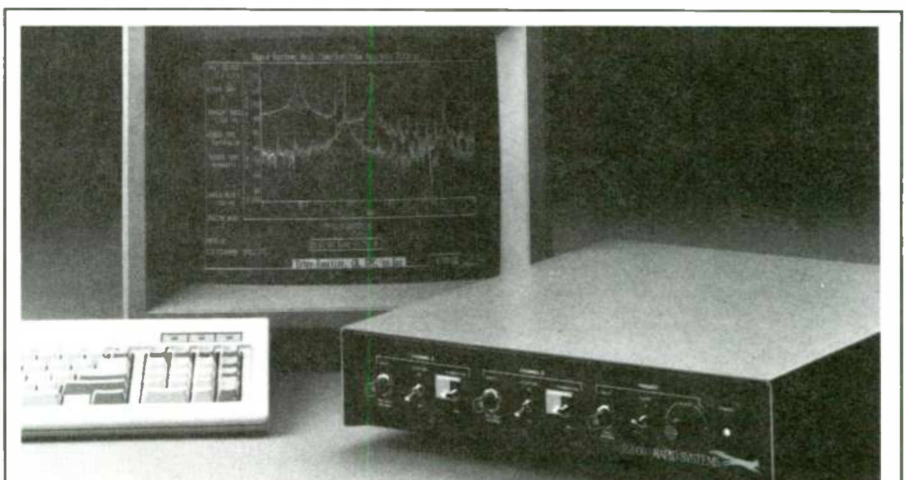
The Model 50 will be available throughout the US at less than $2,000/pair. Contact dbx, PO Box 100C, Newton, MA 02195.

Fast Reply #HB85

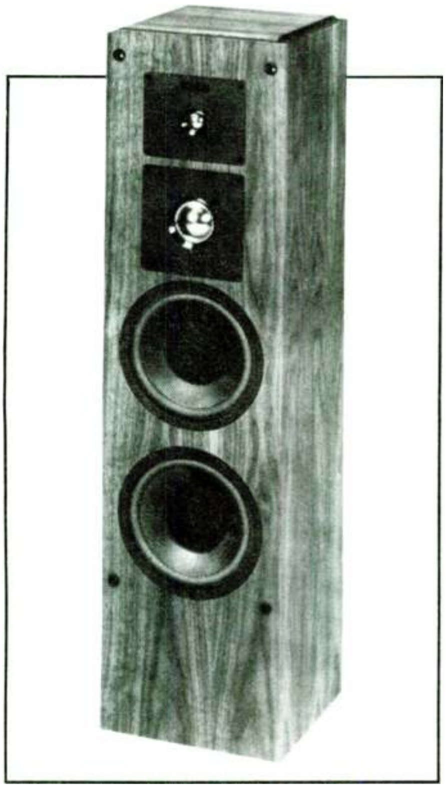
RAPID SYSTEMS offers the R370 two channel, real-time Spectrum Analyzer and Digital Oscilloscope, a unique combination using any IBM PC, XT, AT or compatible computer. This allows the user to view both the input signal and its frequency spectrum in applying the PC to spectrum analysis and transient capture in real time applications.

The oscilloscope features two channels of 20MHz simultaneous acquisition and selectable 50Ω or 1MΩ input impedance; the analyzer features 10MHz bandwidth and 1024 point FFT executed in 60mSec. Available for $4495: contact Mark Olsen, 433 N. 34th St., Seattle, WA 98013, (206) 547-8311.

Fast Reply #HB948



MADISOUND'S Audio Projects Bulletin Board (300, 1200, 2400 band, 24 hours) now has a new, and better telephone line. The new number is (608) 836-9473.



A new, slim tower loudspeaker system, the Model 508, was introduced at the 1987 CES by **ALTEC LANSING**.

Each 40″ high tower loudspeaker features two 8″ woofers with woven carbon fiber cones, a 2″ midrange and a 1″ tweeter, each made with polyimide with vacuum-deposited titanium domes; housed in a high-density, walnut-veneered pressed-wood cabinet with extensive bracing to reduce resonance.

The woofer cones are made of carbon fiber cloth reinforced with epoxy, to optimize low frequency response and maintain the rigidity needed to minimize distortion. The polyimide/titanium combination used for the midranges and dome tweeters is intended for speed and accuracy for good transient response by the drivers and wide stereo imaging with extremely low distortion. Frequency response is 40Hz–20kHz ±3dB with an SPL of 92dB (1W/1M); power handling is 125 nominal, 250W maximum; and an impedance of 4Ω.

Suggested list price is $1,000/pair. For additional information contact Altec Lansing Consumer Products, Milford, PA 18337.

Fast Reply #HB328

SPEAKER BUILDER

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About This Issue

Contributing Editor **Bob Bullock** leads off with the fifth in his series of explorations of the outer reaches of crossover theory. **Paul Graham** follows with some suggestions for easy filter calculations. **Peter Sutheim** concludes his fascinating evaluation of Richard Heyser's work, starting on page 21.

When **Jim Sawchuk** needed a rugged and portable speaker system, he decided to put it together himself. The chronicle of his project begins on page 28. **Jim Frane** is not your typical, shy consumer. Despite the magisterial, lab-coated character in the Polk ads, Jim decided he could make his Polk 10s image better—and succeeded (page 32).

Dave Davenport accepted our assignment to report on the performance of L.A. White's unusual speaker design with some misgivings. While impressed with some things about the unit, his report is less than enthusiastic about other characteristics (page 35).

Our cover is graced by a view of Contributing Editor **Joe D'Appolito**'s Swan, a new version of his popular satellite system, which appeared in these pages in 1984. The setting is Swan Island, Maine, where boat builder Jim Bock built several last winter. See page 39.

Our review section, beginning on page 40, includes opinions by Contributing Editor **Gary Galo** on our *Audio Anthology* reprint, **John Cockroft**'s hilarious review of a very funny British book, *Bluff Your Way In Hi-Fi*, and **Roger Sanders**' evaluative commentary on a new volume about electrostatics. **Bob White** completes the lineup with a review of Scientific Design Software's newest version of *Computer Aided Speaker Design*.

SpeakerBuilder
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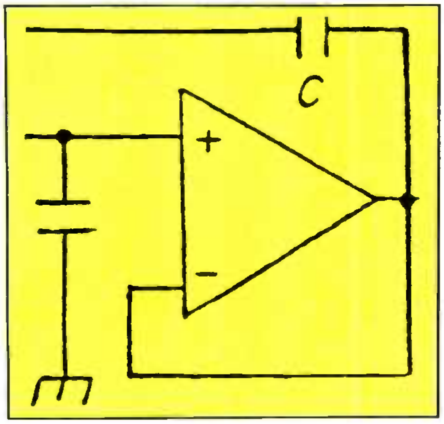


FEATURES

DEPARTMENTS

Editorial

ACCELERATION FACTOR

"Cello playing technique has progressed more during the last 25 years than in all the previous history of the instrument." I heard these words while driving along Vermont's I-89 listening to Vermont Public Radio. The cellist went on to say he believed the prime reason for the acceleration in cello playing technique is recordings.

"We all hear each other do more and more difficult things and we say to ourselves, well, I guess I'd better go practice some more."

What the cellist observes is only a tiny part of what is happening in almost every human endeavor. And in our speaker builder corner of the world, the changes are just as pervasive, accelerated and delightful. Maybe we should call this heightened development the acceleration factor.

Critics seemed to abound when I made the decision in 1980 to separate the loudspeaker interests of *Audio Amateur's* readers from the more general audio hardware topics which that publication served. Doom for both periodicals was predicted, accusations of cupidity were plentiful, and others could not see that the loudspeaker field had all that much content to deserve a separate publication.

We enter our ninth year with *Speaker Builder* growing steadily, having just topped 7,000 worldwide circulation. This issue carries 23 pages of paid advertising.

But far more important to me than size is the acceleration factor. It is always difficult to realize the extent and significance of a major change in our world while it is happening around us. I believe we are all learning at an accelerated pace these days. As one soprano said the other day, "When you look back over a span of twenty-five years of your life, don't you feel you are very different—indeed, that you have lived several lifetimes in that span?"

Looking back over the 32 issues of *Speaker Builder* the pace of learning, as well as the content, has been speeding up in every issue. Our writers and correspondents have been exploring, questioning, searching diligently for reasons for the anomalies and to shed light on unanswered questions. The same process is going on within the companies which manufacture everything from driver parts and drivers, to enclosures and systems.

It is also true that not only are manufacturers supplying more information about their products, but more and more products actually measure closely to published parameters—and the product is more consistent from batch to batch. Indeed, some makers offer matched pairs of drivers to insure system channel balance.

Serious, reliable work on loudspeaker performance took a giant step forward with the discovery of Thiele and Small's work. And explorations of their theses still goes on. A great deal remains unclear in reliable theory about other driver formats. Do we have comparable theoretical bases for planar, electrostatic, dipole, and transmission line types? Much work is still ahead. Loudspeaker voyages of discovery are still not much further along than the second enterprise of Christopher Columbus. A very large proportion of loudspeaker geography remains uncharted.

Readily available computer hardware has been an important factor in theoretical analysis, design, manufacturing and performance tests of both drivers, crossovers and systems. The loudspeaker in its working environment has more variables which must be dealt with in analyzing its performance than any other part of the sound system. Until we had readily available computing power, we could not—or would not—undertake the drudgery required to analyze everything that goes on when a speaker works.

A very large part of this work of theoretical and empirical development of loudspeakers is being done by amateurs. Richard Heyser worked professionally for most of his active career in a company specializing in jet propulsion. The loudspeaker industry hired him from time to time as a consultant. But the audio industry generally does not attract the best engineering brains—and the loudspeaker portion of the industry is no exception. That may have begun to change, but it has been a long time coming.

The collective skills and experience of those who write for *Speaker Builder* is not only impressive, it is as good or better than the staff of all but a few of the loudspeaker manufacturers functioning today. Only three of our authors are actively employed in the business. And they came to it by the route of being active amateurs, consultants and finally full-time employees. The central core of our authors are either academics or highly skilled, technically trained professionals working in a wide variety of high-tech companies. Their enthusiasm for loudspeaker technology is an avocation. Their knowledge is based on training in various branches of physics and the work all of them do with their hands. I believe that *Speaker Builder* has given them a meeting place to swap ideas. And each of them, I suspect, has had his knowledge and experience enriched.

As I talk with these interesting people, which is one of my richest rewards, I sense they all are on their own voyage of discovery. And that is what an avocation is for. Those of us who know far less, in most instances, have the fun of sharing their adventures. And the pleasure of acquiring new knowledge and skills.

I believe *Speaker Builder* has become one force in the acceleration factor affecting loudspeaker development. For that reason we will increase the speed in 1988 by going to six issues per year. We are also publishing the first issue of *Voice Coil*, our newsletter for the industry, as this is written in November. I look forward to the ninth year of *Speaker Builder* with the greatest anticipation and pleasure. I hope each of you are joining this next installment of a great enterprise.—E.T.D.

Part V

PASSIVE CROSSOVER NETWORKS

ALTERNATE BANDPASS CIRCUITS

BY ROBERT M. BULLOCK III
Contributing Editor

Shortly after my article on three-way passive crossovers appeared in *SB* 2/85, contributing editor G. R. Koonce informed me that one of the third-order circuits I described there can cause the input impedance of a system to fall below 50 percent of its nominal design value at midrange frequencies. Since many amplifiers do not like extremely low impedances, I thought the extent of the problem should be investigated and the conclusions brought to the attention of *SB* readers.

I found the first- and second-order networks do not manifest the problem, but the fourth-order and one of the third-order networks do. I also found, however, that all except the first-order network can cause a wide variation in system input impedance with frequency. The impedance variation cannot be eliminated, but it can be alleviated by the use of an alternative bandpass topology. In my paper[1] on all-pass networks, I described two different topologies for implementing bandpass filters, but when I wrote the *SB* article, I decided to use only one of them in the interest of simplicity. It now appears that you should be familiar with both topologies to make the best design trade-offs.

I will start by describing the alternative bandpass topology in detail so you can design a crossover using it. Then I'm going to show you the input impedance behavior of the various crossover topologies and orders so that you can factor this information into your considerations when selecting a crossover network.

NOTE: Part II of this series, SB 2/85, is referred to in this article. It contains incorrect formulas on page 30 that should be corrected as follows:

Formula (9) should read:

$$A = R + 1/R$$

Formula (10) should read:

$$A = R - 1/R$$

Formula (16) should read:

$$A = a\,(R + 1/R)$$

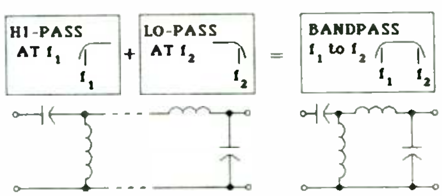


FIGURE 1: The block diagram shows a bandpass filter formed by casading. The circuit is an example of the procedure.

C-BANDPASS. The purpose of a bandpass filter is to pass all frequencies between two specified frequencies and to reject all others. One way to accomplish this is to cascade a low-pass filter with corner frequency at the higher specified frequency with a high-pass filter with corner frequency at the lower specified frequency, as illustrated in *Fig. 1.* This is the method I used to derive the bandpass filters in the three-way crossover circuits described in *SB* 2/85. I will refer to this bandpass circuit as the C-bandpass topology, to indicate it is derived by cascading.

T-BANDPASS. There is another way to obtain a bandpass filter if ladder circuits are used. Start with a low-pass ladder topology, such as the one shown in *Fig. 2a.* Replace each series inductor by a series capacitor and inductor as in *Fig. 2b,* and each parallel capacitor by a parallel capacitor and inductor as in *Fig. 2c.* The resulting circuit, shown for the example in *Fig. 2d,* is a bandpass filter. This procedure can be described theoretically in terms of frequency transfor-

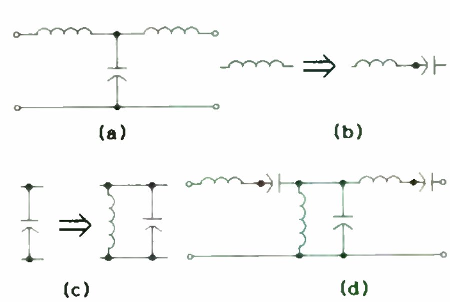


FIGURE 2: Alternative method of forming a bandpass filter. Start with a low-pass filter (a), apply the component transformations shown in (b) and (c). The result is the bandpass filter in (d).

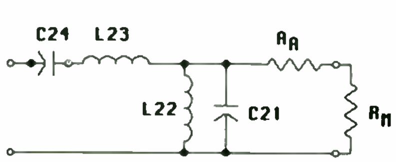
mations, so I am going to refer to the circuit as the T-bandpass topology.

Either of these topologies can be used for the bandpass section of a three-way crossover network, but each requires its own design formulas. These are given in *SB* 2/85 for the C-bandpass topology, and the relevant formulas for the T-bandpass topology are given here in *Figs. 3, 4* and *5*, covering second-, third-, and fourth-order crossover networks. The topologies are identical for first-order networks, so separate formulas are not needed.

To use the T-bandpass topology in any of the networks described in *SB* 2/85, first replace the C-bandpass circuit shown there in *Figs. 8, 9,* or *10* with the proper T-bandpass circuit. Then delete any formulas there for E, F, G, K, M, N, P, Q, and T; and replace them with the relevant formulas from *Figs. 3, 4,* or *5* here. Also delete there the component formulas R_A, R_0 and all those Ls and Cs with first subscript 2; replace them with the corresponding formulas given here.

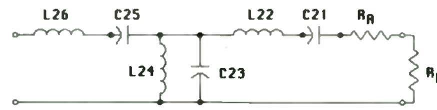
T-BANDPASS GAIN. Recall that resistor R_A in the bandpass circuit is there to cut down on the natural gain of the circuit so its output matches the low-pass and high-pass sections correctly. If you plan to dispense with R_A and adjust for the "excess gain" (EG) some other way, you must calculate the value of EG from Formula (35) on page 34 in *SB* 2/85. Now you have two choices for the bandpass topology, so be sure you use the value of K corresponding to the topology you are considering, since it is different for the two topologies.

SOFTWARE. The Crossovers disk offered by Old Colony Sound Lab (SBK-F1A) can be used to find the component values for three-way crossover networks using either the C-topology or the T-topology. The program to use is called Passive Three-Ways. The updated version also has an option for deleting R_A from the circuit. In this case, the program gives you the amount of excess bandpass gain this creates. If you already own the disk, you can update it yourself by adding the code in *Listing 1.*



$K = B - 2$
$R_A = R_H(K/H-1)$
$R_0 = R_A + R_H$
$C21 = 1/A/R_0/W2$
$L22 = AR_0/W2$
$L23 = AR_0/K/W2$
$C24 = K/A/R_0/W2$

FIGURE 3: Schematic and design formulas for a second-order T-bandpass circuit.



$E = B - C/A$
$F = B - 1 - C/A$
$K = C - 2A$
$R_A = R_H(K/H-1)$
$R_0 = R_A + R_H$
$C21 = A/R_0/W2$
$L22 = R_0/A/W2$
$C23 = A/F/R_0/W2$
$L24 = FR_0/A/W2$
$C25 = K/F/R_0/W2$
$L26 = FR_0/K/W2$

FIGURE 4: Schematic and design formulas for a third-order T-bandpass circuit.

IMPEDANCE. To illustrate the input behavior of these crossovers graphically, I assume that each driver behaves as an 8Ω resistor and the lower crossover frequency is 500Hz. Graphs of system input impedance are drawn for both all-pass and Butterworth crossovers using two different upper crossover frequencies. The case of close crossover frequencies is illustrated by using an upper crossover frequency of 1.5kHz for a spread of about 1.6 octaves.

For the widely separated case, the upper crossover frequency is taken as 4kHz, for a spread of three octaves. A single figure contains information for only one crossover type and one crossover frequency spread. It consists of two curves on the same axes, one being the crossover's input impedance for the C-bandpass topology and the other for the T-bandpass topology. This makes a total of fourteen possible figures after eliminating duplicates.

SECOND-ORDER NETWORKS. *Figures 6* and *7* show the input impedance for all-pass networks of spreads 3 and 8, respectively. Low input impedance is definitely not a problem, but impedance does vary widely with frequency, which is not especially attractive.

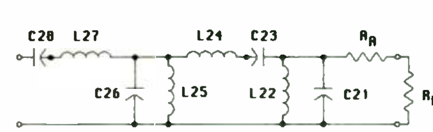
The Butterworth networks exhibit a different behavior. With closely spaced crossover frequencies (*Fig. 8*) the impedance drops to about 75 percent of nominal for the C-bandpass topology. Thus, the T-bandpass topology is the better choice. When the crossover frequencies are more widely spaced, *Fig. 9* indicates that either topology should be okay.

THIRD-ORDER NETWORKS. First consider the all-pass network in which driver polarity is observed in all three channels. *Figure 10* shows that, regardless of topolgy, closely spaced crossover frequencies produce very low impedances between the crossover frequencies. The T-bandpass topology alleviates this to some extent and probably should be used, but even then the impedance is still quite low. If the crossover frequency spread is larger, as in *Fig. 11*, the impedance decrease is not quite so bad, and using the T-bandpass topology could raise it to acceptable level. In general, I would avoid this network because of its poor impedance behavior. By the way, this is the network about which G. R. Koonce contacted me.

The other third-order network is the one in which the polarity of the midrange driver is reversed. As *Figs. 12* and *13* show, it should not present a difficult load, even when the crossover frequencies are close together. The overall impedance variation is, however, large. *Figures 14* and *15* show the impedance for the third-order Butterworth crossover. The midrange polarity does not affect this network, however, as the impedance is the same whether polarity is observed or reversed. Clearly the T-bandpass topology is best for closely spaced crossover frequencies, but either topology should be acceptable if frequencies are widely spaced.

FOURTH-ORDER NETWORKS. Again, from *Figs. 16* and *17*, the T-bandpass is the topology of choice for the all-pass crossovers, especially for closely spaced crossover frequencies. Even then, the overall impedance variation is large.

From *Figs. 18* and *19* we see that a C-bandpass topology is probably okay for Butterworth crossovers with a large



$E = B - 1 - C/A$
$F = D - 2C/A$
$G = C - A - AF/E$
$K = D - 2B + 2$
$R_A = R_H(K/H-1)$
$R_0 = R_A + R_H$
$C21 = 1/A/R_0/W2$
$L22 = AR_0/W2$
$C23 = E/A/R_0/W2$
$L24 = AR_0/E/W2$
$L25 = GR_0/E/W2$
$C26 = E/R_0/G/W2$
$L27 = GR_0/K/W2$
$C28 = K/R_0/G/W2$

FIGURE 5: Schematic and design formulas for a fourth-order T-bandpass circuit.

World Radio History

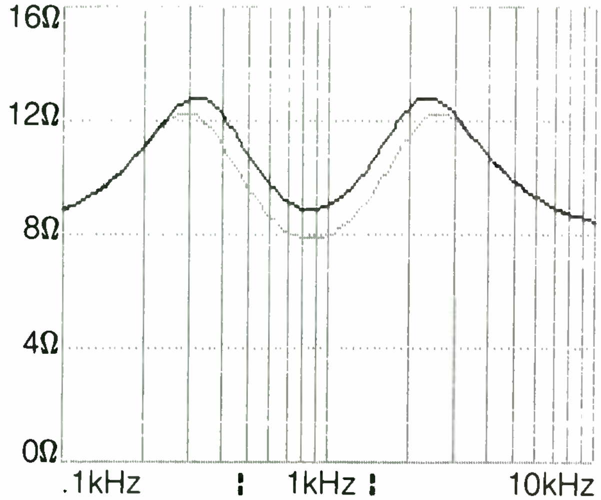


FIGURE 6: Input impedance of a three-way second-order all-pass crossover network with crossover frequencies of 500Hz and 1.5kz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

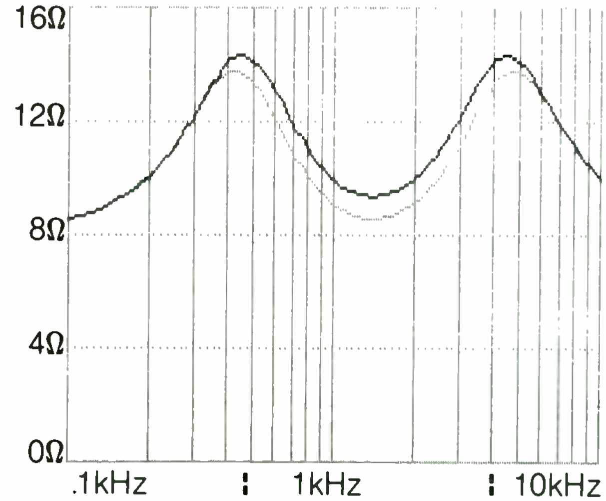


FIGURE 7: Input impedance of a three-way second-order all-pass crossover network with crossover frequencies of 500Hz and 4kHz. [Dark curve is with T-bandpass toplogy, light with C-bandpass topology. Marks | locate crossover frequencies.]

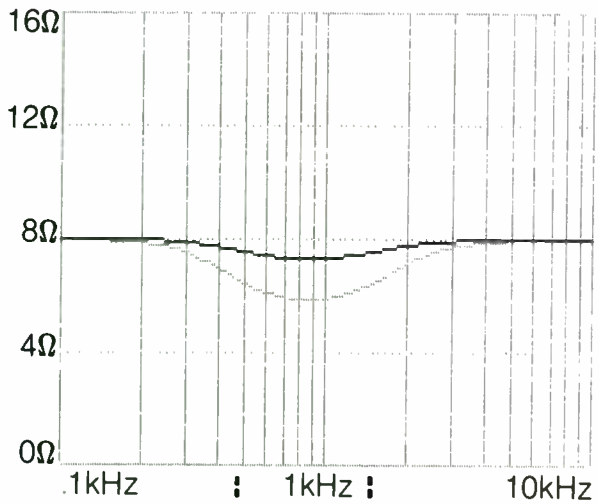


FIGURE 8: Input impedance of a three-way second-order Butterworth crossover network with crossover frequencies of 500Hz and 1.5kHz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

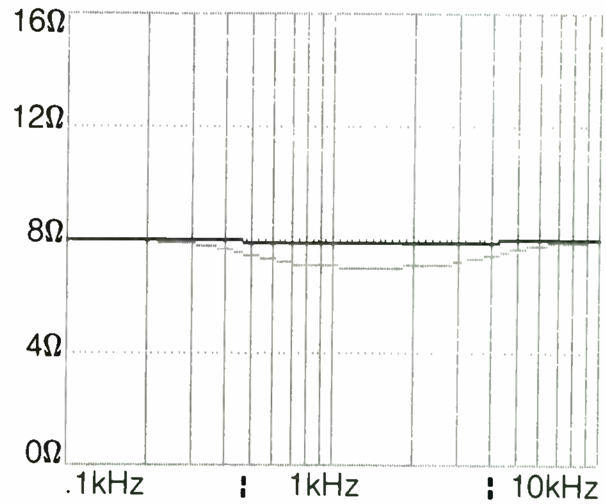


FIGURE 9: Input impedance of a three-way second-order Butterworth crossover network with crossover frequencies of 500Hz and 4kHz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

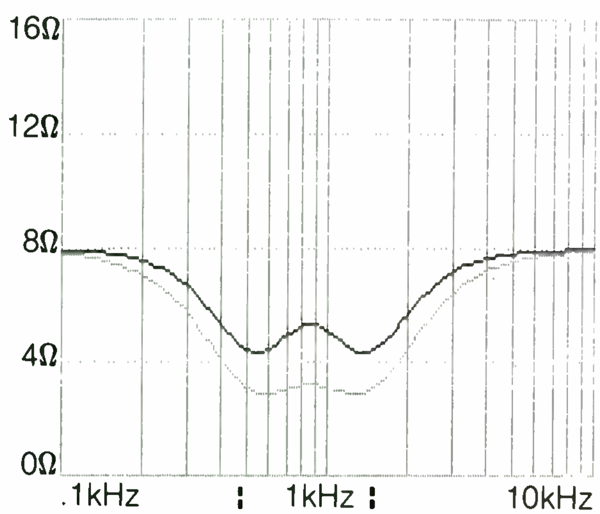


FIGURE 10: Input impedance of a three-way third-order all-pass crossover network with crossover frequencies of 500Hz and 1.5kHz and bandpass polarity observed. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

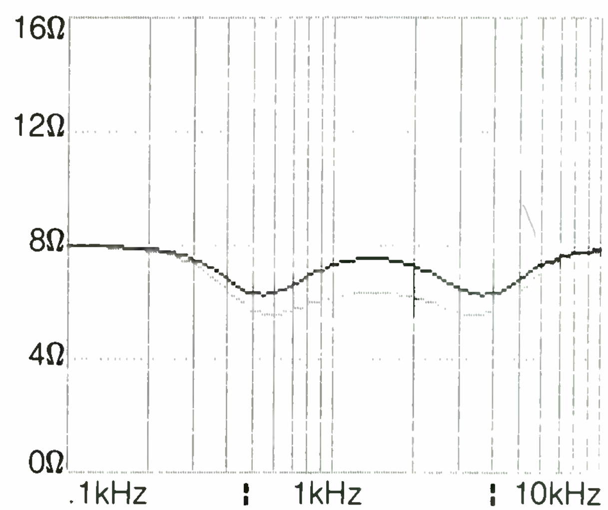


FIGURE 11: Input impedance of a three-way third-order all-pass crossover network with crossover frequencies of 500Hz and 4kHz and bandpass polarity observed. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

crossover frequency spread but not for those with a small spread.

CONCLUSIONS. In general, a T-bandpass topology crossover presents a better behaved input impedance than a C-bandpass topology. Sometimes the difference between the two is marginal, and other considerations dominate. For example, the C-topology can provide more midrange gain, which could be significant for sensitivity matching. You might take advantage of this if you were using a second-order crossover, but if you wanted to use a third-order Butterworth with closely spaced crossover frequencies, it is probably best to stick with the T-topology.

Regardless of the bandpass topology, the all-pass networks exhibit a wide input impedance fluctuation with frequency. It seems that for passive networks, you must decide whether it is worse to suffer the wide impedance variation of an all-pass crossover or the large magnitude response ripple of a Butterworth crossover. I do not know which, if either, is the biggest problem, but I might use a Butterworth crossover when it must be passively realized and reserve the all-pass networks or active crossover systems.

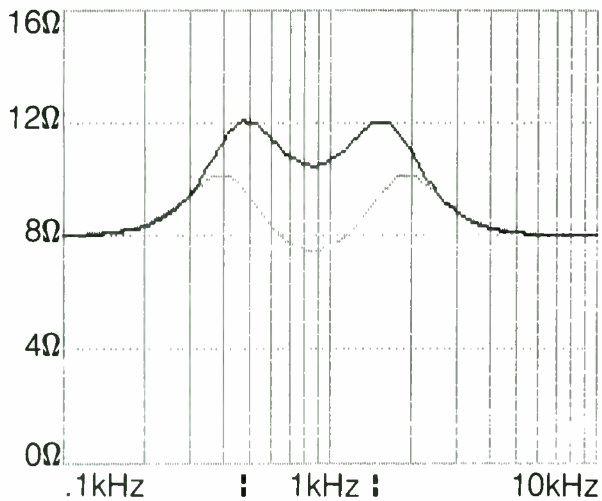


FIGURE 12: Input impedance of a three-way third-order all-pass crossover network with crossover frequencies of 500Hz and 1.5kHz. and bandpass polarity reversed. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

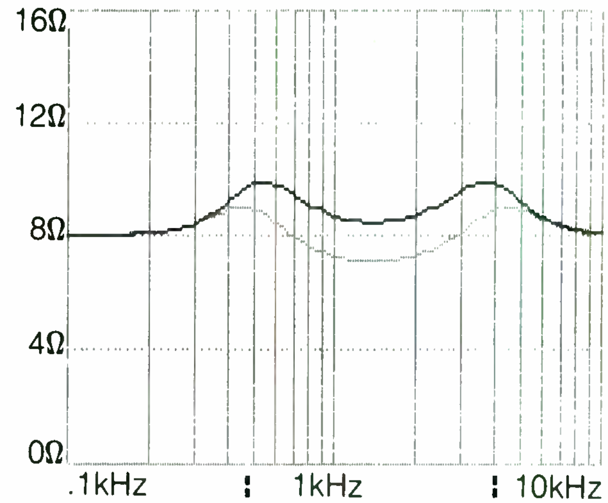


FIGURE 13: Input impedance of a three-way third-order all-pass crossover network with crossover frequencies of 500Hz and 4kHz. and bandpass polarity reversed. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

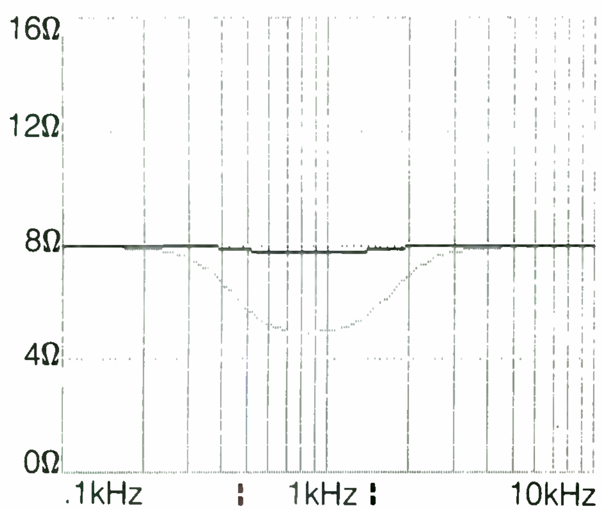


FIGURE 14: Input impedance of a three-way third-order Butterworth crossover network with crossover frequencies of 500Hz and 1.5kHz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

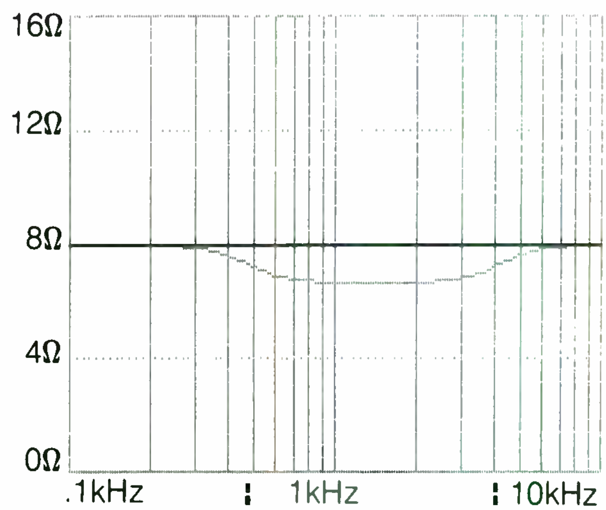


FIGURE 15: Input impedance of a three-way third-order Butterworth crossover network with crossover frequencies of 500Hz and 4kHz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

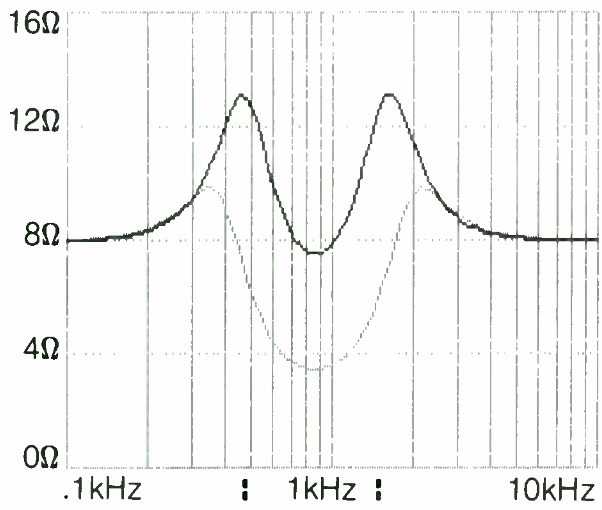


FIGURE 16: Input impedance of a three-way fourth-order all-pass crossover network with crossover frequencies of 500Hz and 1.5kHz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

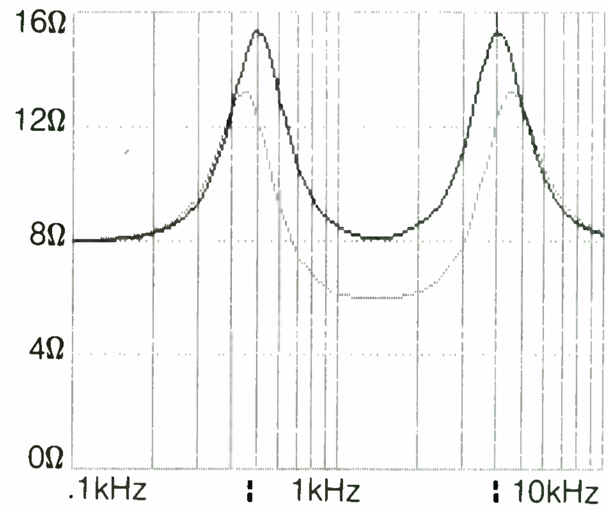


FIGURE 17: Input impedance of a three-way fourth-order all-pass crossover network with crossover frequencies of 500Hz and 4kHz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

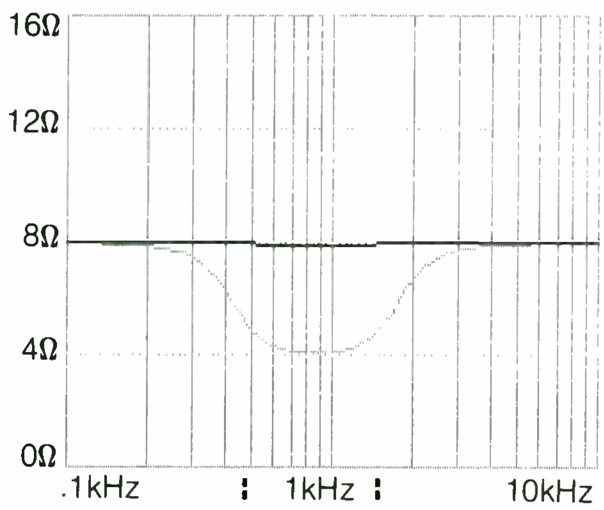


FIGURE 18: Input impedance of a three-way fourth-order Butterworth crossover network with crossover frequencies of 500Hz and 1.5kHz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

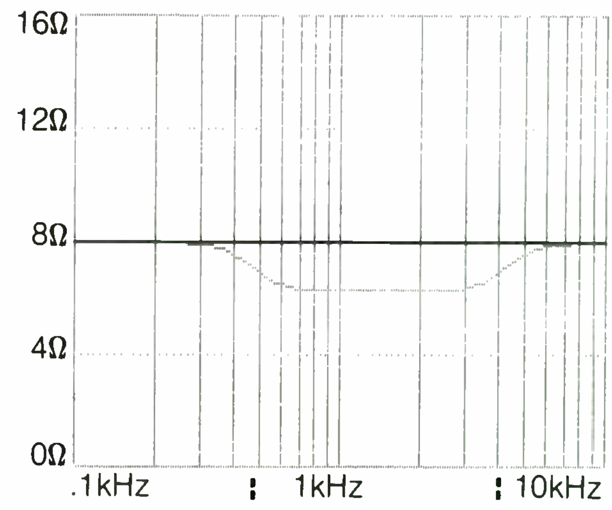


FIGURE 19: Input impedance of a three-way fourth-order Butterworth crossover network with crossover frequencies of 500Hz and 4kHz. [Dark curve is with T-bandpass topology, light with C-bandpass topology. Marks | locate crossover frequencies.]

```
385  IF NC=3 THEN PRINT "DELETE RA(YES=1,NO=0)? " : INPUT Z9
3135 IF Z9=1 THEN GOSUB 6000
3185 IF Z9=1 THEN GOSUB 6100
3545 IF Z9=1 THEN GOSUB 6000
3605 IF Z9=1 THEN GOSUB 6000
3715 IF Z9=1 THEN GOSUB 6100
4095 IF Z9=1 THEN GOSUB 6000
4215 IF Z9=1 THEN GOSUB 6000
4465 IF Z9=1 THEN GOSUB 6100
5075 IF Z9=1 THEN GOSUB 6000
5235 IF Z9=1 THEN GOSUB 6000
5485 IF Z9=1 THEN GOSUB 6100
6000 RA = 0  :  RO = R2
6020 Z8 = 20*LOG(K/H)/LOG(10)
6030 RETURN
6100 PRINT "DELETE RA OPTION CHOSEN.  THIS CAUSES"
6110 PRINT "EXCESS BANDPASS GAIN OF " ; Z8 ; " DB."
6120 RETURN
```

For the delete RA option, add these statements to the PASSIVE THREE-WAYS program Crossovers Diskette (Old Colony SBK-F1A)

LISTING 1: Code to update passive three-ways program in Old Colony's crossovers disk.

FAST AND EASY ACTIVE FILTER CALCULATIONS

BY PAUL W. GRAHAM

When first interested in building my own electronic filters and crossovers, I was mystified about how to calculate circuit values for desired frequencies. Tedeschi's work[1] along with Hoenig's fine op amp tutorial[2] taught me how to do this. *Figure 1* shows the math path I used to synthesize a practical design algorithm. For an in-depth look at the fundamentals from which this proceeds, the reader is encouraged to see Hoenig's and Tedeschi's books.

The algorithm is derived specifically for Butterworth second-order, unity gain, voltage-controlled/voltage source (VCVS) op amp circuits. (VCVS second-order filters are also known as Sallen & Key filters.) The -3dB corner frequency is defined throughout this discussion by F_X. Fourth-order filters are possible simply by cascading a second stage; if the circuit values in both stages are identical, F_X is a corner frequency at -6dB. One example of this special use of B4 filters is the Linkwitz-Riley crossover[3]

My objective was to find the relation between F_X, circuit C, and circuit R in its simplest form, and to find the value of the constant relating them—i.e., the "recipe" and the "magic number." With this algebraic reduction, it should be possible to develop a computer program that relies on the inherent simplicity of Butterworth unity gain, Sallen & Key filters to design filters and crossovers at or near any desired frequency.

Figure 2 is the practical circuit configuration. Note that the circuit values are the same for low-pass and high-pass filters; the only difference between them is that the circuit positions of R and C are reversed. The constant has its decimal point adjusted so that F_X is always in hertz (Hz), C is always in microfarads (μF), and R is always in kiloohms (kΩ).

The circuit is configured with the assumption that, in practice, the physical size of resistors makes the parallel connection comprising ½R in the high-pass filter completely feasible, whereas selecting C so that ½C is a standard value also, permits one physically smaller capacitor to be used in the low-pass filter instead of two larger ones in series. Thus, to build a practical circuit, you would begin by defining C such that ½C is standard, then isolating F_X as a function of R value, whose wider selection is not related to physical size.

Any simple calculator, preferably with memory, can operate on the formulas to get accurate results. If, however, you want to find what the F_X range might be within component tolerance windows or you want to do some arithmetic experimentation with catalog data, the button-pushing can get tedious. *Listing 1* is the source code for a program that will do this effortlessly. The program was built on Commodore BASIC and requires a printer to run. It could be modified with minor syntactical alterations for other machines. The sidebar is a capsule description of what the program can do and how to use it.

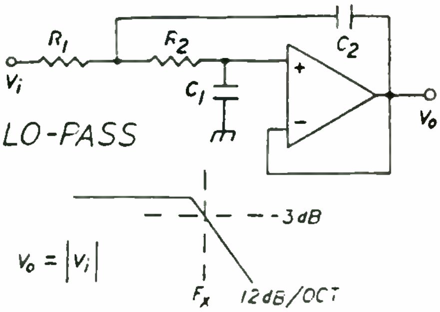
Tables 1, 2 and *3* are Autocalc results for three values of C. The Mallory SXL line of 63WV DC, 2.5 percent polystyrene caps offers these pairs as standard: 0.01/0.02, 0.011/0.022, and 0.012/0.024 (μF). The tables refer to these pairs. The computer reads data that contains all standard R values from 10kΩ to 100kΩ. Note that the tables do not cover all possible F_X values for all possible R values. Inspection of the first and last lines of each table, however, reveals a general principle at work.

If a resistor is reduced by 10^{-n}, then F_X is increased by 10^{+n}, and vice versa. As is the case for 10kΩ and 100kΩ, for any combination of significant figures, only the decimal places change. This is also true for the capacitor value. Looking again at the basic formulas, we see that regardless of the relative position of the three variable terms (F_X, C, and R), the isolated variable is inversely proportional to the product of the other two, related only by a constant with a fixed decimal. Therefore, any multiplier y on one factor, if offset by a multiplier 1/y on the other factor, will hold the isolated term at the same value, because any y/y equals 1.

Note, for instance, that for a single value of C, the F_X for 10kΩ is twice that for 20kΩ, three times that for 30kΩ and so on. Let's consider the special case where y equals 10 and n equals any power. Values for R and C are regularly available incremented by decimals and decades. The F_X value change, for any combination of significant figures for R and C, results only in moving its decimal point. With this in mind, these two convenient expressions derive:

1. If R is held constant, a 10^{-n} change in C will cause F_X to be located n decade(s) up-spectrum, and a 10^{+n}

SALLEN & KEY A.F. FILTERS

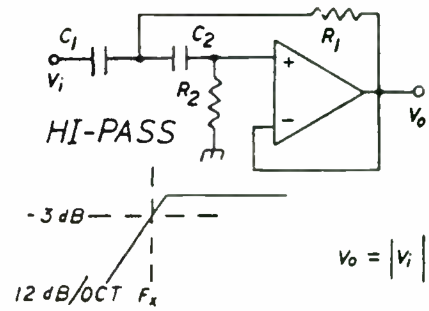


$\eta = 2$ (2nd ORDER)

BUTTERWORTH TRANSFER FUNCTION POLYNOMIAL TERMS:

$b_0 = 1 \quad b_1 = \sqrt{2}$

$K = 1$ (UNITY GAIN)



For the special case $\eta = 2$

$C_2 = 2C_1$; make $C_1 = 0.5\,F$

$$G_1 = \frac{b_1 + \sqrt{b_1^2 - 4b_0(C_1 + 1 + K)}}{2}$$

For the special case $K = 1$ $\quad G_1 = .707$

$$G_2 = \frac{C_1 b_0}{G_1} = .707 \quad \therefore \quad G_1 = G_2 = G \text{ and } R_1 = R_2 = R$$

$\mu = 2\pi F_x$; $\overline{IS} = F_x/20\pi$

$C_{1p} = C_1/\mu\overline{IS} = 5/F_x^2$; $C_{2p} = 10/F_x^2$

For μF $\quad C_2 = 10^7/F_x^2$; $C_1 = .5C_2$

$R = \overline{IS}/G = F_x/20\pi\sqrt{.5}$

For the special case $\eta = 2$

$C_1 = C_2 = C$; make $C = 1\,F$

and $a_0 = 1/b_0 = 1$; $a_1 = b_1/b_0 = \sqrt{2}$

$$G_2 = \frac{a_1 + \sqrt{a_1^2 + 8a_0(K-1)}}{4}$$

For the special case $K = 1$ $\quad G_2 = .707$

$G_1 = a_0/G_2 = 1.4142$

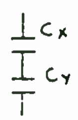
$\mu = 2\pi F_x$; $\overline{IS} = F_x/20\pi$

$C_p = C/\mu\overline{IS} = 10/F_x^2$; For μF $\quad C = 10^7/F_x^2$

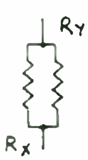
$R_1 = \overline{IS}/G_1 = F_x/20\pi\sqrt{2}$

$R_2 = \overline{IS}/G_2 = F_x/20\pi\sqrt{.5}$ $\quad \therefore \quad R_1 = .5R_2$

Note: For a given F_x, C_2 in lo-pass = C in hi-pass & R_2 in hi-pass = R in lo-pass.
C_1 in lo-pass = $.5C$ & R_1 in hi-pass = $.5R$.

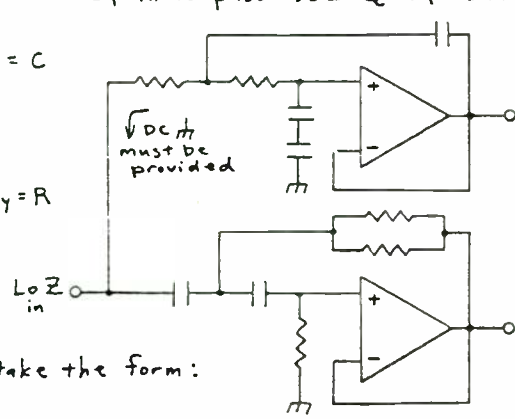


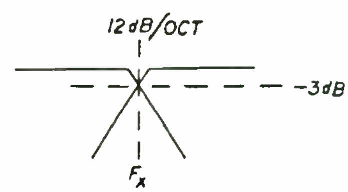
where $C_X = C_Y = C$
$C_T = .5C$



where $R_X = R_Y = R$
$R_T = .5R$

∴ A crossover may take the form:





wherein all R=R & all C=C.

To standardize C select any value C_S; conversion factor $X_k = C/C_S$.
New $R_N = X_k R$. All such combinations are functionally equivalent.
To standardize R select value $R_S \simeq R_N$; actual F_x resulting is found by

$$R_k = R_S/X_k \ ; \ \overline{IS} = .707 R_k \ ; \ F_x = 20\pi\overline{IS}.$$

Where $C\ (\mu F)$, $F_x\ (Hz)$, $R\ (\Omega)$, $\quad C_p = 10^7/F_x^2$; $X_k = C_p/C_s$; $R = F_x/20\pi\sqrt{.5}$.

$$R_N = X_k R = \frac{(10^7/F_x^2)(F_x/44.429)}{C_s} \ ; \text{ for } K\Omega,\ \times 10^{-3}. \ \therefore \ R = \frac{225.078}{F_x C}$$

$$\boxed{F_x = \frac{225.078}{RC}}$$

FIGURE 1: Math path for determining circuit values for Sallen & Key audio-frequency filters.

change in C will cause F_X to be located n decade(s) down-spectrum.

2. If C is held constant, a 10^{-n} change in R will cause F_X to be located n decade(s) up-spectrum, and a 10^{+n} change in R will cause F_X to be located n decade(s) down-spectrum.

Therefore, to get F_X values higher than those taken directly from an Autocalc table, simply move the decimal point in the F_X values to the right and that in the R values to the left by the same number of places. To get F_X values lower than those taken directly from an Autocalc table, move the decimal point in the F_X values to the left and that in the R values to the right by the same number of places. Since C is always in microfarads, R will always be in kiloohms, and F_X will always be in hertz.

Of course, we could expand our data statements to include all R values from, say 0.1kΩ (100Ω) to 1,200kΩ (1.2MΩ) and read directly from a table two pages long. With the floating decimal notation, this is unnecessary. Any value of R that would produce an F_X in the audible spectrum is available from RCA's SK line of ¼W, 2 percent precision metal film resistors. A 2 percent tolerance for R was, therefore, used in these tables.

The F_X column in the tables is the F_X nominal resulting from the ideal case, wherein C and R values are at their centerline. Since, in reality, C and R values vary within their tolerance window ratings, it is worth knowing what effects such deviations could have on the F_X prediction. It follows from the foregoing discussion that the width of the range Hz window, regardless of F_X nominal, is always the same, based on octave scale measurement. The outputs for range Hz give the prediction window between possible worst-case low- and high-frequency error for either filter. We can express the following conditions as derivations of the filter math:

1. Worst-case low-frequency error occurs when all R and C values are at the greatest positive deviation.
2. Worst-case high-frequency error occurs when all R and C values are at the greatest negative deviation.
3. Worst-case F_X spread in crossovers occurs when Condition 1 describes the low-pass filter and Condition 2 describes the high-pass filter.
4. Worst-case F_X overlap in crossovers occurs when Condition 1 describes the high-pass filter and Condition 2 describes the low-pass filter.

For a crossover, the slope intersections will deviate from perfect F_X down-point axis symmetry unless the respective errors are complementary-proportional—i.e., vary in the same direction by the same amount. While the speaker drivers may tolerate a fairly broad window (in fractions of an octave), it is desirable to use close tolerance components to reduce indeterminate symmetry axis errors, which ultimately affect acoustical summing characteristics.

The error-range end-point F_X values defining this window are possible only

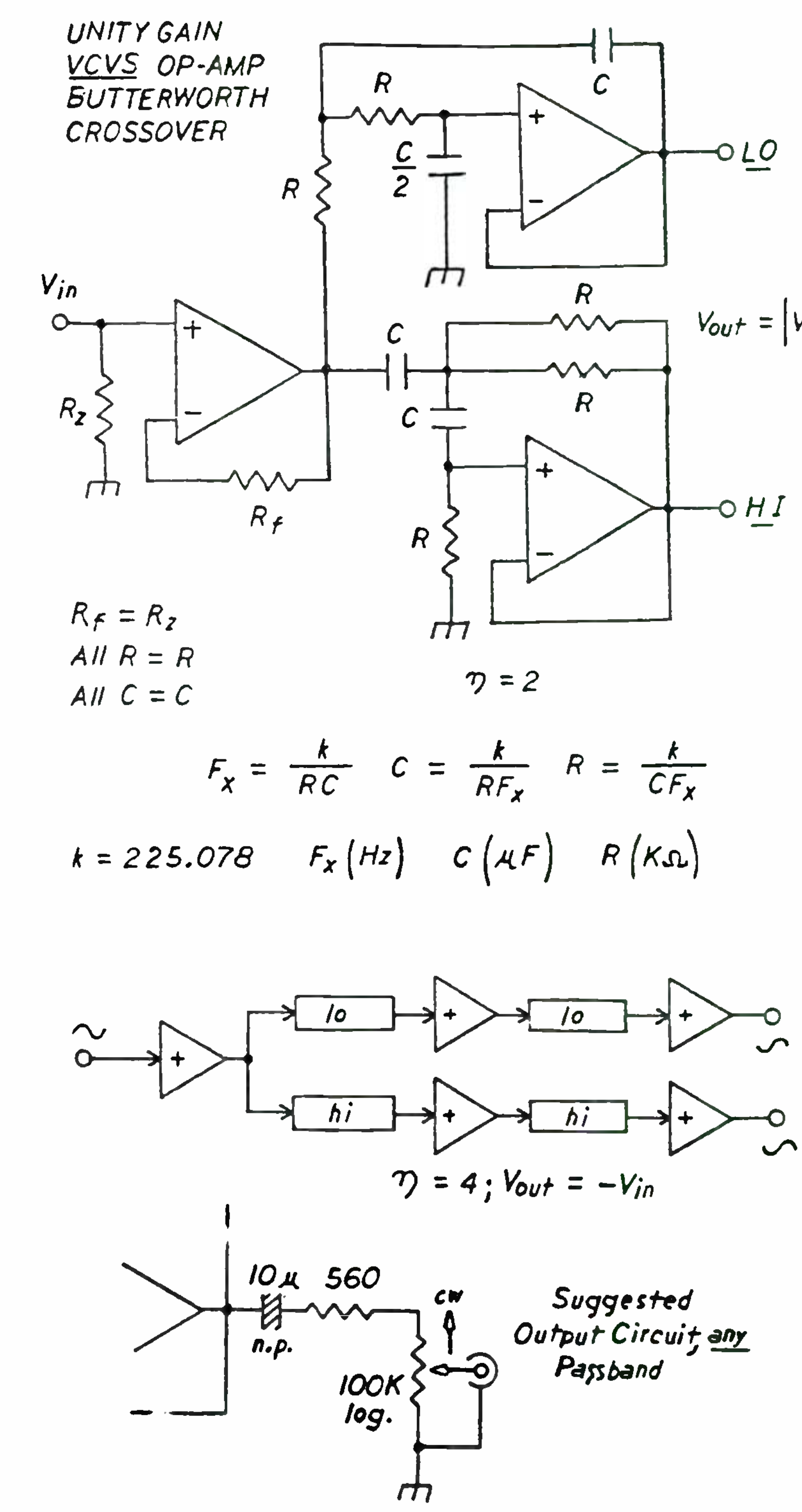


FIGURE 2: Circuit designed with use of math path in Figure 1.

in the event that all component values are at maximum rated deviation from centerline. Short of ready access to a Wheatstone bridge (and readiness to reject resistors), an evaluation of the real effects of deviations can only be treated statistically. The following general statements are appropriate:

1. The incidence of deviation probably decreases geometrically with the severity of deviation. For random sampling of a production pool of any one component value, it is likely that a small percentage will be at maximum rated deviation.
2. In crossovers, the probability of either Condition 3 or Condition 4 is decreased in a manner related to and geometrically mitigated by the incidence-severity curve described above.
3. In multipole cascades, the statistical universe is enlarged, resulting in further error averaging.

Taken together, these factors justify a truncation of the positive and negative deviations in the interests of reasonably practical prediction. I have incorporated a 75 percent-of-tolerance-rating multiplier for both R and C. This is speculative, but I do not think overly optimistic.

I have built several Linkwitz-Riley crossovers and fourth-order infrasonic filters for bass reflex alignments, built up from Sallen & Key cascades. The first crossover I made was checked on a Hewlett-Packard ⅓-octave real-time analyzer for experimental verification of the algorithm. Subsequent models were

```
10 REM SALLEN & KEY A.F. FILTERS:COPYRIGHT(C) PAUL W. GRAHAM,1987:CBM
12 OPEN1,4:I$="ACTIVE FILTER CIRCUIT VALUES":J$="VCVS OP-AMP, B2 UNITY"
14 A$="AUTOCALC":B$="FX FROM RC":C$="R FROM C FOR FX":D$="C FROM R FOR FX"
16 PRINTI$:PRINTJ$:PRINT#1,I$", "J$:PRINT:PRINT"SOLVE FOR FX, C, OR R"
18 PRINT#1:FORX=1TO50:PRINT#1,"-";:NEXTX:PRINT#1:PRINT:PRINT"SELECT:":PRINT
20 PRINT"A-"A$:PRINT"B-"B$:PRINT"C-"C$:PRINT"D-"D$:PRINT"E-DONE":PRINT
22 GETZ$:IFZ$<>"A"ANDZ$<>"B"ANDZ$<>"C"ANDZ$<>"D"ANDZ$<>"E"THEN22
24 E$="CAPACITOR":F$=" 50 WVDC":K$="RANGE":IFZ$="E"THENPRINT"DONE":END
26 G$="RESISTOR":H$=" 1/4 W":P$=CHR$(16):IFZ$="D"THENPRINTD$:PRINT:GOTO56
28 IFZ$="C"THENPRINTC$:PRINT:GOTO50:REM Z$ SELECTOR ORDERS I/O SEQUENCES
30 L$="(FX*10↑+N):(R*10↑-N)":IFZ$="B"THENPRINTB$:PRINT:GOTO50
32 M$="(FX*10↑-N):(R*10↑+N)":IFZ$="A"THENPRINTA$:PRINT:GOTO50
34 CA=.75*CT:CL=C*(100-CA)/100:CH=C*(100+CA)/100:FX=INT(22507.8/(R*C)+.5)/100
36 RA=.75*RT:RL=R*(100-RA)/100:RH=R*(100+RA)/100:IFZ$="A"THENPRINT"R="R"KO"
38 FL=INT(22507.8/(RH*CH)+.5)/100:FH=INT(22507.8/(RL*CL)+.5)/100
40 PRINT"FX="FX"HZ NOMINAL":PRINTK$" HZ="FL"-"FH:PRINT:IFZ$="A"THEN74
42 PRINT#1:PRINT#1,E$:PRINT#1,"C="C"UF"CT"%"F$:PRINT#1:IFZ$="A"THEN70
44 PRINT#1,G$:PRINT#1,"R="R"KO"RT"%"H$:PRINT#1:PRINT#1,"FX="FX"HZ NOMINAL"
46 PRINT#1,K$" HZ="FL"-"FH:IFZ$="B"THEN18:REM ALL ROUTINES DONE GO TO MENU
48 IFZ$="C"ORZ$="D"THENPRINT"A-NEW FX     B-RETURN TO MENU":PRINT:GOTO64
50 PRINTE$F$:INPUT"C UF";C:INPUT"TOL+OR-%";CT:PRINT:IFZ$="C"THEN58
52 IFZ$="A"THENPRINTG$H$:INPUT"TOL+OR-%";RT:PRINT:GOTO42
54 IFZ$="D"THEN34:REM LINE 34 BEGINS MAIN MATH FOR ALL ROUTINES
56 PRINTG$H$:INPUT"R KO";R:INPUT"TOL+OR-%";RT:PRINT:IFZ$="B"ORZ$="C"THEN34
58 IFZ$="C"ORZ$="D"THENINPUT"FX HZ";FX:PRINT:PRINT"STANDARDIZE:"
60 IFZ$="C"THENRX=INT(22507.8/(C*FX)+.5)/100:PRINT"R="RX"KO":GOTO56
62 IFZ$="D"THENCX=INT(2250780/(R*FX)+.5)/10000:PRINT"C="CX"UF":GOTO50
64 GETY$:IFY$<>"A"ANDY$<>"B"THEN64:REM Y$ IS IN-ROUTINE SUBMENU SELECTOR
66 IFY$="A"THENPRINT#1:FORX=1TO50:PRINT#1,"-";:NEXTX:PRINT#1:GOTO58
68 IFY$="B"THEN18:REM 'A' REMAINS IN ROUTINE, 'B' EXITS AND GOES TO MENU
70 PRINT#1,G$RT"%"H$:PRINT#1:PRINT#1,P$"01";"R KO";P$"10";"FX HZ";P$"26";K$
72 PRINT#1:FORR=10TO100:READR:GOTO34:REM DATA IS ALL EIA STANDARD R VALUES
74 CLOSE1:OPEN2,4,2:O$=" 999    99999.99    99999.99  99999.99"
76 PRINT#2,O$:CLOSE2:OPEN1,4,1:PRINT#1,R,FX,FL,FH:CLOSE1:OPEN1,4:NEXTR
78 DATA 10,11,12,13,15,16,18,20,22,24,27,30,33,36,39,43,47,51,56,62
80 DATA 68,75,82,91,100:PRINT#1:PRINT#1,P$"08";L$:PRINT#1,P$"08";M$
82 PRINTL$:PRINTM$:PRINT:PRINTA$" COMPLETE FOR C="C"UF":RESTORE:PRINT:GOTO18
```

LISTING 1: Source code for a program to calculate F_X, C, and R for Sallen & Key filters.

TABLE 1

ACTIVE FILTER CIRCUIT VALUES, VCVS OP AMP, B2 UNITY

C = 0.02μF, 2.5%, 50WV DC
R = 2%, ¼W

R (kΩ)	F_X (Hz)	Range (Hz)	
10	1125.39	1088.35	1164.36
11	1023.08	989.41	1058.51
12	937.83	906.96	970.30
13	865.68	837.19	895.66
15	750.26	725.57	776.24
16	703.37	680.22	727.72
18	625.22	604.64	646.87
20	562.69	544.18	582.18
22	511.54	494.71	529.25
24	468.91	453.48	485.15
27	416.81	403.09	431.24
30	375.13	362.78	388.12
33	341.03	329.80	352.84
36	312.61	302.32	323.43
39	288.56	279.06	298.55
43	261.72	253.11	270.78
47	239.44	231.56	247.74
51	220.66	213.40	228.31
56	200.96	194.35	207.92
62	181.51	175.54	187.80
68	165.50	160.05	171.23
75	150.05	145.11	155.25
82	137.24	132.73	142.00
91	123.67	119.60	127.95
100	112.54	108.84	116.44

$(F_X \times 10^{+N}) : (R \times 10^{-N})$
$(F_X \times 10^{-N}) : (R \times 10^{+N})$

TABLE 2

ACTIVE FILTER CIRCUIT VALUES, VCVS OP AMP, B2 UNITY

C = 0.022μF, 2.5%, 50WV DC
R = 2%, ¼W

R (kΩ)	F_X (Hz)	Range (Hz)	
10	1023.08	989.41	1058.51
11	930.07	899.46	962.28
12	852.57	824.51	882.09
13	786.99	761.09	814.24
15	682.05	659.61	705.67
16	639.43	618.38	661.57
18	568.38	549.67	588.06
20	511.54	494.71	529.25
22	465.04	449.73	481.14
24	426.28	412.25	441.05
27	378.92	366.45	392.04
30	341.03	329.80	352.84
33	310.02	299.82	320.76
36	284.19	274.84	294.03
39	262.33	253.70	271.41
43	237.93	230.10	246.16
47	217.68	210.51	225.21
51	200.60	194.00	207.55
56	182.69	176.68	189.02
62	165.01	159.58	170.73
68	150.45	145.50	155.66
75	136.41	131.92	141.13
82	124.77	120.66	129.09
91	112.43	108.73	116.32
100	102.31	98.94	105.85

$(F_X \times 10^{+N}) : (R \times 10^{-N})$
$(F_X \times 10^{-N}) : (R \times 10^{+N})$

TABLE 3

ACTIVE FILTER CIRCUIT VALUES, VCVS OP AMP, B2 UNITY

C = 0.024μF, 2.5%, 50WV DC
R = 2%, ¼W

R (kΩ)	F_X (Hz)	Range (Hz)	
10	937.83	906.96	970.30
11	852.57	824.51	882.09
12	781.52	755.80	808.58
13	721.40	697.66	746.38
15	625.22	604.64	646.87
16	586.14	566.85	606.44
18	521.01	503.87	539.06
20	468.91	453.48	485.15
22	426.28	412.25	441.05
24	390.76	377.90	404.29
27	347.34	335.91	359.37
30	312.61	302.32	323.43
33	284.19	274.84	294.03
36	260.51	251.93	269.53
39	240.47	232.55	248.79
43	218.10	210.92	225.65
47	199.54	192.97	206.45
51	183.89	177.84	190.25
56	167.47	161.96	173.27
62	151.26	146.28	156.50
68	137.92	133.38	142.69
75	125.04	120.93	129.37
82	114.37	110.60	118.33
91	103.06	99.67	106.63
100	93.78	90.70	97.03

$(F_X \times 10^{+N}) : (R \times 10^{-N})$
$(F_X \times 10^{-N}) : (R \times 10^{+N})$

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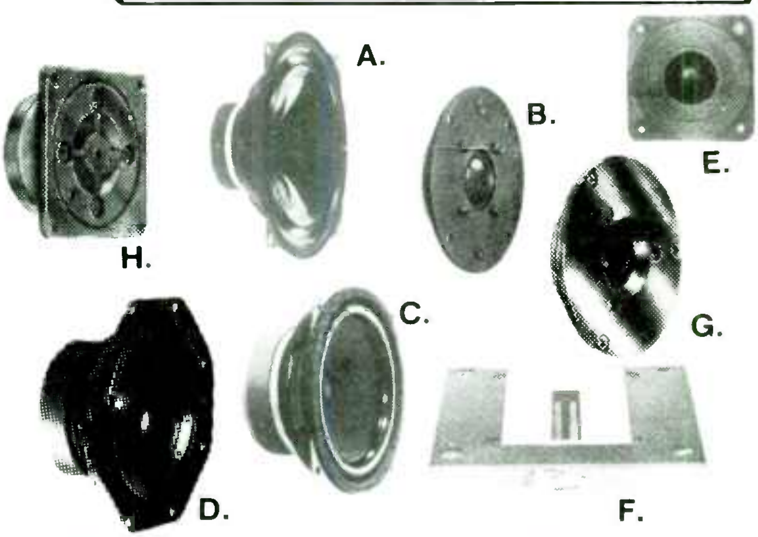

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The computer deals with frequency in hertz (Hz), capacitance in microfarads (μF), and resistance in kiloohms (kΩ). Adjust all input value decimals for correct denomination.

You may choose any one of four procedures. When finished with a procedure, you may end or choose another procedure. You may alternate or repeat procedures in any order with any frequency.

A—AUTOCALC: If you press A, the computer will ask you for value of C, tolerance rating for C, and tolerance rating for R, in that order. It will then consult the EIA standard R values and develop a table listing F_X for each R.

B—F_X FROM RC: If you press B, the computer will ask you for value of C, tolerance rating for C, value of R, and tolerance rating for R, in that order. It will then return F_X .

When completed with either A or B, the computer will go back to menu.

C—R FROM C FOR F_X: If you press C, the computer will ask you for value of C, tolerance rating for C, and desired F_X, in that order. It will then return value of R and ask you to standardize. You enter nearest R value you are able to use, then the tolerance rating for R. The computer will then return F_X for the standardized R.

D—C FROM R FOR F_X: If you press D, the computer will ask you for value of R, tolerance rating for R, and desired F_X, in that order. It will then return value of C and ask you to standardize. You enter nearest C value you are able to use, then the tolerance rating for C. The computer will then return F_X for the standardized C.

When completed with either C or D, the computer will give you a choice: A—NEW F_X or B—RETURN TO MENU. If you press A, it will stay in the procedure and ask you for another desired F_X. Since it remembers the given C or R, you do not have to start at the beginning of the procedure and reenter the same value. It will then complete the procedure as before. You may repeat this as often as desired or press B to go back to the menu.

For procedures B, C, and D, the printer will record C, R, F_X and will draw lines between blocks of output to keep the information organized.

NOTES:

1. The menus are goofproof. If you press a key not on the menu, the computer will wait until you press a key that is on the menu. The inputs are *not* goofproof. If you enter zero or inadvertently hit return, you will get a divide-by-zero error message. In this case, you must rerun.
2. The type and structure of statements used to generate the Autocalc tables may vary depending on your preference and printer requirements. This particular arrangement works well with the Star Micronics SG-10C.
3. This program will run on any Commodore machine using CV2.0. A VIC-20 does not require extra RAM. Without a printer (device no. 4), you will get a device-not-present error.

In Memoriam Richard C. Heyser

1931–1987

BY PETER E. SUTHEIM

A TECHNICAL APPRECIATION

In the first part of this "technical appreciation" of the work of Richard Heyser, published in *SB*'s previous issue, I tried to present some of his early work, particularly time-delay spectrometry (TDS), which had profound practical consequences for audio instrumentation, and to suggest his style of working, in formal papers and in his relations with colleagues. TDS is now widely used; at least two manufacturers—Bruel & Kjaer (B&K) and Tecron—produce commercial devices that have grown out of Heyser's inventions. (They are made under a licensing agreement.)

More problematical is Heyser's later work, which sometimes seems to have a definite beginning, and sometimes appears to have a seamless continuity with his earliest publications—depending, perhaps, on where one stands, or what frame of reference one uses (a play on his own work that Heyser—who died in March—might have appreciated).

That work, distilled to the purest essence I can cook out of it, amounts to this: What perceptive listeners hear from audio systems is not well described by our conventional audio measurements; the correlation between perceptions and measurements is poor, and consequently the measurements are not useful in predicting how a particular system or component will sound. For example, there exist no measurements that will reliably predict the imaging qualities of a speaker system, or the subjective sense of unlimited dynamic range that some amplifier-speaker combinations have, and others do not. What determines the subjective impression of "graininess" or "transparency"?

Even though this discrepancy is quite apparent, the same traditional measurements (frequency response, total harmonic distortion, rise time, etc.), continue to be made by manufacturing engineers and technically inclined equipment reviewers, while "subjective" reviewers fill their reports with adjectives that are poorly defined and highly subject to individual interpretation. The two camps show little or no interest in resolving this peculiar condition, and sometimes use abusive language in referring to each other. The situation seems to have become accepted as natural and permanent.

Heyser's discovery, some 20 years ago, which he says dawned on him with a kind of shock, is that the two factions look out over the same terrain, but with *two different frames of reference each incorporating different dimensionality.* The engineers are making more and more precise measurements in a coordinate system that has one dimension (voltage, for example), or two (time and frequency), while the listener-evaluators are making their increasingly refined judgments in a realm that is clearly multidimensional (loudness, pitch, timbre and location, for example, are four independent, or very nearly independent, coordinates in a "natural" acoustic experience). This is not to say that one is "better" than the other, only that there should be no suprise that the two camps can't talk to each other.

Well developed mathematical tools are available to resolve this difficulty, Heyser asserted, and they are to be found in geometry (used in the broad sense of modern math, not in the limited sense of high school classes). Heyser then began to show, systematically in a dozen or so papers over at least as many years, how geometries with different dimensionality could be "mapped" into each other, proving that the multidimensional realm of the music listener is encoded, or enfolded, into the time/frequency domains familiar to the audio engineer. In the process, he showed that some popular mathematical tools were being used improperly because of misleading assumptions, now solidly entrenched, first made in the late 19th century. Pitch, for example, does not map simply onto frequency—which was an abstraction adopted because it seemed to correlate conveniently with the subjective sensation of pitch.

He showed how one might handle "curved" geometries, in which coordinates are not fully independent, but pull on each other in an interdependent way. This made it possible for the first time to relate the subjective phenomenon of image-smearing to specific and distinct technical properties of amplifiers or loudspeakers. Throughout, he insisted that measurements, no matter how accurate or traditionally respected they might be, are useless if they do not predict what a listener will hear, and that the listener's assessment must be taken as the starting point, because the entire purpose of recording and reproducing sound is to create a convincing sonic illusion in the listener's mind.

In a March 1979 article in *Audio*, Richard Heyser wrote these paragraphs:

"There is evidence from studies of the brain that the perception of music is different from the perception of language, and that words which are spoken are perceived differently than [sic] words which are sung. There is also evidence that the way in which we perceive certain natural sounds, whether as music or language, may be related to cultural differences and learning experience. If these, and many other such things in our perception of sound, be true, then where in our audio technology do we address such factors? If not, then why not?

In an earlier discussion I broached the issue of the end product of audio. [It] is the listening experience...not meter readings, or wiggles on an oscilloscope, or piles of charts and graphs. The end product is that very private and personal experience we have when listening to reproduced sound.

If we are ever going to put a number on the quality of that experience, then it is clear that we must do more than specify the cosmetic perfection of a waveform or pursue an endless quest of reducing measurable distortions on laboratory signals which may have little bearing on the process of perception of sound.

"Somehow in our technical considerations of audio we must also recognize the role played by human emotion. Aggression, paradox, strength of opinion, and conflict of interest may not be considered as control variables by an audio designer, but they can be very important in determining the success of the product which he designs."

Conceivably—the real instrument, and the stereo illusion—that could seem the same to a listener, would appear quite different when measured.

This is not the normal language of the pocket-calculator platoon comfortable with coverage angles and headroom and grounding methods. For most of his working life, Richard Heyser was painfully aware of his stance beyond the pale of what is usually understood as technical audio—the realm of hardware and hard numbers. The closest any of his colleagues ever heard this gentle, cheerful soul verge on bitterness was when he confronted what seemed a general unwillingness—or was it really inability?—to understand what he was getting at.

THE LISTENING EXPERIENCE. What he was getting at—what he quite literally devoted his life to—was, in his own words, "one of the greatest problems of audio engineering, learning how to analyze and measure what we 'hear'". From a study of his many papers and popularizations, it is clear that this occupied him for about 20 years, and that he dug determinedly and confidently and systematically into regions of thought where he knew many would not follow. He saw that the mainstream of audio measurement was, in a fundamental way, a dead end: ever more refined measurements of distortion, or even the discovery of new kinds of distortion, were not going to lead to a better correlation between the measured numbers and the subjective "quality" or realism or listenability of a music reproducing system. More precisely, one could not successfully analyze a complex, nonlinear, many-dimensioned phenomenon—the listening experience—with linear mathematics.

Heyser concluded that at least five dimensions, or coordinates, are needed to accommodate even a minimum list of subjective descriptions encountered in audio. (A full exposition of this would require the duplication of his two-part paper, "The Delay Plane, Objective Analysis of Subjective Properties," which covered 18 pages in two consecutive issues of the *Journal* of the Audio Engineering Society, November and December 1973.) They are time delay, azimuth, elevation, intensity and pitch. As examples, time delay would include perceptions such as sequence, reverberation, echo, and range or distance. Azimuth and elevation are self-explanatory. Intensity and pitch are cross-correlated coordinates that include perceptions of loudness, scale or size, timbre, and pitch. Heyser calls this a "tentative list."

To relate these more closely to the language found in subjective reviews of equipment and recordings, Heyser offers several examples of possible usage, a few of which are quoted here: Change of timbre: bright, peaked, boomy; combined spectral-spatial: wander, shift, indistinct location.

In a later paper ("Geometry of Sound Perception," published only as a preprint for the 51st Convention of the AES in 1975), Heyser took a second whack at this immense and exciting problem, giving the audio engineering fraternity a kind of second chance and proposing a way "to reduce it to a three-dimensional thing. At least we can then sketch it on a piece of paper." He omits the time dimension by assuming "that the clock is frozen at 'now', whenever that may be." And he collapses elevation and azimuth into a single coordinate called "space", explaining almost breezily that "if we need to determine position we can always...go into that axis as a two-dimensional thing." The synoptic quality of this brief paper, and its reliance on graphical illustration rather than mathematical expressions, make it a brilliant exposition of this crucial part of Heyser's work. It's a pity that this piece never appeared in the AES *Journal*.

In a conversation taped in 1976 for broadcast over a weekly audiophile radio program on KPFK in Los Angeles, Heyser spoke of how the stereo illusion is created *in the mind*—in the perceptual realm—by two sources, something which a measuring microphone placed between the two loudspeakers could not "know".

"You may hear the sound of a string

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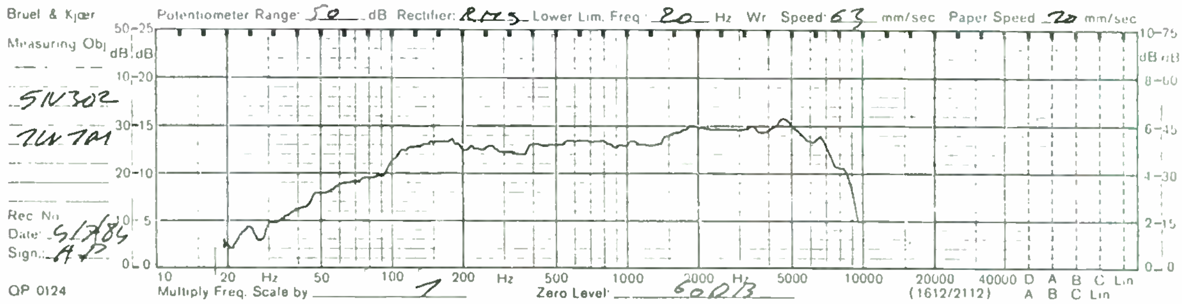

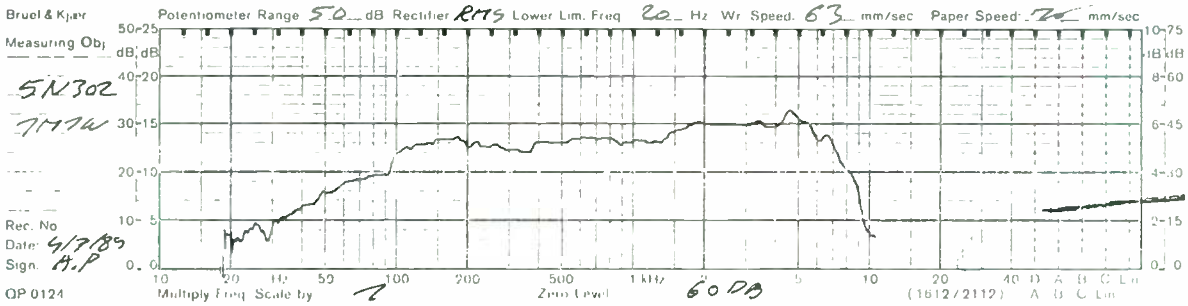

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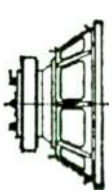
10 N 515 Neoflex 10" woofer	27.0	.33	145.3	250.	92	100 V	32	159.00



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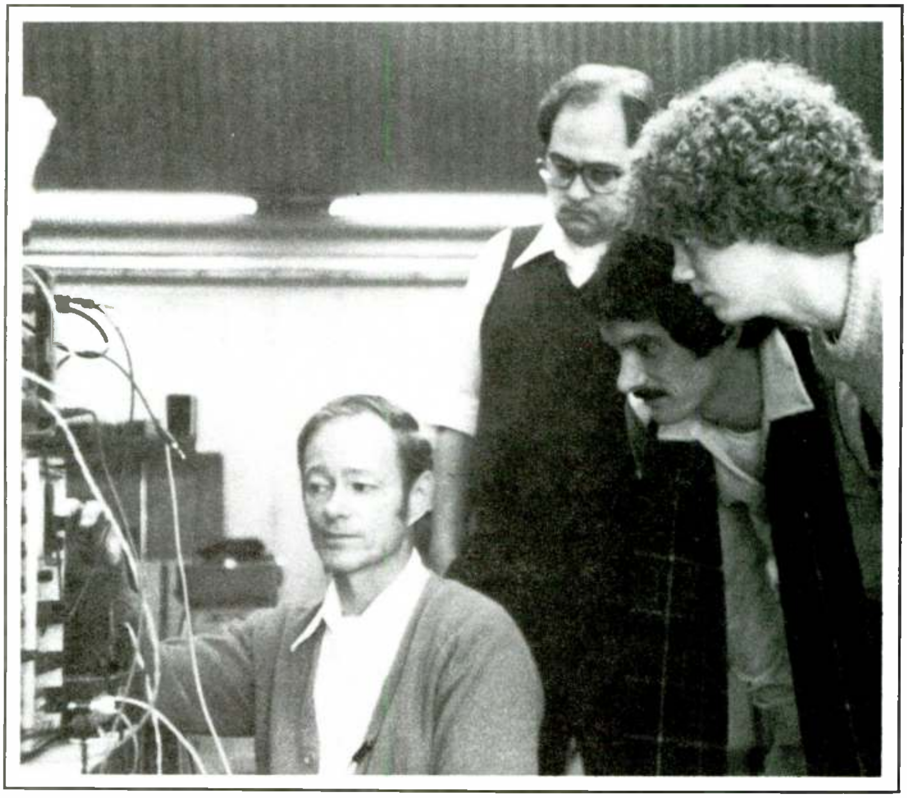
Conceivably, then, two situations—the real instrument, and the stereo illusion of the real instrument—that could seem the same to a blindfolded listener, would appear quite different when measured with a microphone in the listening room.

"Sound," as experienced by a listener, "has a *where*—it may be to the left of center and back a certain distance, and up a certain angle—it has a *when*, it has a tone, it has an intensity. And those are independent properties. You may have an oboe at a position, you may have a stringed instrument at that position. As far as a mathematical description is concerned, that is really a multidimensional expression. And yet, when we measure an amplifier, let's say the amplifier that fed the loudspeakers that gave that illusion, we measured volts as a function of time. We made a one-dimensional measurement. Now, in that volts-as-a-function-of-time is where, when, tone, how much. And if you have two channels the relationship between the two channels and the absolute values of each channel go to make that up. It is, in fact, an encoding.

"It makes no sense to put a sine wave into an amplifier, see how that sine wave is distorted, and try to relate that one-to-one to subjective distortion, until we can relate it to the way we describe sound. The fact that you may have, say, one percent harmonic distortion may interact, depending on the type of distortion, in [different ways].[1] For example, you may have a situation where the thing that gave rise to the distortion you measured caused a lateral smearing of the instruments. That would be one type. Another type that would give you the same *number* for harmonic distortion, but of course having subtle differences, would simply have given you a change in the tonal properties, the timbre, without a change in space.

"So let me define distortion as *a change in the type of geometry*. Now mathematically what happens when you do that, you find that your previously separated coordinates, like where and what tone, now become slightly cross-coupled. You get into what's called a curved geometry. There is now an interaction between where an object is and what tone it's playing. If you have such a distortion, defined in the geometric sense, you can in fact have a lateral smearing of instruments that's a function of pitch. Now a loudspeaker does that: the polar pattern does that. The effect of distortion, of an imperfection, in the 'left-rightness' of a loudspeaker in relation to pitch, is in fact to create a lateral change of the instrument's position. It's a cross-coupling of what was formerly independent properties. Now you find that they are no longer truly independent, but 'where' is a function of tone."



You must measure not only the amplitude of harmonic distortion, but also its phase.

APPLES AND ORANGES. Heyser's task, as he set it for himself, was to map out the enormously complex web of geometrical relations between subjective perceptions and objective measurements, so that there would eventually be enough reliable correlations to allow an observer in one realm to make predictions in the other. During the 1976 radio conversation he said (and it is almost equally true still today):

"You *cannot* say that one percent distortion is going to sound dirtier than one-tenth percent distortion. Flat out. Because you're talking apples and oranges. When I said 'one percent,' I did not specify everything about that device any more than if I'd said that a ± 1dB frequency-response speaker must sound cleaner than a ± 5dB speaker. And I would be in error for exactly the same reason. You must measure not only the amplitude of the harmonic distortion but its phase. If you imagine a device in which as the signal gets louder it creates harmonic distortion, in the conventional sense—that is, harmonics begin coming up which shouldn't be there—if you were to put in a pure-pitch tonal you may get a second harmonic, a third harmonic, but the phase of them may be a function of level. The timbre of an instrument that [naturally] has harmonics will change with sound level when played through this device [in a way that has no relationship to how the timbre of the actual instrument would change].

"Magnetic tape has a very high percentage of harmonic distortion. And many amplifiers have one-hundredth or one thousandth the measured harmonic distortion of tape, yet you'll hear that amplifier through the tape. Well, what are you hearing? You are perhaps hearing a deformation of the sound illusion that you cannot accept as being something natural. [This may be controversial but] if you can imitate nature in a distortion [so that it mimics] the way we hear, it will tend to be absorbed by us as a 'rightness' of perception. Our own hearing mechanism is logarithmic—very nonlinear. Suppose I create a type of distortion that mimics that, maybe increases it just a little bit in the same way, the effect might be that the sound sounds *louder*, but it doesn't really sound terribly distorted, even if the percentage of distortion by conventional measurement is very high. Harmonic distortion in a tape recorder follows almost a log law [similar to that of human hearing]; harmonic distortion in some of the early Class-B transistor amplifiers did not, and they would stand out.

"Now there are other reasons why they would stand out, particularly in stereo. [The old 'transistor sound' Class-B crossover-notch distortion is a good example.] At the zero-crossing, the very low signal levels have a different [lower] gain than the peaks. If you have a left and right channel, and the sound illusion is of an object to the left of center playing softly, and we have it play louder and louder and louder but stay at the same position in space. Now the left channel carries more signal level than the right channel. At extremely low sound levels, the right channel may be virtually nipped off, because you're in the crossover region. In the left channel, you now have bits of the peaks coming through, so what you have are high distortion fragments on the left channel only, nothing coming out of the right channel. (This is an extreme case, a *reductio ad absurdum*.)

"As you get louder, now the right channel begins to get the peaky distortion the left channel had previously, the left channel has more of the fundamental come up—the harmonic distortion ratio is dropping. So what you find if you trace the trajectory of the position in space of the fundamental and the harmonics, first you have nothing but harmonic distortion in the left channel only. As it gets louder, you find the distortion fragments begin to smear towards stage-center. The fundamental begins to rise from extreme stage-left to the proper stage position. So you have a situation where at moderate sound levels you may have a fundamental coming from about the right position in space, but the distortion fragments will be smeared laterally more over towards stage-center. That is a totally unnatural sound!

"Now take the same amplifier and put the object stage-center. Now what you have is distortion fragments that stay in the same position in space as [the sound] gets louder—the fundamental and the distortion fragments stay there. [So you have a system in which you get] a lateral smearing for stage-left and stage-right stereo illusion. The ability of a human to absorb that into what I call a 'rightness of perception' can be strained to the limit. You may not be able to put your finger on it, but [you] know it isn't right. It's brittle. It has all these nasty properties that people didn't like. [So that illustrates] that there was a genuine coupling between position in space of the illusion, its harmonic fragments and fundamental, and intensity, due to that one aberration, crossover distortion."

THE MAGICIAN'S BOX. To help illustrate the gulf between conventional measurements and our perceptions, Heyser invented a brilliant and comic device which he introduced at a "distortion workshop" held by the Los Angeles section of the Audio Engineering Society in 1980. The stated purpose of the workshop was to show those who attended how various kinds of distortion sounded. Several audio engineers worked up demonstrations, including a "box" that would add controlled amounts of harmonic or crossover or transient intermodulation distortion to steady tones and to musical program.

Most observers were astonished that the percentages had to get up into whole numbers before most of the distortions became audible—which was both a

striking discord with the expectation created by the audio industry, with its mania for double-zero fractional percentages, and thus also a perfect cue for Heyser to present his own box. It seems likely that he expected the bewilderment from the distortion-box demonstration, since he had been trying for years to convince engineers that they were sniffing the wrong fireplug. He had announced his mysterious box well in advance as something that would be distortionless by any generally accepted audio test, but would cause music or speech to be absolutely unlistenable when connected into an actual program chain.

With earnest energy and that wisp of a smile worn by those who know there's a treat coming, he hooked up his box to various kinds of instruments, like a magician opening his empty hands to his audience. The box produced no evident distortion. Then he hooked it up to a radio, and the speech and music were spastically interrupted as though someone were randomly wiggling an on-off switch. He was right: speech was unintelligible and music almost unrecognizable. A very simple circuit in the box was responding to the rapid and asymmetrical voltage changes that represent speech and music, and triggering a relay whose contacts short-circuited the audio line, cutting off the audio program in an irregular, unpredictable way.

Some people at the workshop resolutely refused to get it. Most people were amused, some were delighted, but there was a significant sentiment that Heyser's cute parlor trick didn't belong in a serious distortion workshop. One man remarked that he couldn't imagine any kind of real audio device that would function in that way, so what was the point of the demonstration? The point was, of course, that human perception can consider totally unacceptable something that measures not only good, but actually perfect, by all the usual tests, and therefore that the measurements, no matter how refined, are not useful. They won't—they can't—predict how a listener will react. This is not a welcome suggestion to people who have been trained to make conventional measurements and have invested possibly tens of thousands of dollars in conventional instrumentation.

The magician simile is not just a biographer's gimmick; Heyser used the word himself in a 1976 "Communication" to the Journal of the Audio Engineering Society:

"At the present state of sound reproduction technology, the audio engineer shares the professional goal of a magician. Both strive for the creation of an illusion in the mind of the observer. In audio this illusion is that of an apparent acoustic reality. The majority of listeners 'hear' almost the same illusion. An industry is based on that premise.

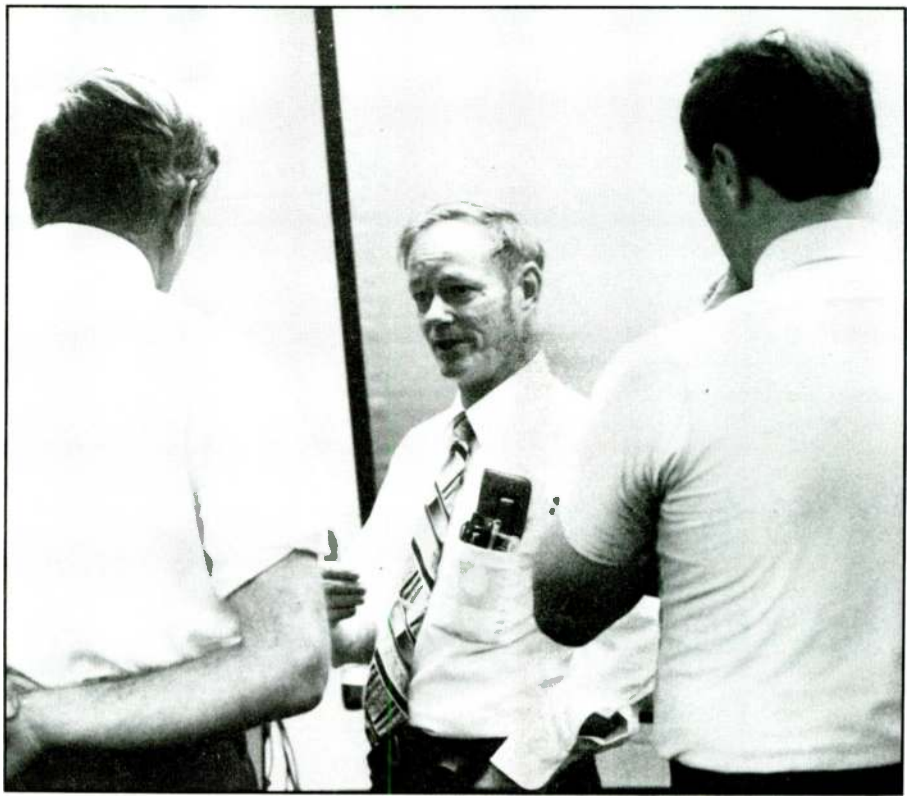
"But if we carefully measure the sound field in the listening environment, we find that the actual sound comes, as we expect, from discrete loudspeaker sources and could not have originated from the apparent space location of the illusory sources....

"If we wish to understand how to 'measure' what we 'hear,' then we must deal with subjective perception and the illusion of sound. We cannot avoid it or pretend that it does not exist."

These words are from "Perspectives in Audio Analysis: Changing the Frame of Reference, Part 1", published in *JAES* for October 1976. It is not usually easy, nor perhaps useful, to mark sharp divisions in the development of a person's thinking, yet this paper suggests an excursion into a new region of thought that was barely hinted at by his paper the previous year titled "The Geometry of Sound Perception." Still, as far back as the 1967 paper relating time-delay spectrometry to acoustical measurements, a major component of Heyser's work involved "transforms" or alternative frames of reference: typically, time and frequency, which resulted in his teaching the audio industry to pay attention to the time domain as well as the more common amplitude-versus-frequency measurements. But geometry? Frames of reference? In audio? Many readers no doubt flipped past the item back into more comfortable terrain.

In "The Geometry of Sound Perception," Heyser argues that if a number of critical listeners independently hear a defect in audio reproduction and describe what they hear, and if the defect is not revealed by analytic measurements, then it is wrong to assume that the listeners are mistaken (as technical people are all too likely to do), or that the analysis is defective. The truth is most probably that we are not using our mathematical tools properly. Pointing out that "it takes a number of independent attributes to describe a sound," he challenges readers, "If you don't believe it, try writing a musical score as a frequency spectrum." In other words, a time domain and a frequency domain aren't enough.

"There is a branch of mathematics which handles form and texture and multiple-dimensionality. Furthermore... our present time-domain and frequency domain mathematics can be considered



Test measurements are useless unless the listener's assessment is taken as the starting point.

to be a special twig on that branch.... It is geometry, [which addresses] the shape and size of things, or [more precisely], the study of invariant properties under specific groups of transformations. In addition, we will sometimes need to use a special form of geometry called topology—the study of things that don't change under plastic deformation of the coordinate basis."

Heyser refers the reader to a previous paper (1973) which "introduced concepts of topology into *linear* audio analysis [emphasis in the original]" and demonstrated that it is possible to describe subjective impressions of form with math models, that present models of time and frequency domain "are correct and sufficient [for linear analysis], but they are a special case of the more general geometry." With his occasional leavening wryness, he added, "...much of the apparent disagreement between objective and subjective views springs from the same thing that caused construction difficulties on the Tower of Babel—namely, they don't speak the same language. Geometry gives a common language."

He continues: "However, the subject that really causes the more violent disagreements between the subjective and the objective person is that of distortion—how to measure it and how to interpret what the measurement means. Distortion is a manifestation of a *nonlinear* process. A nonlinear process or device is one in which the output is not related to the input by a simple multiplier, or constant, such as gain, but in which the output changes as a *function* of some other variable—such as time, or the input voltage, for example; as a result, the curve relating output to input is just that, a curve not a straight line, hence the term "nonlinear."

Heyser's geometrical construct for the entire perceptual domain eventually addresses even this.

Changing frames of reference is not as unfamiliar as one might think at first, although the phrase may seem remote. Anyone who has ever studied plan and elevation views of an object or a structure is aware that the two views are necessary (and maybe sufficient, maybe not), and complementary. They each show different views (frames of reference!) of the same object; an 8-foot 2-by-4 in the elevation view will show up as a little dotted rectangle, perhaps, in the plan view. An example more germane to audio is another pairing: a conventional frequency vs. amplitude plot of a microphone's output, on rectangular coordinates, contrasted with a polar plot showing angle of incidence vs. amplitude, done on polar (circular) coordinates. Either graph could be made to convey all the information by having the draftsman and the reader agree to certain conventions.

In the rectangular plot, one might show a family of curves each indicating the microphone's frequency response at a different angle of incidence, starting with zero degrees (on axis) at the top of the graph, then doing another at, say, 30 degrees, then 60 degrees, and so on. That's one way of looking at that microphone's amplitude response with respect to frequency *and* angle of incidence. But another—and equally valid—way to give the same information is to do a polar plot of the microphone at, say, 1kHz, and at as many other frequencies as seems desirable. One form may be preferred over the other because of its superior ability to convey certain information. In other words, you may see and comprehend something better in one way than in the other way. To put it even more strongly, something crucial may be revealed in one frame of reference that was hidden in the other.

Here is a key to the perplexing question of how a multidimensional experience such as that of listening to music (Heyser considered five dimensions to be appropriate) can be represented by the essentially one-dimensional phenomenon of a voltage (or current) varying with time—which is all that an audio signal is in the frame of reference provided by, say, an oscilloscope—regardless of who's singing or playing what kind of music in which kind of space. Yet in that voltage-varying-with-time are encoded the pitch, loudness, timbre, attack, decay, and location of the sound we are hearing—even, if you care to pursue it further, the identity of the music we are hearing, its composer and performer. Heyser enjoyed making people blink in confusion by asking "What kind of instrument could B&K make that would tell you whether a piece was by Brahms or by Chopin?"

I trust that readers—especially ones highly trained in mathematics and physics—will not judge Heyser's work based on this brief tour by a writer who is neither a mathematician nor a physicist. I ask readers to attribute errors and lack of clarity to me and not to Richard Heyser. Those who wish to read what he himself wrote, complete with mathematical formulations and derivations, are invited to consult his papers in *JAES*. A list of those is available from the Audio Engineering Society, 60 East 42nd Street, New York, NY 10165.

On a more popular level, his writings for *Audio* are illuminating, especially the two articles in the March and April 1979 issues on catastrophe theory, which makes mathematical sense of such seemingly vague matters as why your stereo may not sound as good to you after you've heard a live concert. The highly readable February 1979 article, "A View Through Different Windows," explores the issues of dimensionality in a most persuasive and stimulating way, building on E. A. Abbott's famous 1884 work, "Flatland, A Romance of Many Dimensions."

Articles in the September, November and December 1974 issues of *Audio* lay

What kind of instrument could B&K make that would tell you whether a piece was by Brahms or Chopin?

out in meticulous detail the philosophical and practical foundation for the famous loudspeaker reviews Heyser did for *Audio* for many years. These thoughtful pieces continue into 1975 with a January article on room testing and a May article on loudspeaker polar response. And the reviews themselves are "must" reading for anyone who wants to discover the practical, predictive application of Heyser's new kinds of measurement, even when the loudspeakers themselves may not be particularly good or interesting.

Heyser's body of work has profound implications, not only in audio. Any mathematically sturdy work that bridges the seemingly fundamental and irreconcilable gulf between the objective and subjective realms is likely to offer insights as well in physics or sociology. It is tantalizing to contemplate a mathematical opus that might, after centuries, heal the rift between "the two cultures", to borrow C. P. Snow's famous phrase. And there is more: a "Heyser transform" (not his name for it), of which the more widely understood Fourier and Hilbert transforms are special cases. But at this writing, no one I have talked with can guess who might take up Richard Heyser's work where he left it on March 14, 1987. May he rest in peace.

This is the second part of a two-part article.

[1]Bracketed interpolations in this quoted material are clarifications based on both sides of the conversation it comes from.

MOBILE SPEAKER-ONE

BY JAMES J. SAWCHUK

For the outdoor season, I've always wanted to have a pair of speakers that would have good dynamic range, be mobile and rugged, and able to withstand varying temperatures and climate conditions. I decided to build a simple, closed-box design. Exterior dimensions are 11″ high by 8″ wide by 9″ deep. My drivers are Pyle K-P6940Ds, which have the following specifications:

- 6″ by 9″ coaxial speaker;
- Polypropylene woofer with Polyfoam surround (resists humidity, moisture and extreme temperatures);
- Low-profile, wide dispersion 1″ dome tweeter;
- 40 oz. magnet;
- 8 lb. high-energy power motor structure;
- Polyswitch, which protects the tweeter from power overload—resets automatically;
- 12dB/octave crossover;
- High temperature, polythermal 2″ Kapton voice coil;
- Power-handling: 100W RMS, 200W peak;
- Sensitivity (1W @ 1M): 94dB SPL
- Frequency response: 40Hz–20kHz;
- Impedance: 4Ω.

PLYWOOD CABINETS. The cabinets are constructed from ¾″ outdoor (marine) plywood, glued and stapled at each edge. Using the templates provided with the speakers, I cut out the 6″ by 9″ hole on the front for the speaker, and the 2″ hole on the back for the speaker terminal housing (purchased from McGee Radio & Electronics).

I covered the cabinets with a white (plastic laminate) formica, because white best reflects the sun's rays and because it gives them a "high-tech" look. Protect the cabinets by spraying on Armor-All and wiping-off the excess. Before I mounted the speakers and terminal housings, I installed a metal kitchen cabinet handle (lightly sanded and spray-painted with a few coats of satin-white Rust-Oleum) on the top of the cabinet.

I also installed three teenut fasteners to the bottom of the cabinet (*Fig. 1*), so I could attach and detach the Atlas female microphone stand adaptor, also purchased from McGee. I wanted to use mike stands as speaker stands for indoor and outdoor applications. With the female mike stand adaptor removed, I can reinstall the screws, with rubber washers, to retain the closed-box design.

To make sure the cabinets remained airtight, I put a bead of silicone on all internal joints, (including where I attached the handles) and around the three teenut fasteners. Next, I lined the cabinets with ½″ polyester filling and stapled it in place to prevent shifting.

I soldered a 16″ length of Monster Cable to the terminal housing and to the speaker. I added a bead of silicone to assure an airtight fit, then screwed the speakers and terminal housings into the cabinets.

Finishing touches included using dif-

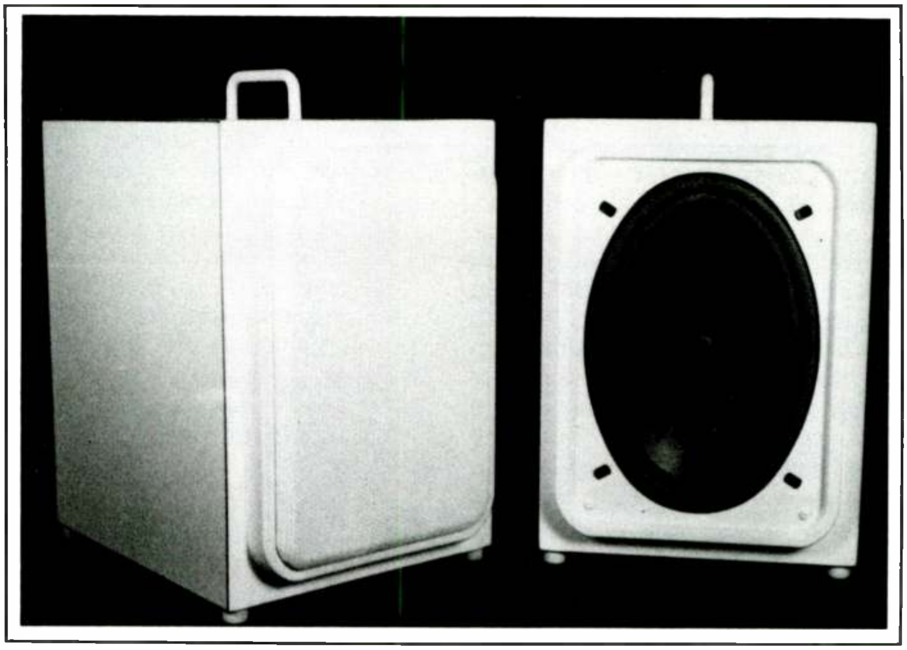


PHOTO 1: The "ultimate," mobile, outdoor speaker system.

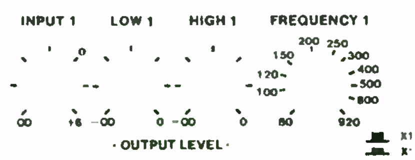
Electronic Crossovers 24CX-2 & 24CX-4

- 4th order State Variable Linkwitz-Riley Filter design. All outputs in phase at crossover point.
- 24dB per octave crossover slopes for greater driver protection than with 12db and 18db per octave types.
- Flat summed electrical response throughout the crossover region.
- Zero lobing error (polar pattern tilt) throughout the crossover region.
- Automatic 3-way operation on Models 24CX-2 and 24CX-4 or independent two or four channel 2-way operation.
- Integral security cover.
- UL recognized.
- Polypropylene capacitors are used in the signal path, active filter section, and even the power supply.
- The input/output connections used are high reliability GOLD phono plug connectors.

24CX-2/4 electronic crossovers represent the latest in state-of-the-art design and manufacturing technology. Innovative electrical and mechanical design concepts and implementation have resulted in a product of superior performance expected of units selling up to several times the 24CX-2/4 price.

A 4th order Linkwitz-Riley type of crossover does **not** guarantee flat summed response. Flat summed response of the low and high frequency outputs of the 4th order Linkwitz-Riley crossover is made possible by using precision 1% metal oxide film resistors, selected capacitors, and the industry's only precision matched and selected 1% four-gang Frequency Range potentiometer. No matter where the control is set, you will get flat summed response, not just the end positions where some products are specified.

High Slew-Rate, Low Noise, Bi FET operational amplifiers guarantee the finest in sound quality and overall electrical performance.



SPECIFICATIONS

Crossover Filter Type
4th Order Variable Linkwitz-Riley design, 24 db/octave slopes.

Frequency Range
x1 80 Hz to 920 Hz
x10 800 Hz to 9.2 kHz

Slew Rate
$R_L = 2$ kohms 12 v/usec.
$C_L = 0.01$ uFd +/-2.5 v/usec.

Total Harmonic Distortion
R_L 2 k ohms
Low Freq Output .01% THD
20-9 kHz @ +8 dBu (1.95 volts)
High Freq Output .02% THD
80 - 20 KHz @ +8 dBu (1.95 volts)
Maximum Output Level R_L 2kohms
+18 dBu (6.2 volts)
@ .05% THD 20 - 20 kHz
Maximum Output Current
25mA peak @ 25°C
Maximum Voltage Gain +6 dB
Hum and Noise (20 Hz - 20 Khz)
Av = 0 dB f_c = 800 Hz

Low Frequency Section
a. Output Attenuater
@ -infinity-100 dBu
b. Output Attenuator
@ 0 dB -92 dBu

High Frequency Section
a. Output Attenuator
@ - infinity 100 dBu
b. Output Attenuator
@ 0 dB 84 dBu

Signal to Noise Ratio 102 dB
Input Impedance
Noninverting Unbalanced 20 kohms
Output Impedance 300 ohms
Controls
Input Level Continuously variable from +6 dB gain to 80 dB attenuation
Output Level Continuously variable from 0 dB (unity) gain to 86 dB attenuation
Crossover Frequency Adjustable from 80 Hz to 900 Hz on the X1 range and 800 Hz to 9 kHz on the X10 range
Power Source U.S. & Canadian Models
120 v AC, 60 Hz
Power Consumption 12 VA
Safety Agency Certif.UL Listed
Dimensions (W X H X D) 19 inches x 1.75 inches x 5.5 inches with security cover
Net Weight
24CX-2 3.5 pounds
24CX-4 4 pounds
0 dBu = 0.775 v rms

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Model 24CX-2 $325.00
Model 24CX-4 $425.00

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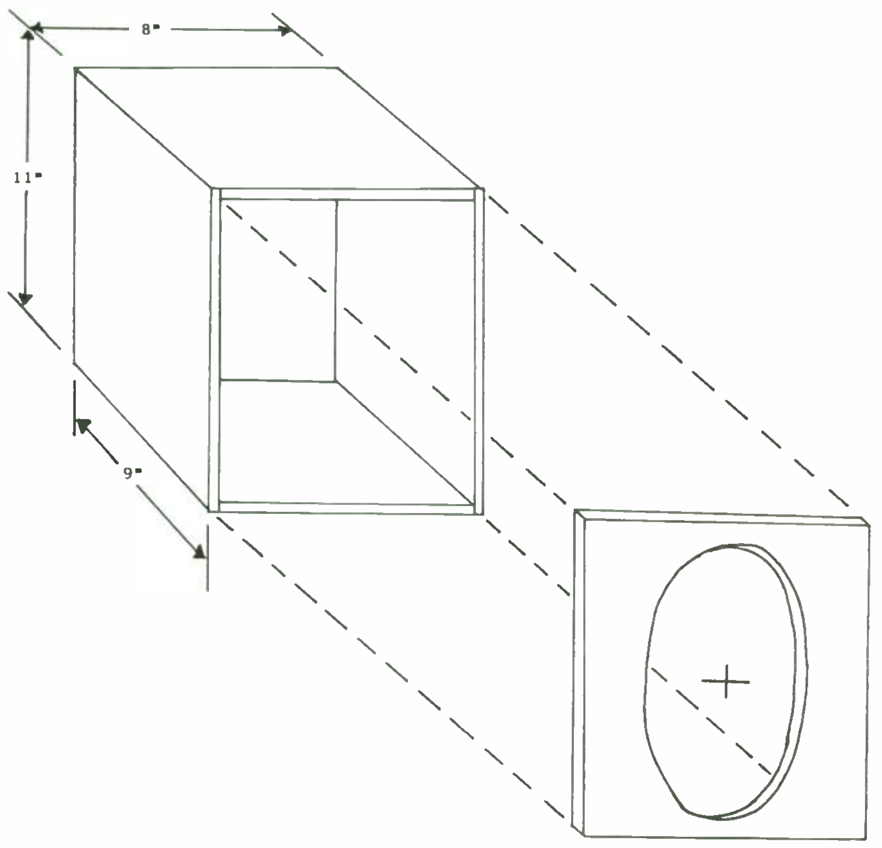

FIGURE 2: Cutout guide.

ferent grilles, and soft-rubber feet on the cabinets. I bought a pair of metal, G-69 Ultimate grilles from Crutchfield, spray-painted them satin-white to match the handles, covered them with a white, lycra-spandex cloth that is both durable and acoustically transparent, and mounted them with four small, wood screws.

With everything completed, I tested my speakers on my 90W/channel Yamaha system. They looked good and sounded great. Easily handling all the power I could put in, they displayed very good imaging and detail, and tremendous bass response.

Because they are efficient, (94dB, 1W @ 1M), you can even use a small, portable "walkman-type" radio/tape player.

Since each finished speaker weighs about 18 pounds, I'm glad I added the metal carrying handles. To protect the speakers when they're being moved, I had a local upholstery shop sew me a pair of custom-fitted, light grey-colored Naugahyde covers.

The total construction cost, including the covers, was about $180.

I have decided to design and build a matching, portable hi-fi unit to power the MS-1s. It will be based on a car hi-fi cassette deck/tuner, power amp and nicad battery, which I hope will give me the "ultimate" custom-built mobile system.

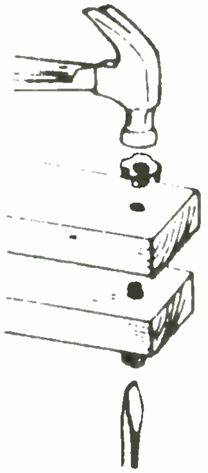

FIGURE 1:

1. Drill hole to fit barrel of Teenuts 10-24, use ¼" drill bit; ¼-20, use 5/16" drill bit; 5/16-18, use 7/16" drill bit; ⅜-16, use ½" drill bit
2. Insert Teenut into hole and hammer into fixed position (see illustration)
3. Attach bolt from opposite end and tighten

ABOUT THE AUTHOR

James J. Sawchuk is an independent, free-lance, television producer/director. He is president of his own production company, High Performance Video, located in Langhorne, PA. He currently resides in Bucks County, PA and has been working in the audio/video industry for the past 10 years. His musical tastes vary from rock and jazz to classical.

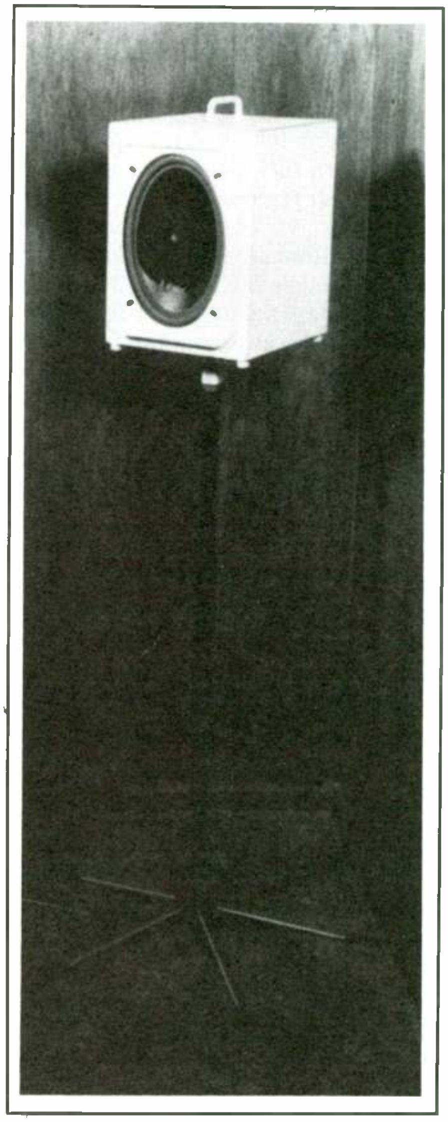

PHOTO 2: Make stands adapted for speaker use, indoors or outdoors.

MODIFYING THE POLK 10s

BY JAMES T. FRANE

In Richard J. Kaufman's article titled "Build A Passive Image Enhancer" (*Audio*, November 1986), the author discusses how to use two sets of speakers to extract more of the ambience inherent in a stereo recording. In Mr. Kaufman's setup, the second set of speakers is placed just to the outside of the primary left and right channel speakers. The first set is connected to the amplifier in the usual manner, while the outside set is wired in series across the amp's hot, or "+," terminals. The second set reproduces only the stereo difference signal (the difference between the left and right channels).

The second set's placement means the information they reproduce will be out of phase with, and cancel, interchannel crosstalk. This allows the listener to discern the recording's ambience.

I successfully experimented with Mr. Kaufman's suggested setup, so I modified my Polk Model 10 speakers to incorporate this feature in a single pair of speakers (much as Polk has done with their "SDA" series).

The Polk 10s are ideally suited to this modification. They have a pair of midrange/woofers installed side-by-side on the front baffle. These 6½-inch drivers handle the frequency spectrum from the 3kHz crossover to the 1-inch dome tweeter centered above them to the 60Hz acoustic crossover to the 10-inch passive radiator below them. Instead of using a second pair of speakers, I planned to use the outboard midrange/woofer in each cabinet for the stereo difference signal.

Many articles have been written on improving stereo imaging by cancelling interchannel crosstalk. I will address this subject briefly, then show you how I improved stereo imaging without electronic enhancements by rewiring my Polk Model 10 speakers.

Before stereo, one-speaker monaural reproduction was the norm. With the advent of stereo, however, two speakers were needed to reproduce the two channels of sound. Stereo was intended to be more than monaural from two sound sources, or speakers; left and right channels were different. Since stereo was to present a three-dimensional, or lifelike, recreation of the original music, phase and/or amplitude differences were recorded into the two channels. The actual effect was initially far short of the goal.

Although many improvements have been made over the past 30 years, only a rare combination of equipment and listening environment can make you feel as though you are at the actual performance. As good as stereo reproduction has become, an overriding problem exists: both ears hear both speakers. In a properly positioned listening environment, the sound of the left speaker will be heard first by the left ear. However, that sound will also be heard milliseconds later by the right ear. This arrival of left channel sound at the right ear is not delayed enough for the brain to discount it as a primary sound source.

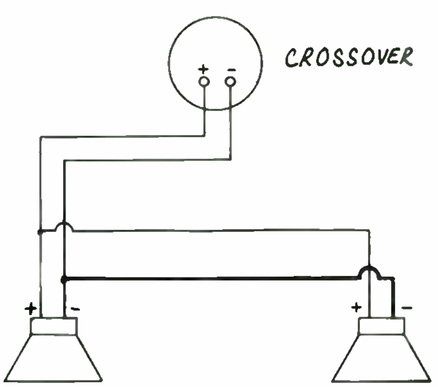

FIGURE 1: Stock wiring of the Polk 10 midrange/woofers.

At the same time the left channel sound is reaching the right ear, the opposite situation is occurring at the left ear with sound from the right channel speaker. Most of the ambience becomes lost. The image may, in a good setup, extend from one speaker to the other, but sounds seldom come from an area outside the space between the speakers.

Joel Cohen of Sound Concepts, Bob Carver, and others have used electronics to address this situation, but Polk Audio has designed their SDA speakers to cancel interchannel crosstalk by a second set of drivers within the same enclosure. According to their literature, an interchannel crosstalk cancelling signal comes from the outboard set of drivers in each enclosure. By spacing the drivers so that the sound path from the outboard set of drivers is about six inches longer to the listener's ear than the sound path from the inboard set of drivers, the cancellation signal's delay is properly timed. Polk's method may be more complex, but after I experimented with separate sets of speakers, and made a phone call to the Polk factory, I decided to modify my Model 10 speakers to attempt to simulate the SDA's effects.

THE MODIFICATION. Be forewarned that my modifications will void your warranty. The inboard set are connected for normal stereo, that is, left and right channels, respectively. The outboard

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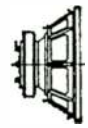
Madisound is has arranged for ample stock on MB Electronics' state of the art tweeters and midranges. They are manufactured in Germany and are used in many high end designs including the Dahlia Debra speaker.



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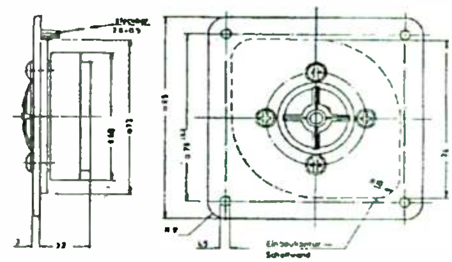
Specifications

	MCD19	MCD25	MCD51
Sensitivity db	91.00	91.00	88.00
Nominal Impedance Ω	8.00	8.00	8.00
DC Resistance Ω	5.60	6.90	6.70
Power Handling			
Slope db/Freq Hz/Watts	12/5K/100	12/2.8K/100	12/900/100
Volume Displacement mm^3	2.49	4.68	20.9
Resonant Freq. Hz	2,430	990	340
Qms	2.89	5.14	2.82
Qes	1.28	1.25	.63
Qts	0.88	1.00	.51
Voice Coil Diam. mm	18.00	25.45	50.50
Voice Coil Height mm	1.80	1.60	4.70
Magnet Diameter mm	60.00	72.00	102.00
Total Weight kg	.32	.76	1.15



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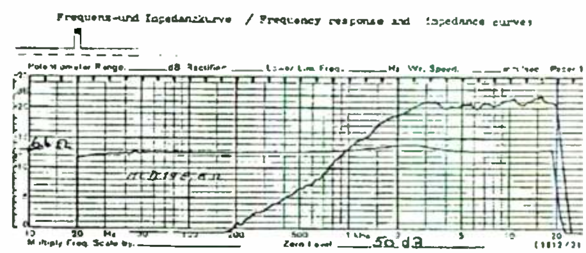

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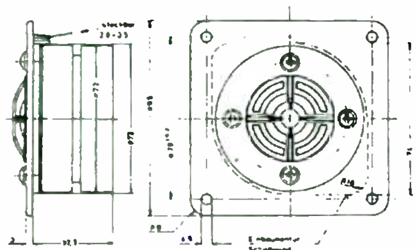
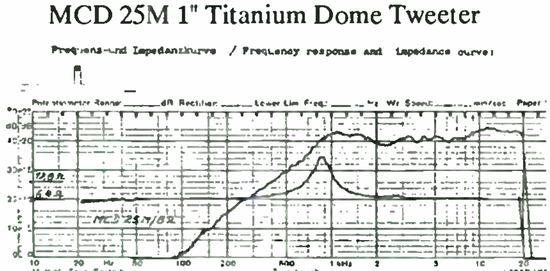


MCD 25M 1" Titanium Dome Tweeter



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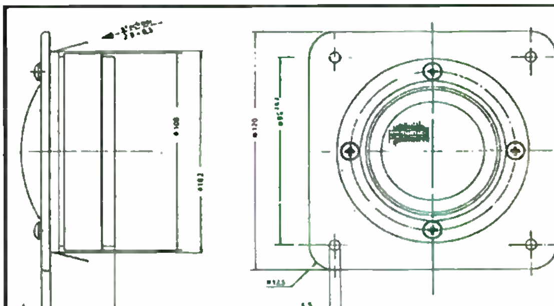
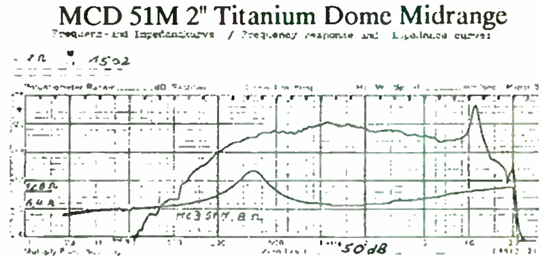


MCD 51M 2" Titanium Dome Midrange



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speakers are connected in series across the amp's left and right channel "hot" leads. (CAUTION: Some amplifiers will not tolerate this type of connection. Also, some amplifiers will not work well with the resulting reduced impedance.) The Model 10s are nominally 6Ω speakers because the midrange/woofers are wired in parallel. My modification takes the outboard midrange/woofer out of the primary circuit, theoretically changing the 10s nominal impedance to 8Ω. The outboard midrange/woofers are of nominal 8Ω impedance and are connected in series, resulting in a 16Ω load.

The two 8Ω (modified) 10s are connected in parallel with the outboard speakers. The resultant impedance is calculated by the parallel impedance formula:

$$(Z1 \times Z2) \div (Z1 + Z2) = Z$$

where Z1 and Z2 are the two impedances and Z is the resultant impedance.

Substituting,:

$$(8 \times 16) \div (8+16) = 5.33\Omega$$

Check your owner's manual to see cautions concerning minimum impedance. If you still have questions, contact the amplifier manufacturer.

The outboard speakers' wiring allows them to reproduce only the difference signal between the left and right channels. No sound will be heard from a monaural source. Except for the speaker placement, this connection is similar to the Hafler hookup for rear channel speakers.

Remove the eight screws from the Polk 10s' two midrange/woofer drivers, and pull them away from the front baffle. *Figure 1* shows the drivers' stock wiring. Cut the common, or "–," lead to each of the outboard drivers, leaving a short length of wire from each driver. After you install a wire nut or tape the portion of the common lead to each crossover network to prevent a short circuit, select a drill bit with the same outside diameter as your wire (in my case, 16-gauge zipcord). Drill a hole through each rear baffle for the outboard speakers' common leads. Strip the ends of short leads from the outboard driver common connections and solder lengths of your zipcord to these short leads (allow about two feet of zipcord).

Figure 2 shows the speakers' modified internal wiring. Insulate the soldered connections by using electrical tape, then push the other ends of the zipcord through the holes in the rear baffles. Connect banana jacks to the ends of the zipcord on the outside of the rear baffle so you can custom-cut the interconnecting cable and install banana plugs on each end. If you reposition the speakers later, it will be easy to connect a different length common cable. Seal the zipcord penetration through the baffle with silicone rubber to ensure that the enclosures remain airtight. Reinstall the drivers on the front baffle and mark the enclosure left and right. To function properly, the difference signal drivers must be on the outside. If you don't like the end result, the modifications are easily reversible.

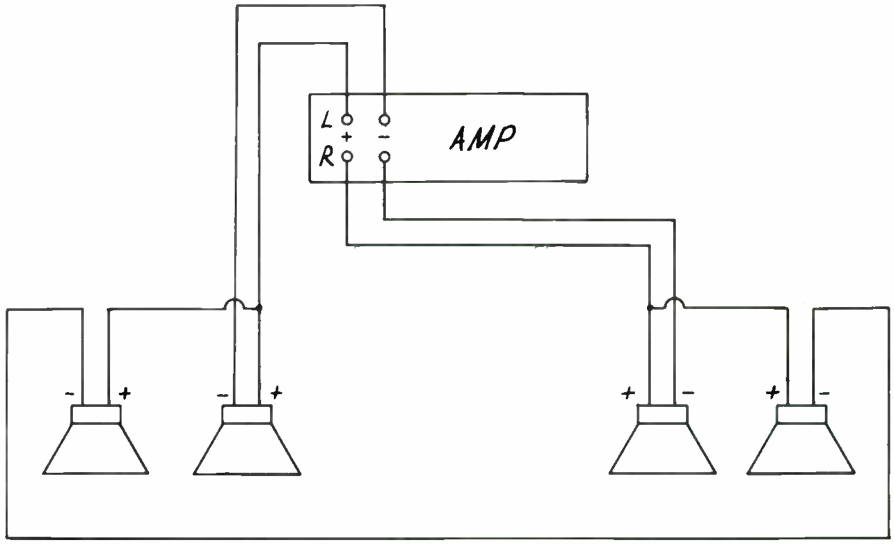


FIGURE 2: Schematic of the modified Polk 10s.

LISTENING TEST. Once I had completed the modifications, I positioned the speakers about seven feet apart, slightly toed-in toward the listening position and raised about nine inches off the floor. I connected the speaker cables to my amplifiers using the factory terminals on the backs of the baffles. In this configuration, the amplifier provides a signal to the "+" side of the inboard and outboard midrange/woofers and the tweeters, as well as to the "–" side of the inboard midrange/woofers and tweeters. I used a 16-gauge zipcord cable to connect the outboard midrange/woofers' "–" terminals together.

To audition the modified speakers, I played jazz, rock and classical music from tapes, records and FM. In general, the modifications added noticeable ambience to familiar recordings. The effects are most pronounced when your listening position is midway between the two speakers, and when the recording had a large amount of ambience (such as an orchestra in a large hall).

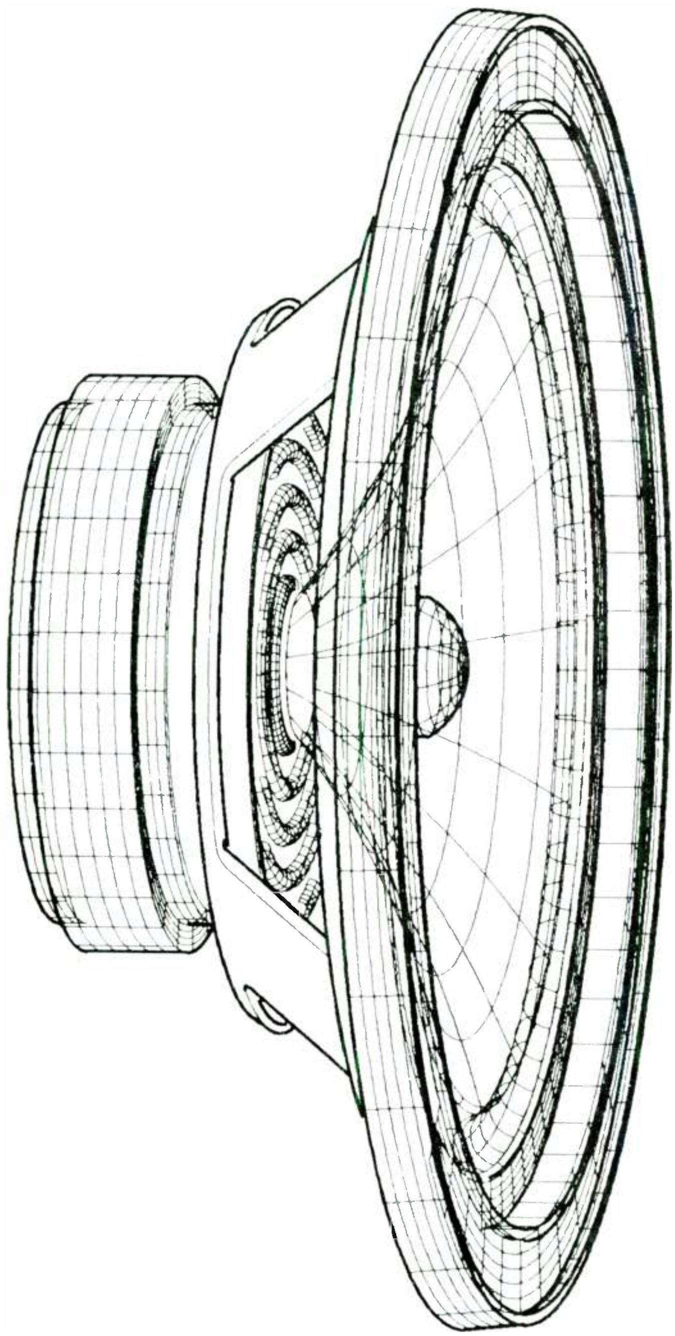
Sound sources sometimes extend outside the two speakers' boundaries. Localizing individual instruments improves from slightly to dramatically, depending upon the recording. Rather than a mass of sound sources floating between the speakers, a much wider sound stage results. Imaging, already good with these speakers, takes on added depth.

Expect some trade-offs, however, with this modification. In particular, bass seems diminished, probably because the Polk 10 midrange/woofers cross over acoustically to the passive radiator at about 60Hz. This modification causes the outboard mid/woofer to reproduce only the stereo difference signals, which diminish below 100Hz. The passive radiator is driven by mid/woofer movement. With the modification, the full frequency range of 60-3kHz is now fully reproduced by only the inboard mid/woofer.

Time will tell whether this modification is truly an improvement, or just something different. Not having a real Polk SDA series speaker set to measure and disassemble, I can't be sure about the differences between my modification and the original.

Polydax speaker corporation

Polydax speaker corporation



PROFESSIONAL RANGE

MEASUREMENT CHART

METHOD OF MEASUREMENT	Symbol	Unit	MEASUREMENT CONDITIONS
AMPLITUDE-FREQUENCY RESPONSE CURVE: In accordance with the measuring methods of the international IEC 268-5 standard. BRUEL & KJAER test Bench. —Heterodyne Analyser 2010 —Power amplifier 2706. —RMS millivoltmeter 2425. —1/2" Measuring microphone 4133. —Level recorder 2307 The curve is traced from a 20 Hz - 20 kHz swept frequency source.			—The speaker is mounted on a IEC baffle. The measurements are made under free field condition. —The B & K type 4133 microphone is placed at a distance of 0.50 meter on axis and at 30° off axis. —The measuring voltage, held constant, is 2.83 V, corresponding to an electrical power of 1 W across an 8 ohms impedance. —The published amplitude-frequency curves are presented with the sound pressure level adjusted for a distance of 1 m. —Writing speed .... 125 mm s⁻¹ —Paper speed .... 10 mm s⁻¹
IMPEDANCE CURVE: In accordance with the measuring methods of the international IEC 268-5 standard. BRUEL & KJAER test Bench. —Heterodyne Analyser 2010. —Power amplifier 2706. —RMS millivoltmeter 2425. —Level recorder 2307. The impedance curve is traced from 20 Hz to 20 kHz.			—The speaker is mounted on a IEC baffle. —It is supplied with a constant current source by means of the compressor. —The measuring current is set at 30 mA. —Measuring potentiometer .... ZR002 LINEAR —Writing speed .... 63 mm S⁻¹ —Paper speed .... 10 mm S⁻¹

DETERMINATION OF THE "THIELE-SMALL" PARAMETERS:

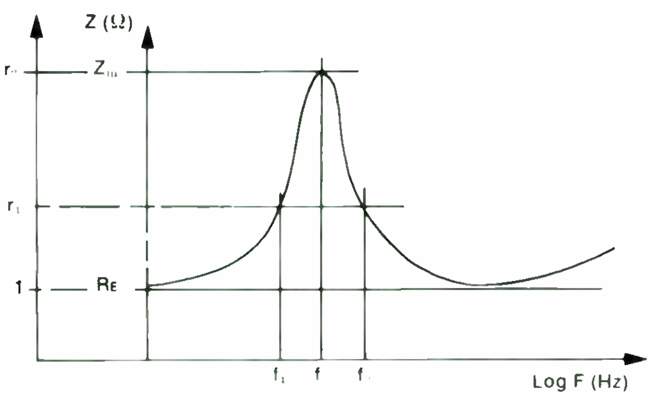


Fig. 1
Impedance curve of a Loudspeaker in the region of its resonance frequency.

IMPORTANT NOTE

The resonance frequency that determines the "THIELE-SMALL" parameters have been established to be accurate all along the life of the speaker: That means, after a complete running-in period.

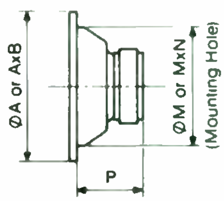
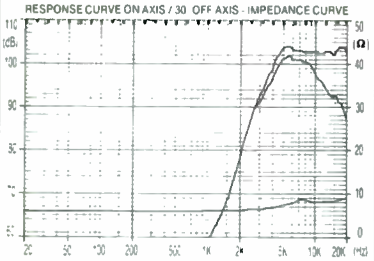
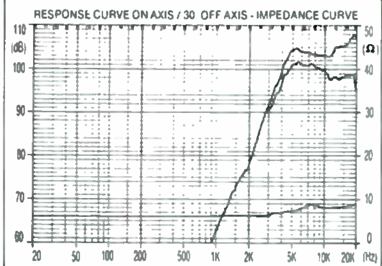
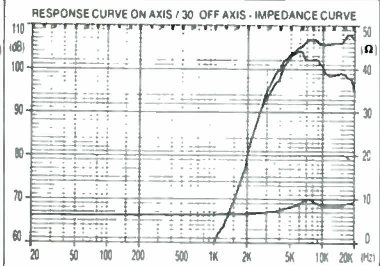
> PLEASE. PAY ATTENTION TO THE FACT THAT A NEW-WOOFER COMING FROM THE FACTORY IS NOT RUN-IN YET AND HAS A RESONANCE FREQUENCY F_s WHICH IS THEREFORE 1,13 TIMES HIGHER THAN AFTER RUNNING-IN.

—The speaker is mounted on a IEC baffle.
—Measurements are then made on the speaker after running-in and rest, when the stability of the resonance frequency has been established.
—The speaker is fed with a constant current source.
—The measuring current is set at 30 mA.
—The voltage across its terminals is measured with a BRUEL & KJAER RMS voltmeter type 2425.

METHOD OF MEASUREMENT	Symbol	Unit	MEASUREMENT CONDITIONS
a - MEASUREMENT OF THE RESONANT FREQUENCY: Sweeping of frequencies shown in Fig. 1. Noting of the frequency f_s which corresponds to the maximum value of the impedance.	f_s	Hz	—The Loudspeaker is mounted or IEC baffle. —f_s is measured using a digital frequency meter with an accuracy of one-tenth of a Hertz.
b - MEASUREMENT OF THE MECHANICAL Q FACTOR: Determination of the ratio $R_o = \frac{Zm}{R}$ Then identification of the frequencies f_1 and f_2 which corresponds to an impedance satisfying the following relationship: $Z(f_1, f_2) = \sqrt{}\ r_o \cdot R_E$ The mechanical Q factor is then given by: $Q_{MS} = \frac{f_s \sqrt{r_o}}{f_2 - f_1}$	Q_{MS}		—R_E = The D.C. Resistance of the voice coil. —R_E is measured with a D.C. current, using an impedance bridge. —The measurement of f_0, f_1 and f_2 is made with as short a time interval as possible.
c - DETERMINATION OF THE ELECTRICAL Q FACTOR: The electrical Q factor is derived from the following relation: $Q_{ES} = \frac{Q_{MS}}{r_o - 1}$	Q_{ES}		
d - DETERMINATION OF THE TOTAL Q FACTOR: The total Q factor is given by the expression: $Q_{TS} = \frac{Q_{MS} \cdot Q_{ES}}{Q_{MS} + Q_{ES}}$	Q_{TS}		*IMPORTANTE NOTE:* The values obtained for f_s, Q_{MS}, Q_{ES} and Q_{TS} relate to a speaker mounted on a IEC baffle. They may be approximated to those obtained using an infinite baffle with reasonable accuracy.

METHOD OF MEASUREMENT	Symbol	Unit	MEASUREMENT CONDITIONS
e - MEASUREMENT OF THE MOVING MASS : A known additional mass m' is added to the moving mass M_{MD}. The new resonant frequency f_s', is then found. $M_{MD} = \frac{m'}{\left(\frac{f_s}{f_s'}\right)^2 - 1}$	M_{MD}	kg	*NOTE :* M_{MD} incorporates the mass of air loading both sides of the speaker diaphragm I.E. as mounted on an infinite baffle.
f - DETERMINATION OF THE SUSPENSION COMPLIANCE : C_{MS} is obtained from f_s and M_{MD} by the relationship: $C_{MS} = \frac{1}{4\pi^2 f_s^2 M_{MD}}$	C_{MS}	mN^{-1}	
CALCULATION OF V_{AS} The equivalent C_{AS} air load calculated from the following relationship: $V_{AS} = C_{MS} . S_D^2 . \gamma . P_0$ In which $\gamma . P_0$ is a constant depending on the thermodynamic condition of the speaker air load. For adiabatic condition at the sea-level pressure: $\gamma . P_0 = 1,4 . 10^5$	V_{AS}	m^3	
MEASUREMENT OF THE INDUCTION AND FLUX IN THE GAP : Exploration of the gap using a coil with a known wire length L. The force due a current I applied to this coil is then balanced with a balance. In this case : $B = \frac{mg}{L I}$ were m is the mass required. The flux is then obtained from a knowledge of the $\Phi = B . S_E$: where $S_E = \pi d \times H_E$.	B Φ	T Wb	—The magnet system alone is tested. —The exploration coil is shorter than the gap. *NOTE :* The magnetic induction derived by this method is that which is obtained in the gap.
DETERMINATION OF THE BL PRODUCT Obtained by the "balance" method. A known current I is applied to the moving coil, which results in a displacement. This displacement is then cancelled out by balancing. This gives : $BL = \frac{mg}{I}$	BL	NA^{-1}	—The axis of the speaker is directed vertically. —The measuring current is a few tenths of an Ampere. — g : gravitational constant = $9.81\ m\ s^{-2}$; —m : mass required to achieve balance.
MEASUREMENT OF THE INDUCTION OF THE MOVING COIL The measurement is carried out at 1 KHz using a GENRAD 1657 measuring bridge.	L_{BM}	µH	
DETERMINATION OF LINEAR EXCURSION CAPABILITY Linear excursion capability X_{MAX} is determined by the Formula: $X_{MAX} = \frac{H_E - (\text{Voice coile height})}{2}$	X_{MAX}	mm	According to JAES Vol. 29 - 1/2 - 1981 "Moving coil Loudspeaker topology as an indicator of linear excursion capability" by Mark R. Gander.
DETERMINATION OR RATED POWER HANDLING the rated Power Handling Capacity is determined by power life test, according to DIN 45573 sheet 2 standard.	P	W	—The Bass and Mid-ranges under test must be loaded by a suitable cabinet. —The Mid-ranges and Tweeters under test are connected with the recommended second order high-pass filter.
CARACTERISTIC EFFICIENCY LEVEL : In accordance with the measuring methods of the international IEC 268-5 standard. Characteristic sound pressure level in dB SPL referred to 0 dB SPL = $2 \times 10 . 10^{-5}$ Pa, is obtained at 1 metre on reference axis of the speaker, under free field condition for an electrical power input of 1 W.	η	dB	—The speaker is mounted on a IEC baffle. The measurement is carried out under free-field condition. —The microphone is at 0.50 meter on reference axis of the loudspeaker. The measurement is adjusted so as to give values for a distance of 1 meter. —The voltage at the speaker terminals is adjusted in such a way as to obtain a power of 1 W relative to the rated impedance of the loudspeaker. —The value is given for the useful frequency range of the loudspeaker.

Polydax speaker corporation

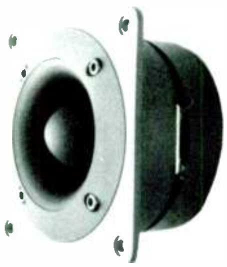
			
ØA or AxB/ØM or MxN/P (mm)	◫ 95 / Ø 74 / 55,7	◫ 95 / Ø 74 / 55,7	Ø 128 / Ø 102,5 / 52,5
Fixing holes N ØT on Ø or ◫ (mm)	4 Ø 5,2 on Ø 95,5	4 Ø 5,2 on Ø 95,5	4 Ø 5,1 on Ø 116,6
Type	**PR 110**	**PR 120**	**PR 130**
Specification number	SP 60 001	SP 60 002	SP 60 003
Designed for	ALMG3 diaphragm — Ferrofluid cooled — high efficiency Tweeter	Titanium diaphragm — Ferrofluid cooled — high efficiency Tweeter	Titanium diaphragm — Ferrofluid cooled very high efficiency Tweeter
Response curve on axis and 30° off axis Impedance curve			

Parameter	Unit	Symbol	PR 110	PR 120	PR 130
Voice coil : Type / Diameter (1)	mm		2 CA 2,2/20	2 CA 2,2/20	2 CA 2,2/20
Nominal impedance	Ω	Z	8	8	8
Minimum Z modulus	Ω	Z_{min}	7	7	8
Dc Resistance	Ω	R_E	6,4	6,4	6,4
Voice coil inductance	µH	L_{BM}	60	60	60
Resonance frequency (2)	Hz	f_s	—	—	—
Mechanical suspension compliance	mN^{-1}	C_{MS}	—	—	—
Mechanical Q factor	1	Q_{MS}	—	—	—
Electrical Q factor	1	Q_{ES}	—	—	—
Total Q factor	1	Q_{TS}	—	—	—
Moving mass	kg	M_{MD}	$0{,}230 \cdot 10^{-4}$	$0{,}230 \cdot 10^{-3}$	$0{,}230 \cdot 10^{-3}$
Effective piston area	m²	S_D	—	—	—
Equivalent C_{AS} air load	m³	V_{AS}	—	—	—
Magnetic induction in the gap	T	B	1,57	1,57	1,87
Magnetic flux in the gap	mWb	Ø	0,193	0,193	0,230
Linear excursion capability	mm	X_{MAX}	—	—	—
BL product	NA^{-1}	BL	—	—	—
Gap height	mm	H_E	1,9	1,9	1,9
Magnet diameter x height	mm	DxH	72 x 15	72 x 15	102 x 18
Rated power handling (3)	W	P	15	15	15
Characteristic efficiency level (4)	dB	η	103	104	106
Speaker mass	kg	m	0,800	0,800	1,685

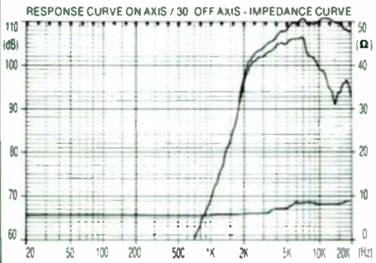
(1) Voice coil coding system

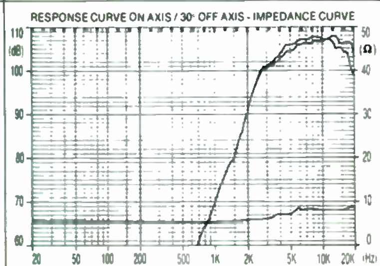
2 C A 12/25
- 12 — Voice coil diameter (mm)
- 25 — Voice coil height (mm)
- A = Aluminium former, N = NOMEX® former, K = KAPTON® former, P = Paper former
- C = Copper wire, A = Aluminium wire
- 2 — Nb. of layers

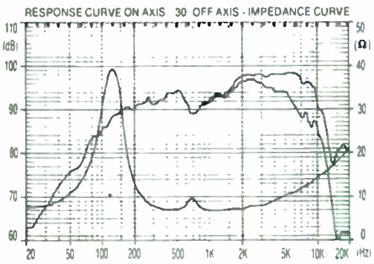
PROFESSIONAL SERIES PROGRAM

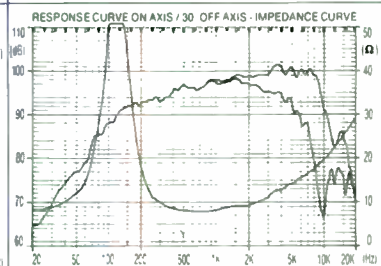


			Ø 132 / Ø 103,5 / 57,5	Ø 189 / 145 / 70,5
			4 Ø 5 on Ø 119	4 Ø 5,2 on Ø 171,6
	PR 145 B	**PR 145 D**	**PR 11 HR 40**	**PR 17 HR 60 2CN7**
	SP 60 004	SP 60 005	SP 60 006	SP 60 008
	Titanium diaphragm — high power and very high efficiency Tweeter	Titanium diaphragm high power — wide angle diffraction Tweeter	Exponential paper diaphragm, high efficiency Midrange	Flat surround diaphragm phase-plug high quality Midrange



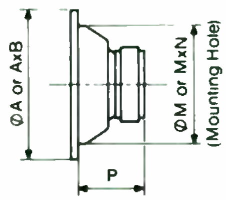
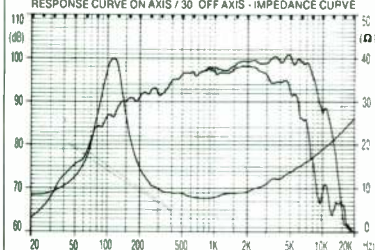
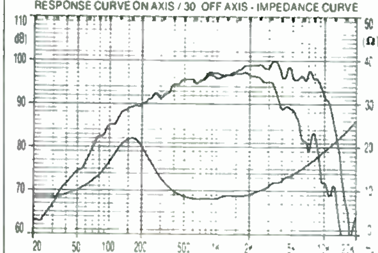
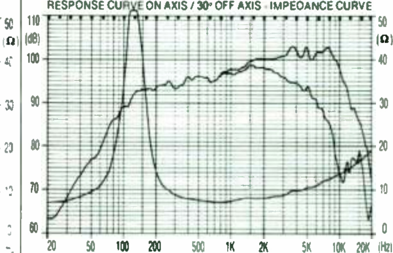






	PR 145 B	PR 145 D	PR 11 HR 40	PR 17 HR 60 2CN7
	2 AA 2,2/45	2 AA 2,2/45	2 CA 7/25	2CN7/38
Z	8	8	8	8
Z_{min}	8	8	7,0 à 1500 Hz	8,6 à 700 Hz
R_E	5,5	5,5	5,9	6,7
L_{BM}	70	70	320	520
f_s	—	—	120± 16	110 ± 16
C_{MS}	—	—	$0,53 . 10^{-3}$	$2,3 . 10^{-4}$
Q_{MS}	—	—	2,9	1,70
Q_{ES}	—	—	0,54	0,34
Q_{TS}	—	—	0,46	0,28
M_{MD}	$0,430 . 10^{-3}$	$0,430 . 10^{-3}$	$3,30 . 10^{-3}$	$9,1 . 10^{-3}$
S_D	—	—	0,00724	0,0143
V_{AS}	—	—	$3,9 . 10^{-3}$	$6,6 . 10^{-3}$
B	1,65	1,65	1,40	1,46
Ø	0,440	0,440	0,560	1,05
X_{MAX}	—	—	1	0,5
BL	—	—	5,20	12,20
H_E	1,9	1,9	5	6
DxH	110 x 20	110 x 20	100 x 18	120 x 20
P	35	35	40/1,5 KHz	60
η	110	107	97	99
m	2,400	2.600	1,250	2,300

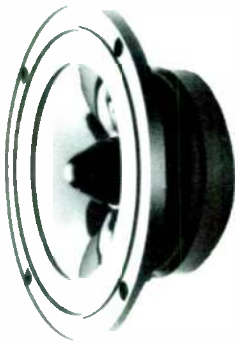
(2) Measured after Run in and Rest.
(3) In accordance with IEC Loudspeaker standard / for the specified cut-off frequency (second order high pass filter).
(4) For 1 W measured at 1 m under free field condition.

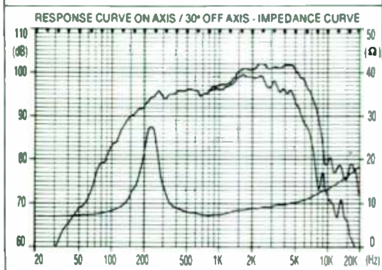
							
ØA or AxB/ØM or MxN/P (mm)			Ø 189 / Ø 145 / 70,5		Ø 189 / Ø 145 / 70,5		Ø 189 / Ø 145 / 70,5
Fixing holes N ØT on Ø or ◻ (mm)			4 Ø 5,2 on Ø 171,6		4 Ø 5,2 on Ø 171,6		4 Ø 5,2 on Ø 171,6
Type			**PR 17 HR 70 2CA7**		**PR 17 HR 90 2CA7 FF**		**PR 17 HR 100 1AK7**
Specification number			SP 60 007		SP 60 009		SP 60 010
Designed for			Flat surround diaphragm high efficiency Midrange		Flat edge diaphragm — Ferrofluid cooled high power Midrange		Flat edge diaphragm — phase plug — high efficiency and power Midrange
Response curve on axis and 30° off axis Impedance curve							
Voice coil : Type / Diameter (1)	mm		2CA7/38		2CA7/38		1 AK7/38
Nominal impedance	Ω	Z	8	Z	8	Z	8
Minimum Z modulus	Ω	Z_{min}	8,6 à 700 Hz	Z_{min}	8,0 à 700 Hz	Z_{min}	7,0 à 700 Hz
Dc Resistance	Ω	R_E	6,7	R_E	6,7	R_E	6,0
Voice coil inductance	μH	L_{BM}	520	L_{BM}	520	L_{BM}	270
Resonance frequency (2)	Hz	f_s	110 ± 16	f_s	180 ± 30	f_s	125 ± 20
Mechanical suspension compliance	mN^{-1}	C_{MS}	$2,3 . 10^{-4}$	C_{MS}	$0,86 . 10^{-4}$	C_{MS}	$2,3 . 10^{-4}$
Mechanical Q factor	1	Q_{MS}	1,70	Q_{MS}	0,71	Q_{MS}	1,94
Electrical Q factor	1	Q_{ES}	0,34	Q_{ES}	0,46	Q_{ES}	0,55
Total Q factor	1	Q_{TS}	0,28	Q_{TS}	0,28	Q_{TS}	0,43
Moving mass	kg	M_{MD}	$9,1 . 10^{-3}$	M_{MD}	$9,1 . 10^{-3}$	M_{MD}	$7,0 . 10^{-3}$
Effective piston area	m^2	S_D	0,0143	S_D	0,0143	S_D	0,0143
Equivalent C_{AS} air load	m^3	V_{AS}	$6,6 . 10^{-3}$	V_{AS}	$2,5 . 10^{-3}$	V_{AS}	$6,6 . 10^{-3}$
Magnetic induction in the gap	T	B	1,46	B	1,46	B	1,46
Magnetic flux in the gap	mWb	Ø	1,05	Ø	1,05	Ø	1,05
Linear excursion capability	mm	X_{MAX}	0,5	X_{MAX}	0,5	X_{MAX}	0,5
BL product	NA^{-1}	BL	12,20	BL	10,80	BL	10,80
Gap height	mm	H_E	6	H_E	6	H_E	6
Magnet diameter x height	mm	DxH	120 × 20	DxH	120 × 20	DxH	120 × 20
Rated power handling (3)	W	P	70	P	90	P	100
Characteristic efficiency level (4)	dB	η	99	η	98	η	101
Speaker mass	kg	m	2,300	m	2,300	m	2,300

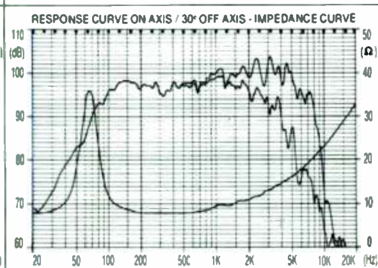
(1) Voice coil coding system

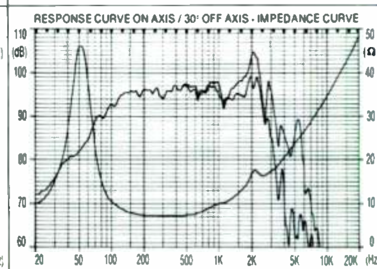
2 C A 12/25
- 12 — Voice coil diameter (mm)
- 25 — Voice coil height (mm)
- A = Aluminium former
- N = NOMEX[R] former
- K = KAPTON[R] former
- P = Paper former
- C = Copper wire
- A = Aluminium wire
- Nb. of layers

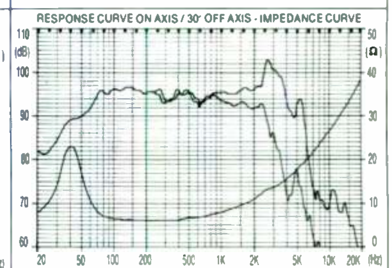
PROFESSIONAL SERIES PROGRAM

			
Ø 189 / Ø 145 / 70,5	Ø 277,5 / Ø 227 / 96	Ø 277,5 / Ø 227 / 93	Ø 322 / Ø 283 / 121
4 Ø 5,2 on Ø 171,6	6 Ø 5,5 on Ø 257	6 Ø 5,5 on Ø 257	4 Ø 6,2 on Ø 305
PR 17 TX 100 1AK7	**PR 24 LXT 100**	**PR 24 ST 150**	**PR 30 ST 100**
SP 60 011	SP 60 024	SP 60 025	SP 60 012
TPX diaphragm — phase plug — very high quality and efficiency Midrange	Very high quality and efficiency Bass-Midrange unit	High power and high efficiency Bass-Midrange unit	Good price-quality ratio — extended Bass-woofer





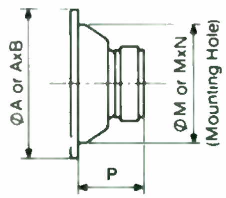


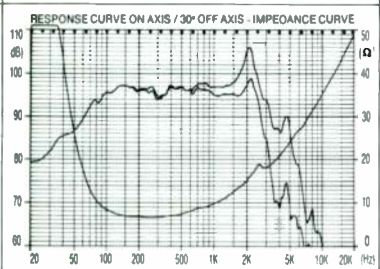
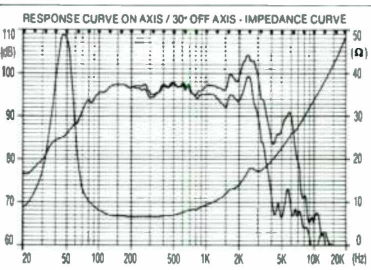
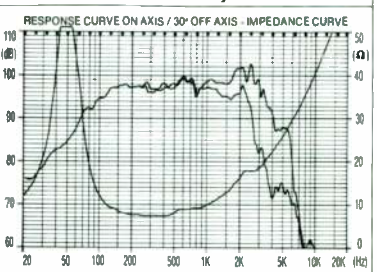


	1AK7/38	2CA12/38	2CA15/67	2CA15/46
Z	8	8	8	8
Z_{min}	70 à 700 Hz	8,0 à 400 Hz	7,3 à 250 Hz	6,2 à 150 Hz
R_E	6,0	7,1	5,8	5,6
L_{BM}	270	800	1280	840
f_s	210 ± 30	63 ± 7	49 ± 6	40 ± 5
C_{MS}	$0{,}74 \cdot 10^{-4}$	$0{,}28 \cdot 10^{-3}$	$0{,}268 \cdot 10^{-3}$	$0{,}35 \cdot 10^{-3}$
Q_{MS}	3,9	2,6	2,8	2,13
Q_{ES}	1,02	0,58	0,29	0,69
Q_{TS}	0,81	0,47	0,26	0,52
M_{MD}	$7{,}74 \cdot 10^{-3}$	$22{,}5 \cdot 10^{-3}$	$39{,}3 \cdot 10^{-3}$	$45 \cdot 10^{-3}$
S_D	0,0143	0,034	0,034	0,053
V_{AS}	$2{,}1 \cdot 10^{-3}$	0,046	0,043	0,138
B	1,46	1,35	1,28	1,26
Ø	1,05	0,970	2,15	1,10
X_{MAX}	0,5	3,0	3,5	4,5
BL	10,80	10,0	15,6	11,40
H_E	6	6	8	6
DxH	120 × 20	120 × 20	180 × 20	120 × 20
P	100	100	150	100
η	101	98	96	95
m	2,300	2,450	6	2,900

(2) Measured after Run in and Rest.
(3) In accordance with IEC Loudspeaker standard / for the specified cut-off frequency (second order high pass filter).
(4) For 1 W measured at 1 m under free field condition.

Polydax speaker corporation



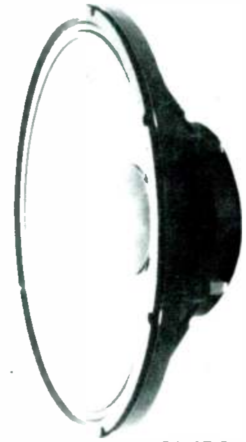
ØA or AxB/ØM or MxN/P (mm)			Ø 334,5 / Ø 284/108		Ø 334,5 / Ø 284/108		Ø 334,5 / Ø 284/111
Fixing holes N ØT on Ø or ⌀ (mm)			6 Ø 6,2 on Ø 311		6 Ø 6,2 on Ø 311		6 Ø 6,2 on Ø 311
Type			**PR 33 SM 150**		**PR 33 ST 150**		**PR 33 LXT 250**
Specification number			SP 60 014		SP 60 013		SP 60 023
Designed for			High Fidelity foam edge Woofer for Studio monitoring		Compact high power — and high efficiency woofer		Compact very high power Low-Midrange for horn loaded Bass-bins or 4 ways systems
Response curve on axis and 30° off axis Impedance curve							
Voice coil : Type / Diameter (1)	mm		2CA15/67		2CA15/67		2CN15/100
Nominal impedance	Ω	Z	8	Z	8	Z	8
Minimum Z modulus	Ω	Z_{min}	6,7 à 250 Hz	Z_{min}	6,7 à 250 Hz	Z_{min}	7,4 à 300 Hz
Dc Resistance	Ω	R_E	5,8	R_E	5,8	R_E	5,8
Voice coil inductance	µH	L_{BM}	1280	L_{BM}	1280	L_{BM}	1430
Resonance frequency (2)	Hz	f_s	24 ± 3	f_s	46 ± 5	f_s	48,6 ± 6
Mechanical suspension compliance	mN^{-1}	C_{MS}	$0{,}68 . 10^{-3}$	C_{MS}	$0{,}23 . 10^{-3}$	C_{MS}	$1{,}39 . 10^{-4}$
Mechanical Q factor	1	Q_{MS}	2,17	Q_{MS}	3,6	Q_{MS}	10,8
Electrical Q factor	1	Q_{ES}	0,23	Q_{ES}	0,36	Q_{ES}	0,32
Total Q factor	1	Q_{TS}	0,21	Q_{TS}	0,33	Q_{TS}	0,31
Moving mass	kg	M_{MD}	$65 . 10^{-3}$	M_{MD}	$52{,}2 . 10^{-3}$	M_{MD}	$77 . 10^{-3}$
Effective piston area	m^2	S_D	0,053	S_D	0,053	S_D	0,053
Equivalent C_{AS} air load	m^3	V_{AS}	0,267	V_{AS}	0,090	V_{AS}	0,055
Magnetic induction in the gap	T	B	1,28	B	1,28	B	1,50
Magnetic flux in the gap	mWb	Ø	2,15	Ø	2,15	Ø	3,30
Linear excursion capability	mm	X_{MAX}	3,5	X_{MAX}	3,5	X_{MAX}	4,0
BL product	NA^{-1}	BL	15,6	BL	15,6	BL	20,3
Gap height	mm	H_E	8	H_E	8	H_E	7
Magnet diameter x height	mm	DxH	180 × 20	DxH	180 × 20	DxH	220 × 23
Rated power handling (3)	W	P	150	P	150	P	250
Characteristic efficiency level (4)	dB	η	96	η	97	η	98
Speaker mass	kg	m	7,500	m	7,500	m	10

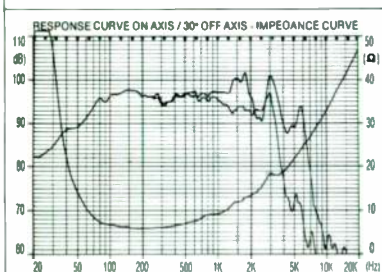
(1) Voice coil coding system

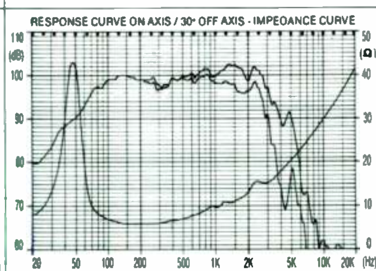
2 C A 12/25 — Voice coil diameter (mm) — Voice coil height (mm)

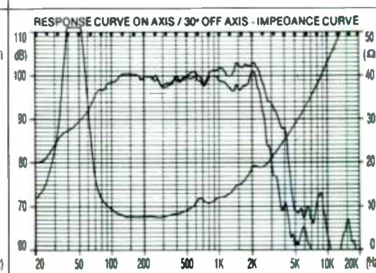
A = Aluminium former
N = NOMEX[R] former
K = KAPTON[R] former
P = Paper former

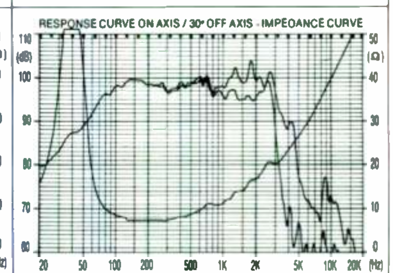
C = Copper wire
A = Aluminium wire
Nb. of layers

			
Ø 386,5 / Ø 356,5/107,5	Ø 386,5 / Ø 356,5/107,5	Ø 386,5 / Ø 356,5/110,5	Ø 386,5 / Ø 356,5/110,5
8 Ø 6,5 on Ø 372,5	8 Ø 6,5 on Ø 372,5	8 Ø 6,5 on Ø 372,5	8 Ø 6,5 on Ø 372,5
PR 38 XM 150	**PR 38 XT 150**	**PR 38 LXT 250**	**PR 38 XT 250**
SP 60 018	SP 60 019	SP 60 017	SP 60 016
High efficiency woofer for Studio monitoring	Low Bass foam edge woofer for Studio monitoring	Very high efficiency Bass-Midrange unit — Light Exponential diaphragm	High quality woofer Exponential diaphragm high efficiency and power



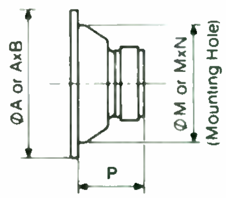






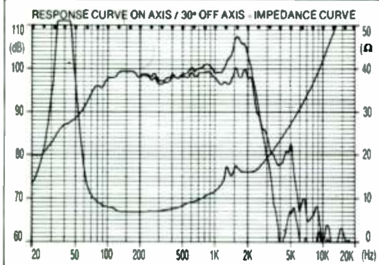
	PR 38 XM 150	PR 38 XT 150	PR 38 LXT 250	PR 38 XT 250
	2CA15/67	2CA15/67	2CN15/100	2CN15/100
Z	8	8	8	8
Z_{min}	6,6 à 200 Hz	6,6 à 200 Hz	7,0 à 250 Hz	7,0 à 250 Hz
R_E	5,8	5,8	5,8	5,8
L_{BM}	1280	1280	1430	1430
f_s	19,5 ± 3	43,6 ± 5	43,3 ± 5	38 ± 4
C_{MS}	0,64 . 10^{-3}	0,185 . 10^{-3}	1,78 . 10^{-4}	1,78 . 10^{-4}
Q_{MS}	2,70	4,0	4,7	5,3
Q_{ES}	0,30	0,47	0,29	0,33
Q_{TS}	0,27	0,42	0,28	0,31
M_{MD}	104 . 10^{-3}	72 . 10^{-3}	76,5 . 10^{-3}	98,5 . 10^{-3}
S_D	0,088	0,088	0,088	0,088
V_{AS}	0,690	0,200	0,193	0,193
B	1,28	1,28	1,50	1,50
Ø	2,15	2,15	3,30	3,30
X_{MAX}	3,5	3,5	4,0	4,0
BL	15,6	15,6	20,3	20,3
H_E	8	8	7	7
DxH	180 × 20	180 × 20	220 × 23	220 × 23
P	150	150	250	250
η	96	99	100	99
m	7,100	7,100	10	10

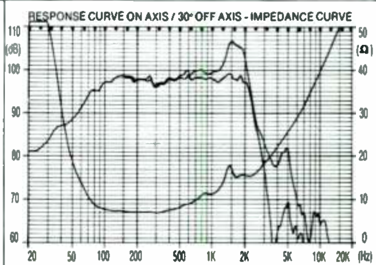
(2) Measured after Run in and Rest.
(3) In accordance with IEC Loudspeaker standard / for the specified cut-off frequency (second order high pass filter).
(4) For 1 W measured at 1 m under free field condition.

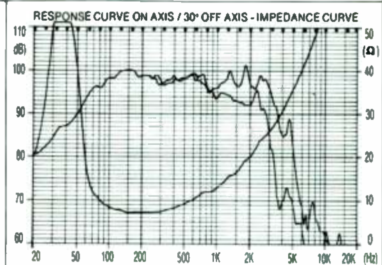


ØA or AxB/ØM or MxN/P (mm)	Ø 386,5 / Ø 356,5 / 110,5	Ø 386,5 / Ø 356,5 / 110,5	Ø 386,5 / Ø 356,5 / 110,5
Fixing holes N ØT on Ø or ◫ (mm)	8 Ø 6,5 on Ø 372,5	8 Ø 6,5 on Ø 372,5	8 Ø 6,5 on Ø 372,5
Type	PR 38 ST 250	**PR 38 SM 250**	**PR 38 XT 350**
Specification number	SP 60 015	SP 60 020	SP 60 021
Designed for	High efficiency and power woofer corruged straight shaped diaphragm	High Fidelity foam edge woofer for Studio monitoring	Very high power and high efficiency woofer

Response curve on axis and 30° off axis Impedance curve







Parameter	Unit		PR 38 ST 250		PR 38 SM 250		PR 38 XT 350
Voice coil : Type / Diameter (1)	mm		2CN15/100		2CN15/100		1CK18/100
Nominal impedance	Ω	Z	8	Z	8	Z	8
Minimum Z modulus	Ω	Z_{min}	7,0 à 250 Hz	Z_{min}	7,0 à 250 Hz	Z_{min}	7,0 à 250 Hz
Dc Resistance	Ω	R_E	5,8	R_E	5,8	R_E	5,8
Voice coil inductance	µH	L_{BM}	1430	L_{BM}	1430	L_{BM}	2210
Resonance frequency (2)	Hz	f_s	38 ± 4	f_s	19,5 ± 3	f_s	36 ± 4
Mechanical suspension compliance	mN^{-1}	C_{MS}	$1{,}78 \cdot 10^{-4}$	C_{MS}	$0{,}61 \cdot 10^{-3}$	C_{MS}	$1{,}78 \cdot 10^{-4}$
Mechanical Q factor	1	Q_{MS}	5,3	Q_{MS}	9,1	Q_{MS}	8,4
Electrical Q factor	1	Q_{ES}	0,33	Q_{ES}	0,18	Q_{ES}	0,29
Total Q factor	1	Q_{TS}	0,31	Q_{TS}	0,175	Q_{TS}	0,28
Moving mass	kg	M_{MD}	$98{,}5 \cdot 10^{-3}$	M_{MD}	$109 \cdot 10^{-3}$	M_{MD}	$110 \cdot 10^{-3}$
Effective piston area	m^2	S_D	0,088	S_D	0,088	S_D	0,088
Equivalent C_{AS} air load	m^3	V_{AS}	0,193	V_{AS}	0,660	V_{AS}	0,193
Magnetic induction in the gap	T	B	1,50	B	1,50	B	1,50
Magnetic flux in the gap	mWb	Ø	3,30	Ø	3,30	Ø	3,30
Linear excursion capability	mm	X_{MAX}	4,0	X_{MAX}	4,0	X_{MAX}	5,5
BL product	NA^{-1}	BL	20,3	BL	20,3	BL	22,4
Gap height	mm	H_E	7	H_E	7	H_E	7
Magnet diameter x height	mm	DxH	220 × 23	DxH	220 × 23	DxH	220 × 23
Rated power handling (3)	W	P	250	P	250	P	350
Characteristic efficiency level (4)	dB	η	99	η	98	η	100
Speaker mass	kg	m	10	m	10	m	10

(1) Voice coil coding system

2 C A 12/25

- 12/25: Voice coil diameter (mm) / Voice coil height (mm)
- A = Aluminium former
- N = NOMEX® former
- K = KAPTON® former
- P = Paper former
- C = Copper wire
- A = Aluminium wire
- Nb. of layers

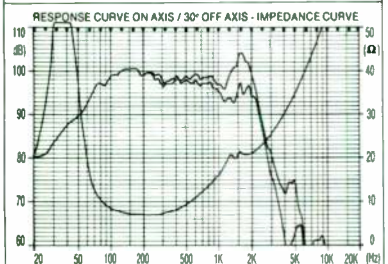
Ø 386,5 / Ø 356,5 / 110,5

8 Ø 6,5 on Ø 372,5

PR 38 ST 350

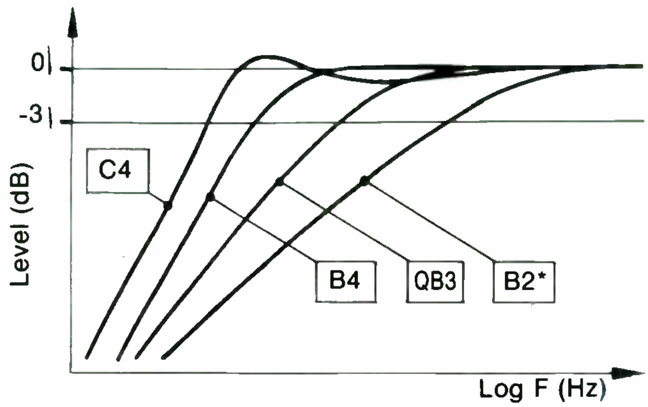
SP 60 022

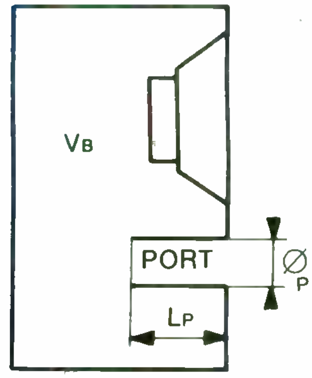
Very high power and high efficiency Woofer



1CK18/100

Z	8
Z_{min}	7,0 à 250 Hz
R_E	5,8
L_{BM}	2210
f_s	36 ± 4
C_{MS}	$1{,}78 \cdot 10^{-4}$
Q_{MS}	8,4
Q_{ES}	0,29
Q_{TS}	0,28
M_{MD}	$110 \cdot 10^{-3}$
S_D	0,088
V_{AS}	0,193
B	1,50
Ø	3,30
X_{MAX}	5,5
BL	22,4
H_E	7
DxH	220 × 23
P	350
n	100
m	10





* Sealed enclosure (Q_B = 0,707)

BASS-REFLEX ALIGNMENTS CHART FOR R_g = 0 Ω

Loudspeaker Reference	Alignment		Box tuning			Port dim.	
	Type	X	V_B	F_B	F_{-3}	Ø P	L_P
	—	(x. $V_{AS}.Q_{T2}$)	(liters)	(Hz)	(Hz)	(mm)	(mm)
PR 24 LXT 100	QB3	4	41	54	66	92	95
	B4	5,7	58	54	55	92	65
PR 24 ST 150	B4	5,7	18	71	74	92	145
	C4	7,5	23	71	68	92	115
PR 30 ST 100	QB3	2,3	85	35	48	110	210
	B4	5,7	210	35	32	150	115
PR 33 SM 150	QB3	4,5	52	45	50	92	130
	C4	7	80	45	44	130	170
PR 33 ST 150	B4	6,1	60	56	54	130	130
PR 33 LXT 250	B4	5,7	30	61	62	92	120
PR 38 XM 150	B2	—	150	—	51	—	—
	QB3	4	200	28	33	150	240
PR 38 XT 150	QB3	4	140	41	51	150	140
	B4	5,7	200	41	42	2xØ 150	140
PR 38 LXT 250	B4	5,7	82	63	63	185	140
PR 38 XT 250 / PR 38 ST 250	QB3	5	93	53	52	150	110
	B4	5,7	106	48	49	150	130
PR 38 SM 250	QB3	4	83	47	50	150	200
	C4	6,5	132	43	43	2xØ 150	250
PR 38 XT 350 / PR 38 ST 350	B4	5,7	86	53	52	185	220
	C4	7	106	48	48	185	220

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Tel: (0703) 611444 — Telex: 47497

Affiliated Agents Worldwide

catalog 300-P

Kit Report
L. A. WHITE'S NONRESONANT LOUDSPEAKER

Manufacturer: L.A. White, Jr., Inc., 1920 Medi-Park Dr., Suite 1, Amarillo, TX 79106.

From time to time it is good to step back and question some of the conventional concepts in designing loudspeakers. This is what L. A. White Jr. did when he presented his unconventional *Low-Resonance Loudspeaker* in *SB*, 4/85. In this article, he contended that the primary function of a conventional enclosure was to provide resonance to enhance the bass output. Believing that resonances in the enclosure introduce undesirable colorations in the sound, he goes on to propose a nonresonant enclosure with nonrigid sides that are not airtight. The article includes a description of a novel nonresonant crossover network for the speaker. Having refined the design, L. A. White Jr. is now offering his nonresonant loudspeaker as a kit.

When Ed Dell asked me to review the kit, I admitted that I was somewhat skeptical. The design challenged my practice of constructing an enclosure like a battleship—the sturdier the better. I never allow any leaks. I believe that if you do not contain the backwave, the output of the bass will be adversely affected. Also, an enclosure provides a load-to-control cone excursion. My engineering instincts tell me that all enclosures are resonant—you can control the resonance, but not eliminate it. And of course, the driver itself is resonant.

I realize that creating a kit is a lot of work; if LAW feels strongly enough about his design to go to the trouble of creating a kit, he deserves a review. I set aside my biases and agreed to give it a fair evaluation.

PHOTO 1: The nonrigid structure challenges conventional design enclosures.

Kit Contents

I was pleasantly surprised with the quality of the parts and materials used in the kit. Everything is first rate. The front and back panels are cabinet-grade, oak plywood connected at the four corners by solid oak 2″ × 2″ struts. They are finished with what appears to be a semi-gloss polyurethane varnish. The two sides, as well as the top and bottom are open, designed to accept the clear plastic panels which were provided.

I am told smoked or black painted panels will be available as an option for those not wishing to display the innards of the speakers. There is a large opening in the back of the enclosure that is covered by an acoustically transparent grate rather than by a plastic panel. The designer found that the opening eliminated a minor resonance. The front panels are pre-cut to accept the Dynaudio 21W5404 4Ω woofers and Dynaudio D28 8Ω tweeters which are included with the kit.

The crossover is a conventional first-order design, which surprised me since a good portion of the original article was devoted to a unique crossover design which I was looking forward to seeing firsthand. However, it seems that the original design was not compatible with the tweeter now being included in the kit. The crossover, which is fully constructed and mounted on the back panel, uses high quality components throughout. The high-pass section uses a "WonderCap® " while the low-pass section has a Madisound "Sidewinder" inductor. I am pleased to see that suppliers are starting to offer such quality components in their crossovers—it really makes a difference in the sound. Insulated wire, 16 gauge, is provided to connect the crossover to the drivers.

The kit does not include grilles because LAW feels that grilles degrade the quality of the sound of this speaker, a claim that I did not verify in my listening evaluation. The price of the kit, excluding stands, is $795. Optionally included ($75) are a pair of Mordaunt-Short speaker stands fitted with oak plywood platforms.

Constructing the Kit

Construction was straightforward, requiring no special tools or techniques. First, I installed the drivers and wired them to the crossovers. Then I mounted the metal grille over the opening in the rear of the enclosure and made sure that it was snug and did not rattle. The plastic panels for the sides, top and bottom were pre-cut, but I had to trim them a little to make them fit.

The panels are mounted with plastic shelf brackets which are inserted into holes in the frame. Double-stick foam tape was provided to secure the panels to the brackets. I used the tape but don't recommend it—it didn't last and the panels came. I think a good grade of plastic glue would be permanent. That's all there is to it; all that remained was to assemble the stands.

Listening Evaluations

When I accept a speaker kit for review, I request that it will be in the final form offered for sale. That way the review will be useful to someone considering buying the kit. I relaxed my requirements somewhat for this review because L.A. White is an amateur who believes he has an innovative design good enough to be offered as a commercial product. I wanted to give the nonresonant enclosure a fair chance; therefore, the evaluation took longer than normal due to several mid-review design changes in the crossover.

When I evaluate a speaker, first I break it in for a couple of weeks. This gives me a chance to casually listen and form a general impression before proceeding with a more formal evaluation. My first impression was that the speaker was dull and veiled. I would even say muffled. I expected better; and thought to myself with disappointment that this would be a short review.

Within a week I received two modifications to the crossover (*Fig. 1*). The first changed R2 from 34Ω to 30Ω. I wondered why R2 had been changed without C2 being affected. Sure enough, a few days later I received an accompanying change for C2, from 1μF to 1.47μF. I installed both modifications at the same time. The change was dramatic—I now had something worth reviewing. At least I had not wasted any time on detailed listening.

My general impression now was this was an easy speaker to live with: mellow and pleasant. The sound was reserved, without sparkle or deep bass. That is not to say it was without coloration. I would categorize the speaker as having a "British" sound, similar in many respects to that old stand-by, the LS3/5A. The effect achieved with classical music suggests that the designer primarily used orchestral recordings while developing the speaker. Such recordings presented a nice soundstage and a smooth, mellow balance. However, a lack of dynamics and deep bass detracted from an otherwise fine performance. Small ensembles and recordings of solo instruments or vocalists did not fare so well. They suffered most from a lack of a sense of being there, and seemed to lack definition. I also noticed this speaker didn't have the punch or dynamics needed for rock recordings. It seemed to swallow or absorb dynamic peaks.

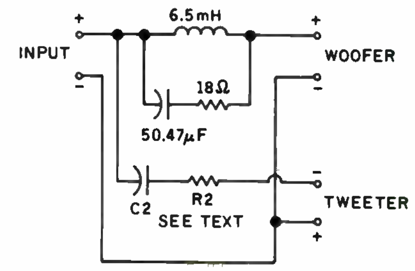


FIGURE 1: Crossover modifications.

Formal listening served to identify the factors that give the speaker its sound. To its credit, most of its faults were "sins of omission."

Not surprisingly, this speaker totally lacks lower bass. This is a conscious sacrifice made in the design of the unique enclosure. The LAW speaker in this midstage of development did not have any more or deeper bass than the LS3/5A, but it was somewhat cleaner, a result I suspect of the use of a larger woofer. Too little midrange energy gives the recessed image characteristically found in speakers exhibiting a "British" sound. Side-to-side imaging was good, but front-to-back perception of depth was poor. A deficiency in the upper midrange and lower treble imparted a dullness and lack of life to the sound. The upper treble was also weak, but because of the limitations in the lower treble, it is hard to say if the amount of upper treble in relation to the lower treble was correct.

I originally believed that the crossover should be adjusted to provide more energy to the tweeter. However, in rereading my notes, it seemed that instead of a weakness in the deep bass and also in the midrange and higher frequencies, there was a broad hump in the response from the mid-bass through the lower midrange. Interesting—LAW told me that the purpose of the RC network across the inductor in the low-pass section of the crossover in *Fig. 1* was to provide a resonant circuit which flattens the response of the woofer, giving more low bass and more midrange. Perhaps it was overdone.

The only area where I detected something definitely wrong (as opposed to something missing) was in the upper bass and lower midrange. A certain "thickness" in this area was exhibited as roughness or coarseness in a solo male voice. Because of the the speaker's general coloration, I did not attempt to pinpoint the cause.

Let me make a couple of general observations about the sound of this speaker. First, the broad hump in the response provided a euphonic sound that is easy to live with. Second, both the very high treble and the very low bass were equally attenuated. I remember that many years ago, perhaps in the late 50s or early 60s, the goal of a designer was to balance the bass and treble. There used to be a "magic number" that was the arithmetic product of the lowest bass frequency and the highest treble frequency produced. This product was to be held constant: If the bass was limited by the enclosure, then the treble was artificially rolled off to balance it. Likewise, If the bass could be extended by one octave, then the treble was extended by one octave to balance it.

Designers thought this was better than to have one extreme extended and the opposite extreme limited. The design principle today is to achieve both extended high and low frequencies with no consideration of a balance. If we fail at one extreme, there is no thought of reducing the other extreme. The results obtained by this speaker indicate that perhaps there was some merit to the concept of balance. The general result, while limited and inaccurate, provided a pleasant effect.

Well, such was my review, or so I thought. A credible first attempt from a new company, but it needed further refinement before it could be a real contender. I returned the speakers with a synopsis of my evaluation. Then I received a request from L.A. White Jr. that I again listen to the speaker. He had used my review as constructive criticism and redesigned the crossover. Since I had already sent my report to *Speaker Builder*, I conferred with the editor. We agreed to give the speaker a second chance.

The speakers came back with a completely new crossover (*Fig. 2*). Component values were changed, and the capacitor in the low-pass section was gone. And indeed, its sound had changed for the better. Although it retained its British character, it had lost its mellow, pleasant, euphonic sound, which was, after all, artificial.

The treble, the bass, and the response from the mid-bass through the lower midrange were all improved. There was more treble than before, but it was still deficient. For example, a solo banjo was soft, not crisp as it should be. Although it was still lacking deep bass, the mid-bass had been increased to the point where it was now adequate for a small speaker, and, to its credit, the bass was clean and not colored. The broad hump in the midrange was gone.

Although the speaker's sound was generally more accurate, many of its original flaws were still present: The weak dynamics, the poor imaging, and the "thickness" in the upper bass and lower midrange which was exhibited as roughness or coarseness of a solo male voice. And there was something new: listening to solo piano, I detected a strange resonance that might be described as a ringing, smear, or hangover. It was something that I would expect to hear from a cartridge rather than a speaker.

Listening Conclusions

As a reviewer I do not judge the characteristic generally referred to as British sound to be either good or bad, just a preference. I do judge how well a designer has met his objectives to achieve this effect. My opinion of the LAW speaker is that it provides a nice sound that is easy to listen to and can grow on you, but that it is not a state-of-the-art speaker within the British arena. Even in classical music, which is its strong suit, the lack of dynamics is a serious deficiency. I think its output in the higher frequencies needs to be enhanced. Also, if it is to be considered for jazz and rock, its deep bass needs to be improved.

Hunting Nonresonance

OK, so I had completed my evaluation of the speaker as presented to me, but I still had a nagging feeling that there was more to the story. I hadn't evaluated the design concept of the nonresonant enclosure. In order to detect the limitations of the enclosure, I decided to go further than my normal review practices and correct several areas of the design that were masking the characteristics of the enclosure. It was an interesting process—as each problem was corrected, another was uncovered. Each new problem had always been there, masked by the worse problem.

I thought I would start by experimenting with the vertical tilt of the enclosure. I found that the best sound was obtained by tilting the enclosure back 20 degrees. The front-to-back and side-to-side imaging were improved to where they were now acceptable. Tilting the enclosure back 20 degrees had a further benefit. The roughness or coarseness I had detected earlier was eliminated. The general improvement in the sound achieved by tilting the enclosure was not without a price—since the tweeter was now grossly off axis, the high frequencies were diminished. This speaker does not have treble to spare, so I decided to pursue an alternate approach. I rewired the tweeter so its polarity was the same as the woofer. Again, I experimented with the tilt of the enclosure. Now I found the best sound was obtained with the speaker in its normal vertical position. Except for the level of the high frequencies, I could tell no difference between a 20 degree tilt with the tweeter inverted, and no tilt with the tweeter not inverted. I left the tweeter not inverted.

Next, I added a felt diffraction ring around the tweeter. The ring cleaned up the sound, but the effect was slight because of the soft treble.

Proceeding with my diagnosis, I decided to isolate the resonance I had detected while listening to solo piano. First I disconnected the tweeter. The ringing was still present. Then I eliminated the tweeter and the entire crossover by connecting the woofer directly to the amplifier. The resonance was gone, so I knew the problem resided in the low-pass section of the crossover.

I connected the woofer only through the inductor. The resonance was not present. I then connected the woofer only through the resistor. Sure enough, the ringing was back. In retrospect, it is obvious what was happening—the resistor in series with the woofer completely destroyed the amplifier's ability to control the woofer.

If you ever wondered about the importance of damping factor, here is an excellent example of what happens with a poor damping factor. But why should there be a resistor in series with the woofer anyway? Good question!

Let's discuss the purpose of the resistor and inductor in the crossover. In the strictest sense these components are not part of the crossover at all. In fact, there is no low-pass section of the crossover—the designer has relied on the normal high frequency roll-off of the woofer to provide this function. The resistor and inductor are a low frequency contouring network. The resistor shelves the level of the woofer so that it has a low output across its entire range. The inductor bypasses the resistor at low frequencies so as to boost the bass output.

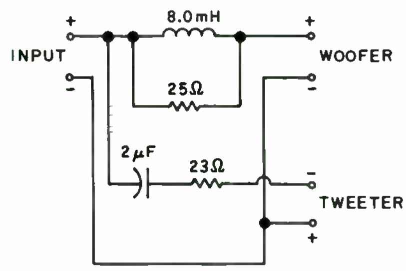


FIGURE 2: Crossover design with new component values eliminates the hump in the midrange.

The idea works: the bass is enhanced, but at the expense of a smeared midrange. I don't think that it is a good trade-off. I would rather have something missing than something wrong.

To proceed with listening to the effects of the enclosure, I put together a crossover that minimized the effects of the crossover. It included minimum series resistance, impedance compensation, and a better treble balance. I did not attempt to optimize the balance or extend the bass. I would not consider it a final design—just a tool to allow me to listen to the enclosure.

Enclosure Conclusions

The enclosure does not contain any damping material, and its open back does not allow the drver to be correctly loaded. I decided to test the effect of lightly stuffing the enclosure with polyester batting. The sound was definitely improved. Without the damping material, male voices were "chesty" and too full. There was a "boxy" effect, slightly reminiscent of an old console radio that my family had when I was a boy. With the damping material, music was more relaxed and open. It sounded refined with better definition and imaging. However, the soundstage that I had liked was severely diminished.

In spite of its being called a nonresonant enclosure, this enclosure definitely does have resonances that need to be controlled. I do not consider this enclosure design to be appropriate for a full-range speaker system: The bass is abysmal, and the dynamics are unacceptable. I suspect that the nice soundstage is a result of the sound emanating from the sides, back, and top of the enclosure. This is similar to the effect achieved with dipole speakers and those speakers employing some degree of reflected sound. Such speakers are known for their nice soundstage but less than optimum imaging.

David Davenport
Raleigh, NC 27615

L. A. White Jr. comments:

I feel that this magazine has made a poor choice in choosing Mr. Davenport to review these speakers. As he himself states, he had strong prejudices against the enclosure before he even agreed to do the review. Furthermore, he has a mistaken idea about what high fidelity is all about if he believes that rock music can be used to judge the accuracy of a speaker system.

With the exception of the drums and the vocals, all of the usual electronic instruments of rock music are always heard only through loudspeakers. The loudspeakers that are usually used are made for the express purpose of playing music as loud as possible with as little power as possible. When rock music is recorded, the sounds of the electronic instruments such as bass, the guitar and the keyboard are taken directly from the instruments and not from the loudspeakers that the instruments are played through.

Because of this, if Mr. Davenport wishes to have the sound of rock instruments as they are played at a concert, he should be using musical instrument and public address speakers. But of course these speakers are of such poor quality for other kinds of music that they are seldom chosen for home use. The fact that my speakers do a good job of reproducing an orchestra (as he states that they do) is a very favorable thing to say about them. The fact that these speakers do not produce rock music in the way that public *address speakers do, also speaks well of them since those speakers all have strong resonances which I find totally unacceptable. I also think Mr. Davenport's comment that the speakers are easy to listen to is very favorable since speakers that are unpleasant to listen to are of no use whatever.*

In regard to Mr. Davenport's criticism of the sound of his recording of a piano, I would venture to assert that my speakers are more accurate than whatever system he is using as a reference.

With respect to the frequency balance of my seakers, it is true that the high frequencies are not as prominent as on most commercially obtainable speakers. This is because high-definition electronics usually sound metallic on such speakers. Because of this, I have found many high-end dealers demonstrate speakers with an equalizer or tone controls which enable them to attenuate the treble.

In constructing the kit described in this article, the builder has various options he may use to adjust the frequency balance so that it is compatible with his listening room and the rest of his equipment. Of course there is no way these speakers can be made to have a strong resonance in the bass such as Mr. Davenport seems to prefer. Since speakers with strong bass resonanace and prominent treble (necessary to be heard over the bass resonance) are very easy and cheap to buy at your local electronic dealer or even at a discount department store, I see little use in supplying a kit for the construction of such speakers.

If someone (such as myself) wants speakers that do not have a strong bass resonance and prominent metallic treble, then this kit is the most practical way to go. The alternatives are all large dipole speakers that are very heavy and expensive and require a lot more power than these speakers do. Such speakers which are much favored by the "underground" audiophile press, have many of the same characteristics that Mr. Davenport finds so objectionable in my design.

The low-resonance property of my enclosure can be proven by a simple experiment. First, remove the woofer from the enclosure and suspend it some distance from any hard surface. Then apply an alternating current through the woofer and measure the impedance across the terminals of the woofer as the frequency of the current is varied from 20Hz to 20kHz. Then perform the same measurements with the woofer mounted in the enclosure. If the enclosure has low resonance, such as the enclosure tested in this article, the measurements will be similar. If the woofer is mounted in a typical acoustic supension enclosure, there will be a marked rise in impedance at the resonant frequency of the system. If the woofer is mounted in a typical bass reflex or vented enclosure, there will be two frequencies at which the impedance is markedly increased corresponding to resonant frequencies of that system.

Since Mr. Davenport did not perform these tests, his comments on the resonance characteristics of my enclosure must be dismissed as unsupported opinion and conjecture.

Editor's note:

Mr. White is seriously mistaken in accusing our reviewer of prejudice. Mr. Davenport uses the word "skeptical" to describe his attitude toward Mr. White's design when I asked him to review it. Skepticism is a necessary ingredient in any objective review. Prejudice, on the other hand, is a prejudgement; a mind made up beforehand. Mr. Davenport gave Mr. White's device every opportunity to show its capabilities. He worked with it further, beyond the design, using techniques that are in widespread use and for which there is abundant documentation for their legitimacy, to try to improve its sound quality.

I would also take the view that loudspeakers for home listening should reproduce any music faithfully. To excuse a system that does not reproduce rock music accurately on the grounds that the system is really designed for orchestral music which is not electronically reproduced is begging the question.

—E.T.D.

Speaker Design Considerations—

What You Should Know

Over the years *The Audio Amateur* and *Speaker Builder* have included many articles on specific topics of loudspeaker system design. The astute reader could build a pretty good system by incorporating the material from these articles in his design. However, I have noticed that many speakers presented to me for review have been lacking one or more features that should have been employed. Therefore, I would like to summarize some of the things I think a designer should consider when developing a speaker.

First, listen to the speaker objectively while trying different things. Experiment with tilting the enclosure. Try a diffraction ring around the tweeter. I must say that I haven't found a case where it doesn't help, but see for yourself. Listen to selected parts of the system. For example, disconnect the tweeter, bypass the crossover, and listen for enclosure colorations using only the woofer.

I can't overemphasize the importance of the crossover design. First, and foremost, use high quality components—it really makes a difference. The crossover does three things: It selects the range of frequencies presented to the drivers; it sets the level of the signal presented to the drivers; and it compensates for impedance aberrations of the drivers.

When adjusting the relative signal level presented to the drivers, keep the series resistance as low as possible—particularly with the woofer. It is a good idea to make the level of the treble easily adjustable, either with an L-pad or external resistors. If the speaker is to be sold, then it is important to allow the user to set the level to his taste in his system. It's not as important if you are building a speaker for yourself and can fix the level for your system.

There have been several articles in *SB* on impedance compensation. The effects of correct impedance compensation are audible. Use it.

I'd like to close by saying: "it isn't nice to fool mother nature." By this I mean that the designer should understand the underlying physical principles and work within them. That isn't to say that he shouldn't try something new in applying those principles. There are many different loudspeaker designs, and I expect that there are as many again yet to be discovered. Remember, "there ain't no such thing as a free lunch." There are many tradeoffs to be made. It is important to understand what is given up when something else is gained. Physical laws cannot be broken to achieve some desired effect, such as more bass.

David Davenport

Craftsman's Corner

The Swan IV

Since it was first described by our contributing editor, Joe D'Appolito, in *SB* 4/84; his high-power, symmetrical dual mid-bass driver, satellite speaker system has undergone an evolutionary change.

In its latest incarnation, dubbed the Swan IV and shown in the accompanying photo (and on the cover), the complete system delivers SPLs in excess of 115dB throughout the audio spectrum with amplifiers rated at 200W/channel into 4Ω.

Improvements include the selection of new mid-bass drivers, redesign of the satellite passive crossover and enclosure, and the development of an all new complementary bass speaker for the satellites with electronic crossover for bi-amplification.

For those of you unfamiliar with the original design, each satellite enclosure (16.5″ by 8″ by 11″) contains a single tweeter driver centered between a vertically symmetric pair of mid-bass drivers. This unique geometry creates a virtual acoustic center for the satellite coincident with the tweeter. This driver arrangement stabilizes the satellite's polar response pattern and therefore its stereo image, to provide extraordinary reproduction of the spatial aspects of localization and recording hall ambience.

Cast-frame mid-bass drivers, which are matched to within 1dB, are now used in the satellites. The satellite crossover network has been redesigned to accomodate these new drivers. In addition to conventional frequency domain measurements, time domain measurement techniques were used in the redesign of the tweeter/mid-bass driver crossover. The new crossover produces essentially perfect response to band-limited pulses, especially in the critical crossover region around 2100Hz where interdriver time delay has been reduced to less than 5mSecs. Overall satellite response is ±1.5dB from 100Hz–16kHz, with a slow, smooth roll-off beyond these limits.

Separate bass speakers are used for each channel to maintain the low-frequency stereo integrity now available on CDs. Each bass speaker contains a pair of polypropylene cones, heavy magnet bass drivers in a modestly-sized cabinet (32″ by 16.5″ by 16″) of 100 liter volume. The enclosure height correctly positions the satellites for listeners either standing or seated.

The bass speaker employs an unconventional sixth-order, bass reflex design. An electronic crossover is used to obtain bi-amplification of the satellite and bass speakers at a crossover frequency of 165Hz. The first-order crossover uses a subtractive circuit topology to produce "transient perfect" melding of satellite and bass speaker acoustic responses. The crossover also equalizes the bass speaker with 8dB boost at 25Hz, to provide flat response to 22Hz. Frequencies below 22Hz are rolled-off at 12dB/octave to prevent excessive cone motion at infrasonic frequencies. For those preferring a closed box response, the bass speaker ports may be sealed to yield flat response to 27Hz using the same equalizer. Maximum undistorted SPLs in the lowest octave are reduced by up to 12dB, relative to the bass-reflex response.

Jim Bock, a longtime audio enthusiast and master cabinet maker, has taken great care in the cabinet redesign. He has replaced the cumbersome asphalt/sand damping compound application with a simpler and equally effective, epoxy-bonded laminate construction. A new grille cloth mounting technique and front panel treatment eliminate high-frequency diffraction effects. All cabinets are finished in natural oak veneer, stabilized with epoxy resin and coated with a satin polyurethane varnish.

An article detailing the complete system, including driver selection, passive and electronic crossover descriptions, and cabinet construction, is due for *SB* publication in the near future.

Book Report

Familiar Beginnings

Audio Anthology, Vol 1. Compiled from *Audio Engineering,* May 1947 to December 1949, by C. G. McProud. Marshall Jones Co., Francestown, NH. Distributed by Old Colony Books. $16.95.

Reviewed by Gary A. Galo

Some readers will remember the early days before *Audio Engineering* magazine was renamed *Audio.* How times have changed! Today, a consumer-oriented publication with such an intimidating title would probably count its circulation in the dozens.

There was a time, however, when audiophiles (once called hi-fi buffs) were not appliance operators. They were technically-minded and could read a schematic diagram and work from it. Today, owners of high quality audio equipment may have little technical knowledge of their equipment. *SB* readers are an unusual group, not at all typical of the audiophile mainstream.

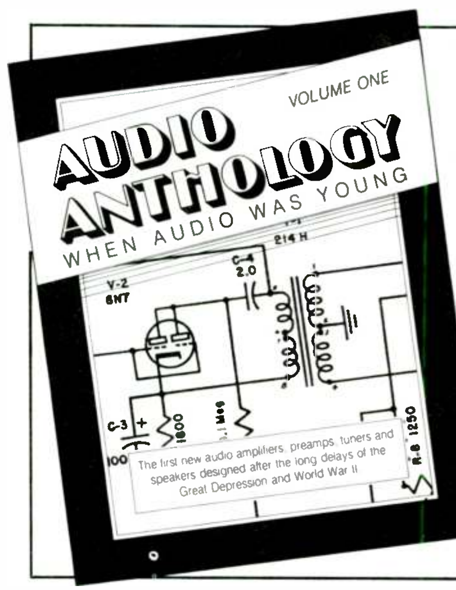
Over the last 30 years a similar phenomenon has occured in the field of amateur radio. Today, many radio amateurs have never plugged in a soldering iron. They purchase equipment off the shelf and operate it as a glorified CB, in contrast to the days when hams built all their own gear.

The legendary C. G. McProud founded *Audio Engineering* in 1947. This anthology, from its first two and a half years of publication, thirty-eight articles in all, covers virtually every essential component in the sound reproduction chain.

Electrical recording and reproduction had been in existence less than 25 years when these articles were published. The introduction of the modern long playing record and the first use of magnetic tape recording in the US (1948) were significant contemporary developments. This volume documents the beginnings of what soon became known as "high fidelity."

Ten articles are by McProud, himself, who was well versed in many areas of audio design. His articles cover amplifiers, loudspeakers, radio receivers (both AM and FM) and dynamic noise suppression.

I believe that knowledge of our past helps us to better understand the present. This anthology provides a wealth of such information. But these articles are not to be treated as mere historical documents. McProud had high standards. Devotees of vacuum tube equipment will find many circuits which can be modified to bring their performance up to current standards, and much of the circuitry was state of the art at the time. McProud's article, "Residence Radio Systems," gives detailed coverage of a series of modules which together make up an entire high fidelity receiver. The phono section features variable equalization.



Those who have built, or even aligned, radio frequency circuits know how critical are component selection, layout and construction to the proper operation of such circuits. McProud's work here is extremely impressive. His handcrafted chassis show a mil-spec approach to construction and in-depth knowledge of proper layout. Quality construction appears to be a hallmark of all of the designs discussed in *Audio Anthology,* as numerous photographs illustrate.

As many readers may know, I am an avid collector of 78rpm recordings. For building equipment for playback of historical recordings, this anthology contains no less than six articles on preamplifiers with variable equalization. Two of these are by McProud, one describing circuitry allowing three bass turnover frequencies, and the other covering high frequency equalization.

David C. Bomberger describes a complete phono preamplifier with continuously variable bass and treble equalization. Another excellent preamp, wih switchable bass and treble turnover, is described by Paul W. Saint-George and Benjamin B. Drisko. Those in need of a preamp equivalent to the McIntosh C-8, for historical recordings (*TAA* 1/85), will find the two articles of enormous value.

McProud gives special attention to the RCA 6AS7G twin triode tube. The use of indirectly heated cathodes in triode output tubes was relatively new in 1947 and McProud discusses the advantages in his article "High Quality Amplifier with the 6AS7G." It is interesting to note the current trend returning to the use of triode output tubes rather than pentodes. This sounds vaguely familiar!

McProud has two other articles on 6AS7G-based amplifiers. In "Compact 6AS7G Amplifier for Residence Audio Systems" another modular design is analyzed, the components forming a complete integrated amplifier. McProud's favorite dual triode is used in the control preamplifier as well as the output stage. This amplifier features a power output of 6.5W with 1% harmonic distortion. Paul Klipsch was, no doubt, pleased.

I can't resist quoting McProud from this article: "The performance of any amplifier depends to a large degree upon the quality of the components used." If the foremost audio designers realized this in 1948, why should that statement provoke controversy in 1987? We now have techniques to measure component quality, thanks to Walt Jung and others, but similar arguments made today, many of which have been stated in these pages, invariably result in disagreements. We have had great teachers, yet how little some have learned over the past 40 years.

Nearly all the authors represented in *Audio Anthology* are unfamiliar to most contemporary readers. One author who was still active not long ago is Richard S. Burwen. In the 1970s he designed and marketed the Burwen Dynamic Noise Suppressor. His 1948 article discusses the use of negative feedback for preamplifier equalization. Like most of the preamplifier articles in this anthology, the G.E. variable reluctance (VR) magnetic pickups are treated as a standard whose characteristics must be considered when designing a magnetic phono pre-amp. The days when General Electric was associated with high quality audio equipment ended decades ago, but the G.E. VR series of moving iron pickups were industry standards.

Two articles discuss dynamic noise suppression, but neither is by Burwen. John D. Goodell's piece describes the legendary H. H. Scott's circuitry. Goodell makes realistic claims for this device, that it will not remove all the background noise, or preserve all the information contained on every record under conditions of maximum compression.

The power amplifier projects are a reminder of our increased power requirements over the years. Walter T. Selsted and Ross H. Snyder describe a 30W power amplifier as a "High Power Triode Amplifier." This design uses a pair of 211s as output tubes in a push-pull arrangement. Since the 211 employed a directly heated cathode, the designers used a standard hum reduction scheme for the output tubes.

Commenting on its performance, the authors say: "Although all the tests indicated the amplifier was suitable for its purpose, the final judgment would have to be that of prolonged listening by critical people. The nature of the distortions which offend a 'golden ear' still defies analysis, but whatever their causes these offenses had to be excluded from the present amplifier if it were to be useful."

Again, how little has changed in 40 years. Considering the controversy surrounding double blind testing, I find the reference to prolonged listening to be especially interesting. I wonder whether the authors of the article received any malicious letters as a result?

The cathode follower is given its due in an article by W. E. Gilson and Russel Palvat. The amplifier damping factor is addressed, and a push-pull, cathode follower output stage is used to keep the source impedance as low as possible. Pulse testing was used to evaluate loudspeaker damping and oscilloscope photos are included.

One of the most interesting articles is by David Sarser and Melvin C. Sprinkle, titled the "Musician's Amplifier." Sarser was a violinist in the NBC Symphony Orchestra under Arturo Toscanini. Sprinkle was an electrical engineer with a serious interest in music. Their amplifier is based on the classic British "Williamson" circuit (no, not our friend Reg Williamson). This article was the first published in the United States which described a Williamson-based circuit. The original Williamson circuit was published two years earlier (1947) in *Wireless World*.

Sarser and Sprinkle said the sound of the amplifier is all that matters. Their goal was for a music system in the home to reproduce what they heard in the concert hall. This amplifier measured remarkably well by 1947 standards. Frequency response is flat from 20Hz to 80kHz. Power output is a modest 12W. [The design was reproduced almost part-for-part by Heath Co. shortly afterward.—Ed.]

The articles on loudspeakers feature designs based on horn tweeters and bass reflex woofer loading. Efficiency was of prime concern in the late 1940s, which is understandable given the power output of even the largest amplifiers. There are no three-way systems. McProud describes a two-way system in which you must construct your own multicellular horn. This is truly a "from scratch" project, involving cutting sheet metal, soldering the pieces together, and then pouring molten tar onto the horns to coat the inner walls, deadening resonances, with appropriate warnings about the dangers of handling the hot tar. McProud gives calculations necessary for determining the dimensions of the horns.

McProud's low frequency section is a large bass reflex design. None of the mathematics we now use for tuning vented loudspeakers was available. The enclosure dimensions and port size are based primarily on empirical knowledge. The author notes the importance of reducing the impedance rise at resonance if output tubes other than triodes are used. His driver recommendations include two classics of that time, the Lansing (JBL) Signature D-130A and the Altec-Lansing 803. The completed loudspeaker looks very much like Altec's Voice of the Theater.

McProud elaborates on his views about crossover design in "Practical Dividing Networks." He preferred 12dB/octave crossovers in a series configuration. He also provides charts for those who wind their own inductors. Most home builders had to wind their own, since crossover coils were not readily available in 1947. Calculations, without concern that loudspeakers are not resistors, are based on standard resistive loads, with no consideration for impedance compensation networks. This is a reflection of the state of loudspeaker design at that time.

Perhaps the most unusual article is McProud's "A New Corner Speaker Design." This article describes a corner horn loudspeaker enclosure which also houses the chassis and picture tube of a television receiver, introducing high fidelity sound with TV installation. Television was even then "firmly established as a home entertainment medium." McProud's design separates the picture tube from the rest of the TV chassis. The chassis is housed behind the eight cell horn tweeter. The picture tube sits in its own subenclosure, which is below the tweeter housing.

McProud's system used an RCA 630TS television chassis, available as a kit from Tech Master Products Company. His photo shows the blank, punched chassis ready for assembly. Building your own television was, undoubtedly, very exciting in 1949. It is interesting that McProud measured the frequency response of the loudspeaker above 200Hz using warble tones, rather than sine tones from a conventional signal generator. Forty years later the warble tone measurement system still fails to receive the attention it deserves, although the efforts of Audio Control and others are encouraging.

Historical perspective on the vented loudspeaker is enlightening in the article by Planer and Boswell. Dr. Neville Thiele and Dr. Richard Small now have transformed the bass reflex boom boxes of the past into sophisticated designs, which are mathematically predictable. Bob Bullock's superb articles have translated their work into useful information for the speaker builder. Planer and Boswell did not opt for the trial and error approach, unlike many of their contemporaries. Their approach was a mathematical one.

Audio Anthology is detailed, fascinating and enjoyable reading, and offers a unique perspective on the early days of high fidelity. Perhaps most enlightening are the many references to the listening-versus measurement approach to equipment evaluation, particularly because the debates on this subject are far more heated today than they were 40 years ago, even though defenders of the measurements-only approach are relying on similar tests to those used in 1947.

This anthology from *Audio Engineering* is the first of six. I highly recommend this anthology, and I hope reader interest will justify more of them.

Electrostatic Design

Electrostatic Loudspeaker Design and Construction, by Ronald Wagner, TAB Books Inc., P.O. Box 40, Blue Ridge Summit, PA. $14.95, 248 pp.

Reviewed by Roger T. Sanders

This book's title suggests it is a guide to electrostatic loudspeakers, but it is considerably more than that. It has an interesting section on the history of loudspeaker transducers; and an extensive section on electromagnetic speakers, including theoretical and mathematical design parameters, plus a discussion of the various compromises involved.

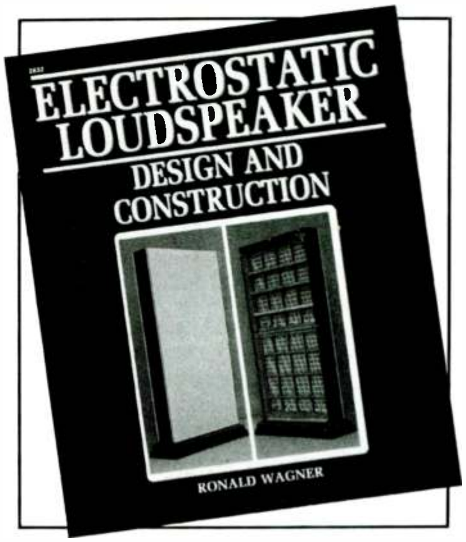
The author gives us the expected discussion of electrostatic loudspeaker theory and problems as well as a complete step-by-step guide to building the author's design for a full-range electrostatic loudspeaker. Any loudspeaker design serves the designer's preferences. I wish I could have read Mr. Wagner's discussion of these preferences as he designed his loudspeaker, which could easily have been a separate chapter.

This book's best feature is that it offers, in one text, the best accumulation of formulas related to speaker design parameters and performance that I have seen anywhere. Readers with an engineering background will find this a gold mine of information. Readers without an engineering background will find this difficult reading and very few practical, rule-of-thumb guidelines. Even so, those readers can follow the step-by-step approach for building a speaker that the author outlines and should achieve excellent results.

Furthermore, there is considerable discussion of testing procedures, electronics for electrostatic speakers; and an overview of commercial electrostatic loudspeakers and noteworthy articles. Of particular interest are Peter J. Walker's hard-to-find, original articles on electrostatic loudspeakers, reprinted from *Wireless World*, 1955.

In any book of this sort, the most interesting part is the end result, in this case, the speaker. I considered building Mr. Wagner's design to evaluate its performance, as I have built many kinds of electrostatic loudspeakers. I had no ambitions in this case, because this is the most complicated, time-consuming, expensive, and inconvenient way to build an electrostatic speaker I have ever seen.

For example, instead of using perforated aluminum or wire-grid stators, he used plastic plates, which required that an industrial firm perforate them. The sharp edges must be trimmed, the plates have to be coated in special patterns with conductive paint, and then painted with insulation as well. This process is very costly and time-consuming, and since many readers do not live in large industrialized areas where perforating is available, the difficulties of arranging for shipping; providing perforating specifications and pat-



terns, plus precise drawings for the work; as well as getting the stators back unbroken; would be overwhelming for most amateurs.

Mr. Wagner uses an electrically segmented stator and coats this with conductive paint. Avoiding electrical bridges between the adjacent sections is another difficult task and the plastic stators are not very strong mechanically, requiring hard-to-build wood ribbing to support them. How much simpler to cut strips of perforated aluminum with scissors and glue them to an underlying insulating structure. I have built such structures and found them easy to make and excellent performers.

Mr. Wagner notes that graphite, used for the conductive coating on the Mylar membranes, is typically used in amateur electrostatic loudspeakers. But he believes he has found a superior product—liquid Ivory soap. In correspondence with several thousand readers and builders of electrostatic loudspeakers, this is not the first time I have heard of this. I have not used anything but graphite myself, but two correspondents who used Ivory soap said initially it worked fine; but with time the soap came off, and the speaker became inoperative.

Mr. Wagner also built a mechanical stretcher, another complicated structure, to tension his diaphragms. This is an alternative to Mylar film, which may be glued to the stators and then shrunk to good tension with a heat gun. While his main purpose in using a diaphragm stretcher is to achieve uniform tensions, I have found excellent results can be obtained by heat-shrinking Mylar. Furthermore, slack areas in the diaphragm or areas that arc excessively can be further heat-shrunk after the speaker is in operation. The author goes to great lengths to insulate and test his stators. I have found insulation to be unnecessary. I could detail more examples of difficulties. In sum, the design is very complicated and difficult to build, compared to other methods.

The author's speaker appears to have good frequency response and I would expect it to have the legendary clarity typical of electrostatics. Still, Mr. Wagner does not give much information regarding the performance parameters of his loudspeaker. The frequency response is specified on a real-time analyzer, but we are given no information regarding directionality. Also lacking are sound pressure levels, which are important in electrostatic speakers and satisfactory ones are difficult to achieve, particularly in a full-range design. I see many areas of ineffeciencies in this design, such as large perforations in the stators, electrical segmentation, stray capacitance, and large diaphragm-to-stator spacing, which decreases sound pressure levels. I imagine many builders will be disappointed with the speakers in this regard.

The author offers no alternative ways to solve various problems. For example, the dipole radiators used in this design will have severe bass loss unless this is compensated for in some way. This particular design uses segmented diaphragms to accomplish this, but it can also be achieved with equalization or with a hybrid electrostatic/electromagnetic system. All of these techniques are compromises, and I am not saying Mr. Wagner's choice of using segmented diaphragms is good or bad; only that other options were not discussed.

In a book whose purported goal is to provide information to design electrostatic loudspeakers, all of the various design parameters and all possible solutions should be discussed in detail so a designer can choose from the widest variety of options.

Mr. Wagner clearly has done a thorough research and presents references and reprints of articles in electrostatic construction and design. He is aware of articles in *Audio Amateur* and *Speaker Builder* which offer simple practical designs and I find it odd that he excluded these from his list of noteworthy articles, choosing instead complicated, engineering-oriented articles lacking how-to information. He undoubtedly used an extensive bibliography which he could have, but did not list, to assist those looking for further information.

In summary, this book's mathematical theory and expressions of design criteria for loudspeakers are important, but the speaker design offered is unreasonably complicated, difficult to build and expensive. A useful, lively discussion regarding the speaker design and Mr. Wagner's preferences was, unfortunately, excluded.

Software Review

Loudspeaker System Design

Computer-Aided Speaker Design, V1.3.2 by Scientific Design Software, P.O. Box 3890, Northridge, CA 91323. Includes two 5" disks and 80-page manual (3-ring binder). Suggested list price, $200.

Reviewed by Bob White

New loudspeaker design software always interests me. In the old days all I had was my HP calculator and plenty of leisure time to experiment. The days of the calculator and spare time have fallen by the wayside in favor of computers and children. Over the last 19 years of speaker building, I have collected nearly all the equations I can use. I now have trouble finding the equations when I need them.

The software package presented by Scientific Design Software's Ted Telesky is the best collection of loudspeaker design technique I have seen. It seems to be based on work by Dr. Richard Small, Dr. Neville Thiele and the interpretive works of Don Keele. Design solutions that took a calculator and many hours now can be done in no time. You get a display of all the parameters and a full graphic representation. The package comes with two diskettes and a manual, to explain the various parts of the program.

Review

The software is entirely menu-driven and comes with a program to configure itself in your hard drive. The main menu consists of six sections:

1. Sealed/Vented system design
2. Crossover design
3. Utilities
4. Driver file maintenance
5. Graph file maintenance
6. Other activities

Section one, system design, includes the two choices which constitute the main thrust of this package. Both are excellent in their presentation and execution. I have one minor complaint though. I think the maximum sound pressure of the design should be displayed in the same manner as the small signal response and the maximum input power. I have spoken to the author of the program and he assures me it can be revised.

Section two has several features included with the calculation of crossover components. All of the crossover rolloff choices are Butterworth and have a handy feature of component recalculation in the event of an unobtainable component. Also, the crossover design section offers two ways to compensate for the driver inductance, depending on the data. You can either input the driver inductance directly or get the program to calculate it for you by entering some magnitude values from the impedance curve. Either way, a network is calculated to equalize the inductance of the driver.

This section has an additional surprise. It has a menu choice for designing a circuit for removing the resonance of any infinite baffle driver. This is most helpful when crossover points are too close to the fundamental resonance of the tweeters. The L-pad attenuator design, which calculates the required resistors for a user input dB loss, is also useful.

The third section presents nine design utilities. The ones I find most useful are: V_{as} from moving mass, reference efficiency from T/S parameters, and F_b from vent diameter and volume of box.

In section four, Driver File Maintenance, this software package justifies its high price. It has an entire, built-in data base. You can enter specifics for any speaker manufacturer, as well as any model from a manufacturer. The version I reviewed includes data from 41 manufacturers with 722 models and all the parameters the sealed/vented design section needs to design a system. The program can search by parameter and by enclosure/F3. It also has complete housekeeping capabilities for making a new driver data disk, reading free disk space, and so on.

Section five contains a group of file utilities for displaying, listing, and deleting graphs saved under section one. There are also provisions for creating a new graph data disk and displaying the amount of free disk space.

The final section contains updated information and two helpful utilities for making extra data disks, and configuring screen colors.

Conclusion

Scientific Design Software is commended for producing such an attractive package. Although it is really aimed at the commercial designer who can't waste time searching for that right combination, the home hobbyist should find it very useful.

Although higher priced, this is a beautiful piece of programming expertise with many useful, time-saving features. It is presently menu driven with a window driven version in the works. I look for more software from SDS in the future. (I hear rumors they have a crossover optimization program and a horn design program in progress.)

Book Report

continued from page 43

Our Sacred Realm

Bluff Your Way in Hi-Fi by Susan Hudson and John Crabbe, Ravette Ltd., Star Road, Horsham, Sussex, England, price: £1.

Available from Old Colony, PO Box 243, Peterborough, NH 03458, $2.75 plus 50¢ for postage and handling.

Reviewed by John Cockroft

This is a small book as books go (and maybe this one should). On a weight/cost basis it defies the laws of physics, coming in at 44 grams a pound. It is probably one of the most magnificent examples extant of the type of human enterprise that caused the British Empire to slip from a multi-continental conglomerate to the two pence and ha'penny little island it is today—and this little book has the lack of pages to prove this point. If Alastair Sim and the late Dame Margaret Rutherford were a book about high fidelity, they might read like *Bluff Your Way In Hi-Fi*. Those balmy, blimey, blooming Britishers are at us again and I haven't had so much fun since reading Stephen Pile's *The Incomplete Book of Failures,* (Stephen Pile, E.P. Dutton, 1979; the official handbook of the *Not-Terribly-Good Club* of Great Britian), which also came crawling our way from that little rocky rock, whose basic function is apparently to create the English Channel out of a rather large body of salt water.

Those who must take themselves seriously (or ex-Marines) may balk a bit at the biting treatment dished up here, but most others will probably find much to enjoy

even though it will be mostly at their own personal expense. If this tiny tome were any more of a mirror than it is, it could be used in the process of shaving one's self (providing a razor was also supplied for the job), but I fear one will have to pay in full, for there can be no cut rate. With such a vast display of naked truth revealed in words it might have been far closer to the fleshy point if Hudson and Crabbe had canted a bit and named their bare-boned little thing, *Buff Your Way In Hi-Fi.* Of course that might defeat the general purpose, because in the buff is hard to bluff.

This little book is an allusion to an illusion, but it is based on sound principles. It sometimes offers the kind of fun one experiences while receiving a swift kick in unnamed regions. But we laugh, knowing that we are all indicted; that each one of us has egg to wipe from his jowls.

BYWIHF is a volume the family man would be well advised to order in the traditional "plain brown wrapper." If the little woman ever gets wind of it, the jig is up. She'll get the wind up and his hobby will be all over. He'll be back to raking leaves, cutting grass and sitting in a pink satin chair in Bloomingdale's doing Fast Fourier Transforms on a hand calculator, as the dresses and shoes pile up on the satin setee beside his satin chair (the old Dracula and the light bit). The book is laid out somewhat in the form of a hi-fi encyclopedia that somehow became interleaved with a book on abnormal psychology. If you could but read between the lines you just might catch a glimpse of the final doom of Armageddon staring you in the face, winking.

Bluff Your Way In Hi-Fi is a profound book. In it you will find material that you hope you could not find in any other source. For instance, the section on *Phase* takes one into the bizarre problem of arbitrary speaker morality and all the ruckus that could involve. In the section on *Loudspeakers* we are confronted with the following noble British statement: "The perfect speaker would have no mass and no dimensions. The perfect speaker does not exist, and if it did, it still wouldn't." This is obviously a book that one will not take lightly (except for the tiny fact that it weighs a mere 44 grams).

Among the abstruse and arcane sections and selections are such catchy little gems as *Tweakery, Failures & Oddities, Ears, Controversies,* and *Right Stuff.* The erudite audiophile now finally has, at his very fingertips, a volume worthy of his closest perusal and his most involved attention. With the listing of hi-fi types one can learn just where he fits in the celestial audio cosmos scheme. (Maybe with a little effort and a change of address he could slip from one placement to another, rather like a shifting of alignments [of course he'd have to do it on the Q_t].) You may not learn how to "tweak up" the old system, but you'll possibly come away with a bunch of glib reasons (long yellow ones) why it is better to leave it just the way it is and that could be a sound decision.

Should any feel a touch of paranoia and the yoke of persecution being set upon his shoulders at the thought of these two strange foreign people twisting and clawing their fiendish way into the tender portals of *our sacred realm,* let me reassure him that we have not alone been singled out for this cruel sport. This horrid little book is only one of 25 similar flagellation instruments (with 40 more in preparation!) from accounting to publishing, (Dell, your day of doom has dawned); from marketing to sex, from bluffing (!) to law, from opera to golf. From here to the horizon; skinny little pocket books, marching ever marching, seeking to subjugate us for the Crown.

Noticeably absent from this skinny little patch of pages are real issues that face audiophiles today, such as that most crucial problem of just how does one get the damn plastic jacket removed from compact disc packages without cutting the hell out of one's fingers (to say nothing of how to open the little box once you go through the initial Crusades) and the critical problem unique to speaker builders, concerning dogs and cats and rabbits. (A transmission problem definitely not for the sq*weem*ish.)

My ultimate conclusion is that *Bluff Your Way In Hi-Fi* is a slimy little Limey book. To show those Britishers that we stand firm and shall never be bluffed, we should take all copies (except perhaps mine) and toss them over the gunwales into Boston Bay. But I suppose if you managed to hang on through the eternal entirety of this report, you might just possibly like this little cukoo excuse for a book just as much as I did. Besides, how can I pan something that is just about thin enough to significantly reduce the deleterious effects of edge diffraction?

Technology Watch

Loudspeaker Psychoacoustics

By Peter Muxlow

A term you hear quite often nowdays: the loudspeaker is "psychoacoustically optimized," is pretty daunting when you encounter it for the first time. All it means is how you *hear*, or the brain interprets, the acoustic signal which appears at your ears.

Interaural crosstalk is a psychoacoustic effect which alters our ability to tell where the sound is coming from and therefore alters the stereo effect. Polk's patent (Am4630298) for his latest loudspeakers is designed to minimize this effect. Under ideal stereo conditions the sound from the left speaker would reach the left ear only—as in headphone listening. In practice, the sound spills over to the right ear—although a little later and at a lower level. This is known as interaural crosstalk.

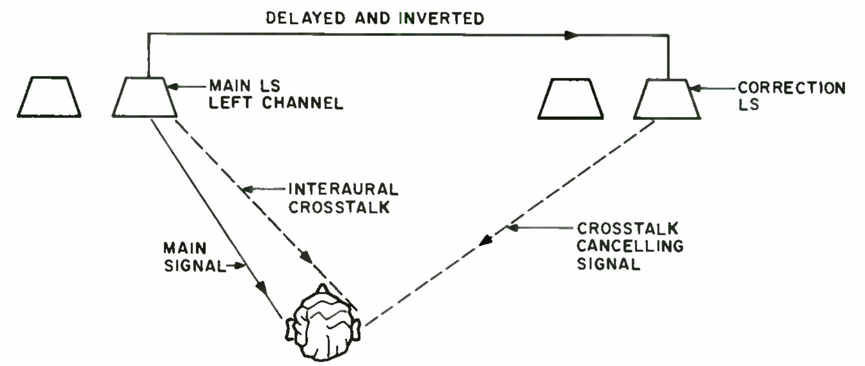


FIGURE 1

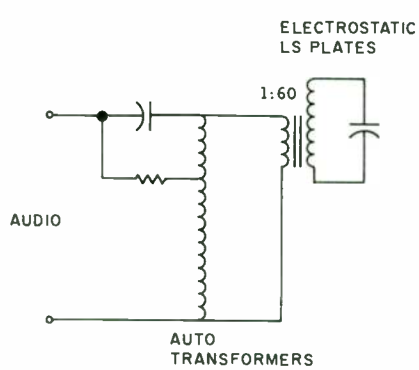


FIGURE 2

To overcome this crosstalk Polk takes some signal from the left channel and applies it, out of phase and delayed, to the right channel. The same process is carried out in the left channel (*Fig. 1*).

More details of Polk's approach can be found in his article in *Audio*, June 1984. This interaural correction signal has also been electronically applied to the other channel but radiated out of the main speaker, not a correction speaker as Polk uses. Carver's Sonic Hologram preamplifier has details of this.

A novel solution to the problems of driving electrostatic loudspeakers is given by Peters of the Dutch firm, Audiostatic. Normally, ESLs are driven by stepup transformers to get the required audio voltage to the plates. One of the difficulties is matching the amplifier to the ESL over the audio range, because the loudspeaker is pure capacitance. The impedance of this capacitor varies between 1000:1, at 20Hz to 20kHz, for a full range panel. The driving amplifier, a constant impedance source, must match this 1000:1 variation.

Audiostatic's answer is to equalize the signal which is fed to the primary of the stepup transformer by driving it with an auto transformer (*Fig. 2*). The auto transformer increases the drive to the primary of the main transformer at low frequencies. At higher frequencies the capacitor acts as a short circuit and bypasses the auto transformer. The cost of the auto transformer is relatively inexpensive because of its construction and its use at low frequencies only.

Passive crossover networks for loudspeakers are difficult to design and imple-

Continued on page 47

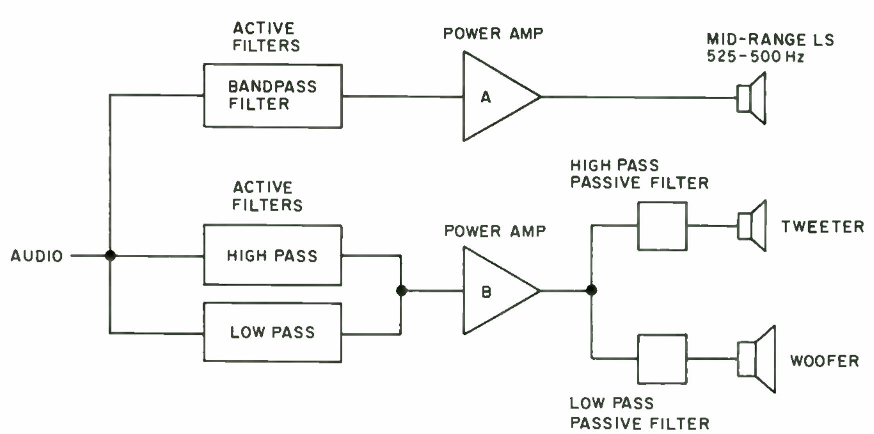


FIGURE 3

QUICK ELBOWS

PVC vents with limited space sometimes require tight or "quick" elbows. If the shallow bend elbow won't work for the diameter size of the vent you want to use, try the tight 90° elbow. It is better to use the larger vent with a tight elbow than going to a shallow turn and a smaller diameter vent.

For an example, take a straight piece of pipe 10" long plus the thickness of the saw blade you will use to cut it. Cut a 45° angle so the longest piece (A), outside of the cut, measures 7". The shorter piece (B), will measure 3" (*Fig. 1*).

Then cement pieces A and B together at a 90° angle (*Fig. 2*). Measure the length of piece A by placing the tape far enough inside to touch firmly the back of the bend. This should be 7" minus the wall thickness of the pipe. Next, measure from the open end of piece B on the inside of the bend by hooking the tape on piece A. This is 3" plus the pipe's wall thickness.

So, you can disregard the pipe wall thickness:

$$(7'' - W_t) + (3'' + W_t) = 7 + 3 = 10''$$

The pipe was 10" in length when laid out straight so it must be 10" long at the 90° bend configuration.

Now proceed with measuring the 90° tight turn manufactured elbow. I usually have overlength pieces of pipe on hand. After removing the burrs, I dry fit them firmly into the elbow. I make certain the pipes are all the way in for cementing.

If after dry fitting and measuring you find X = 8½", cut off 1½" for the proper length (*Fig. 3*). For the Y length, place a piece of dowel from the X end so it touches the elbow at the opposite end. Place the doweling on the inside of the bend and measure to the dowel from the Y open end (X + Y = 10").

To insure accurate vent lengths use a millimeter tape measure. This certainly beats trying to measure along the pipe centerline with a string. To those who ask, why don't you just use those two illustrative pieces for the vent? My answer is, go ahead; they will work as well.

James M. Tripp
Edmonds, WA 98020

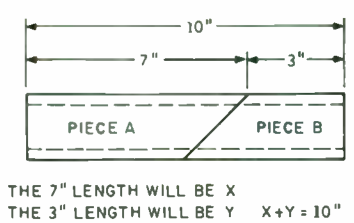


FIGURE 1.

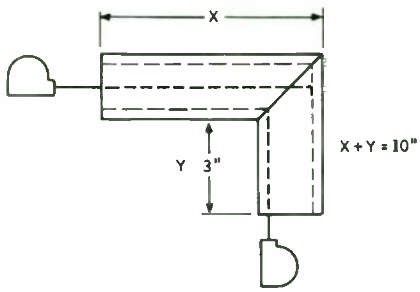


FIGURE 2.

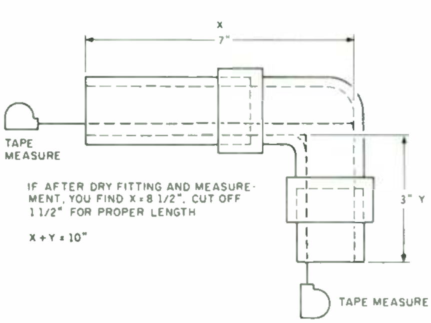


FIGURE 3.

Continued from page 46

ment. One way out of this difficulty is to use active crossovers, though a big disadvantage is the cost of multiple power amplifiers. If you want to split the audio into three frequency ranges, this requires three power amplifiers. James Kates shows a way of using two power amplifiers to cover three frequency bands (*Fig. 3*). Power amp A covers the frequency range between 525–5000Hz. Power amp B covers the full frequency range minus 525–5000Hz. At the output of amp B there are simple passive networks. These prevent the high frequencies from getting into the woofer and the low frequencies into the tweeter. The passive filters can be simple because its active filters provide the sharp cutoff at the bandpass frequencies.

COIL WINDING FORMULAS

I recently dealt with a coil winding problem and after reading the article ("Custom Wound Inductors," *SB* 3/82, p. 20) promoting 2:4:1 proportions (ID:OD:height) and suggesting that "random-wound" coils (presumably slightly sloppy) perform as well as neat ones, I thought this needed further study.

If you trouble to interpret the formula (I believe it's commonly accepted for air cores):

$$\text{Inductance} = .8(N \times A)(N \times A) \div (6A + 9B + 10C) \quad (1)$$

(where A = mean winding radius, B = axial coil dimension, C = radial winding thickness, and N = number of turns); understand that ideal proportions for A:B:C are 1/6:1/6:1/10. It's that simple because A, B, and C have an equal impact on the numerator. This translates into ID:OD:height ratios of 2.1:3.9:1. Another formula I've seen:

$$\text{Inductance} = .100275(n \times a)(n \times a) \div (.9a + b + .84c + .32bc \div a) \quad (2)$$

suggests ratios of about 2.07:4.45:1. I imagine other formulas will produce different answers to the question of optimum coil proportions.

We need not worry about proportions because their effect on efficiency or cost of the finished coil is minor. For example, winding on a former with a diameter 50 percent greater or less than ideal adds only a few percent to the DC resistance of the coil. In comparison, the result is roughly the same that random-wound, compared to neatly-wound, adds to the DC resistance.

I wrote Table 1 to study the effect of various factors upon inductance and efficiency. Winding by counting turns, as opposed to wrapping a pre-measured length upon a former, places a premium on accurate measurement of former dimensions. If you wind by counting turns, an error of a few percent in measuring the ID or height of your former appears to have a comparable effect on the inductance of the finished coil; if you wind to length, these errors are negligible. You can modify the method of estimating turns/inch by adapting the routine (see note 60) to your own style of winding. Also, adapting the routine to alternate formulas is a simple matter of changing line 510. The routine can provide reasonable dimensions for a coil former. It can also tell you what to expect with any handy former, and it will provide you with information to do the winding.

The program first requests the inductance of your coil. Next it asks for a wire gauge—give it one and, if that doesn't work out, rerun the entire routine with another. It also asks for an ID and height; you can refuse either, and let your machine optimize the unspecified dimension(s).

Incidentally, I did run the routine against Daniel Coyle's incomparably more elegant procedure. Minor differences in results occur, largely attributable to a difference in the method of estimating the number of turns/inch. Changing line 60 to read:

TPI = .841 ÷ DIA: EEF = 0

yields consistent results. Again, line 60 is a matter of user judgement. The data in line 60, as it stands, is a little tight, as well as feasible, and winding to those criteria will produce a coil a few percent more efficient.

David Mulkins
Oneonta, NY 13820

NOTES

10-20 request inductance and wire size.
30-50 calculate wire diameter, resistance/1,000 ft. and number of feet in a pound.
60 estimate the number of turns/inch—allow .003″ for insulation, 2% for sloppiness, and deduct .6 turns/layer for end effect. (This is not terribly critical, an error of 3% in estimating TPI will only throw inductance of the finished coil off by 1 or 2%.
70 provisional estimate of wire length.
80-110 request dimensions and branch to appropriate subsection.
120-180 use the results of subroutine 300 to estimate an optimum ID.
190 enter here with a specified ID but no specified height.
200 enter here when both ID and height are specified.
210-240 print routines.

SUBROUTINES

500 use wire length, ID, and height to calculate OD and inductance.
400 adjust rough estimate of length before executing 500.
300 estimate optimum height before executing 400.

```
10 INPUT "required inductance in milliHenrys ";L
20 INPUT "preferred wire gauge ";AWG
30 DIA=EXP(-.115938*AWG-1.12444)
40 RKFT=EXP(.231886*AWG-2.31999)
50 FTLB=EXP(.231889*AWG+1.14108)
60 TPI=.98/(DIA+.003): EEF=.6
70 LENGTH=257*L^.6/TPI^.4
80 PRINT "preferred coil dimensions in inches - enter 0 if no preference"
90 INPUT "id= ";ID: IF ID<DIA THEN 120
100 INPUT "height= ";HEIGHT: IF HEIGHT<DIA THEN 190
110 GOTO 200
120 ID=15*L^.2/TPI^.8
130 DID=.05*ID
140 ID=ID-DID: GOSUB 300: R1=LENGTH*LENGTH/L1
150 ID=ID+2*DID: GOSUB 300: R3=LENGTH*LENGTH/L1
160 ID=ID-DID: GOSUB 300: R2=LENGTH*LENGTH/L1
170 ID=ID+(R1-R3)*DID/(R1+R3-2*R2)/2
180 IF ABS((R1-R3)/R2)>.00001 THEN 130
190 GOSUB 300
200 GOSUB 400
210 PRINT:PRINT L;"mH", AWG;"AWG", "DC resistance=";LENGTH*RKFT/1000;"ohms"
220 PRINT "id=";ID;"in.","od=";OD;"in.","height=";HEIGHT;"in."
230 PRINT "length of wire=";LENGTH;"ft.","# of turns=";7.63944*LENGTH/(OD+ID)
240 PRINT "weight of wire=";LENGTH/FTLB;"lb."
250 END
300 HEIGHT=SQR(ID*3.4*L^.2/TPI^.8)
310 DH=.05*HEIGHT
320 GOSUB 400:S2=L1
330 HEIGHT=HEIGHT-DH:GOSUB 500: S1=L1
340 HEIGHT=HEIGHT+2*DH:GOSUB 500: S3=L1
350 HEIGHT=HEIGHT-DH+(S1-S3)*DH/(S1+S3-2*S2)/2
360 IF ABS((S1-S3)/S2)>.00001 THEN 310
400 GOSUB 500
410 LENGTH=LENGTH*(1+(L-L1)/(L+L1))
420 IF ABS((L-L1)/L)>.0001 THEN 400
500 OD=SQR(ID*ID+15.2789*LENGTH/(HEIGHT*TPI*TPI-EEF*TPI))
510 L1=.0058361*LENGTH*LENGTH/(13*OD-7*ID+18*HEIGHT)
520 RETURN
```

TABLE 1: The above routine uses Formula 1. For Formula 2 use: 510 L1 = 3.6576E−04 (LENGTH)(LENGTH) ÷ [.645 × OD − .195 × ID + Height × (1 + .64 × (OD − ID) ÷ (OD + ID)].

SB Mailbox

CORRECTION: BOXRESPONSE

In regard to the corrections made to the BOXRESPONSE program (*SB* 1/86, p. 42): my Apple would not digest Mr. Bullock's suggested change as printed. It should be:

```
400 IF O=2 OR O=4 THEN Y9=1: GO TO 420
```

In regard to the changes themselves: I am an enthusiastic builder and technical preliterate; I can't argue the "correctness" of my observations. However, I wonder if the utility of BOXRESPONSE has been sacrificed in favor of theory? The applications of BOXRESPONSE shown by White (*SB* 1/85) seem useful for predicting the potential power handling capacity and maximum SPL of a sixth-order system. This seems to be lost in the latest changes. Do Mr. Bullock and Mr. D'Appolito suggest that White's application is in some way invalid?

If Mr. White is in error, explain the practical advantage of a program that shows "very large maximum input power at low frequencies" and relies on Bullock's observation "This is alright because the equalizer will cut them down to levels the driver can handle." I don't see the revised numbers in the maximum input power column as being helpful unless unused, subsonic watts are free.

Jeffrey J. Nixon
Houston, TX 77071

Robert Bullock responds:

Your version of line 400 is correct. As far as the correction to the program is concerned, it in no way invalidates Mr. White's application. I think you are confusing two separate functions served by the equalizer in a sixth-order alignment.

Its primary function is to help shape the system response in the passband. *It is this function that concerned Mr. White. That is, he dealt with displacement limitation only above the cutoff frequency of the system. If he had used the corrected BOXRESPONSE, he would have reached the same conclusions, although some of his passband power numbers would have been slightly altered.*

A bonus function of the equalizer is that it serves as a "rumble filter"—it attenuates the input signal below the system's cutoff frequency. A vented system without an equalizer (fourth-order) may exceed the driver excursion limit with only a few watts input at low frequencies. This is reflected in BOXRESPONSE by the small power numbers at low frequencies with unequalized alignments. The large input power numbers at low frequencies for an equalized alignment (sixth-order) indicated the equalizer is performing its rumble filter function. The total system may now accept possibly hundreds of watts at low frequencies without exceeding the driver excursion limit. The absence of such large numbers in the original program was one of the signals that the program was incorrect. Now it is clear that a large subsonic signal can be safely handled by a sixth-order system.

CORRECTION: IMPEDANCE COMPENSATION

Concerning my article, ("Is Driver Impedance Compensation Worthwhile?" *SB* 3/87), some readers may wonder why I used a high-pass rather than a low-pass filter in evaluating the effectiveness of impedance compensation. It is certainly true that a rising impedance characteristic is potentially more troublesome when using a low-pass filter, however the article shows a case where impedance compensation may be desirable even when applying a high-pass filter.

The article contains an error that was present in the original manuscript: item 3, on page 32 should read "about one-third of an octave," instead of "one octave."

David J. Meraner
Scotia, NY 12302

OCTALINE ERRATA

Somewhere 'twixt the cup and the lip a few things got out of hand in my article on the Octaline (*SB* 3/87). On page 11, column one, we have feedthroughs suddenly shooting up, like Alice, until they are several inches taller than the speaker enclosure. On page 14, column two, we find the tweeter buried in a veritable forest of 4″ foam, its tiny voice lost forever. Of course, feedthroughs can't grow and foam must learn to keep its place. It is the speaker leads that protrude from the box (at that point in the construction) and the foam shrinks by the factor of its reciprocal to a mere ¼″.

Some foam, or a diffraction ring is a requirement for the Octaline. The sound without it is different. The particular tweeter used has an extremely wide dispersion, because its dome is only about ⅜″ in diameter (even though it is designated a ¾″ tweeter). Incidentally, I note that Radio Shack has dropped this tweeter from their new catalog. Apparently, the Audax TW-74 is the same tweeter and may be obtained from Madisound and other dealers. It is possible that the speaker will be found on the shelves at RS for some time to come.

The article did not mention that the Octaline should be constructed in mirror image pairs for stereo in order to maintain a balanced sound.

The equations suddenly appearing out of deep space on page 12, column three, preceded by the cryptic remark: "Here are several equations should you wish to experiment further," must have been a source of confusion to many. (I could paraphrase that and say to the editor: "Here is a bunch of words that I call an article should you wish to experiment further.") The equations should have been prefaced by the statement that I prefer my music slightly understated and others might desire a bit more treble. To this end, I presented the equations concerning the effect of changing the value of the 2Ω resistor. Changing the value of the resistor also changes the crossover frequency, requiring a possible change in the value of the 6.6μF capacitor. This could be eliminated by the use of a variable L-pad, but I wanted the constructor to hear the sound as I presented it before he made a decision to alter it.

In the beginning of the article I had no intention of stating that Bailey was referring to "especially the Hartley Boffle" (indeed, he didn't mention it at all). I mentioned the Boffle as it was the only system of the genre that I could recall the name of. The Boffle, as I recall, actually differed from *Fig. 1*, (p. 10) in two respects: It was open-ended and the baffles were of fiberglass [or felt—Ed.] batts rather than wood. The Hartley product was much closer to what Bailey was talking about than the Stromberg-Carlson product.

Further experimentation and listening has convinced me that the Octaline sounds better when the woofer is modified as shown. This reduces the efficiency of the system. Please remember the Octaline was designed for a small room, so efficiency can take a back seat when compared to the quality of the sound. The lead ring reduced the resonant frequency of the woofer by about 20Hz and raised Q_{ts} to about 0.5.

John Cockroft
Mountain View, CA 94041

TL PROJECT TIPS

A friend and I are planning to spend the rest of this summer and early autumn experimenting with transmission line enclosures. At present we are pretty well tooled up for the venture, having read helpful articles by Gary Galo (*SB* 1, 2/82) and everything else we could find, back to Bailey's contributions. (*Wireless World*, Oct. 1965.)

Since we've not dealt with this sort of enclosure before, we do have a couple of questions which we hope *SB* will help with.

How do we select 8″ and 10″ drivers appropriate to the service? What do we look for? We've considered the Dynaudio 21W54 (which are now pegged at nearly $100 each,) and a Speakerlab unit, the W84P. Letters to Speakerlab have gone unanswered. Are they still in business?

Our goals include low, clean bass sufficient for a moderate-sized listening room. We also welcome any advice.

A.J. Steen
Los Angeles, CA 90025

Mr. Galo replies:

I believe you will find much of what you are asking for in my two reviews of woofers (SB 2/85 and 3/87). Woofers for use in transmission line enclosures should have a Qts of less than 0.5. The Dynaudio driver you mention is excellent. The 21W54 is expensive, but its superb construction and excellent power handling capability make it a worthy contender.

Speakerlab was reorganized under new management, but I have not seen any signs of that.

Note: The address we have for Speakerlab, Inc. is 5321 Kirkwood Place, North, Seattle, WA 98103. The new head of the company is David Graebener. If you do not receive answers to your letter it is probably because the new management is primarily interested in local retail sales.—Ed.

FRIED MAKES A DIFFERENCE

I note, in an article by Gary Galo on woofer performance (*SB* 3/87, p. 47), Meniscus drivers are given high ratings, and it is stated that they are virtually identical to the drivers Fried Products uses in various designs, including our transmission line kits.

The words "virtually identical" are not precisely correct, because the drivers Carbonneau makes for us have some important differences from the specifications in the article relating to Meniscus drivers. The drivers we employ are made to our specification, for proper performance in our designs, and the differences cost us.

Properly speaking, we are not in the driver business, but the loudspeaker business. We do not solicit driver sales, but loudspeaker sales; and the prices we set on replacement drivers reflect the special nature of the drivers, as supplied through our dealer structure.

I note, in the same issue, Bill Sommerwerck's coverage of Chicago CES (p. 56) compliments our various loudspeakers and their steady progress and improvement. He specifically mentions our Beta, at $325 a pair—a speaker of which we are most proud—as having "remarkable performance," with all models taking "a big jump forward in immediacy and aliveness."

Coming from Bill, who is a firm advocate of exotic loudspeakers such as ribbon and electrostatic dipoles, this is high praise indeed. I firmly believe that high quality dynamic drivers, with proper crossovers (we employ series networks, unlike the preponderance of current designs), and time proper enclosures (we have designed enclosures that are unconventional, in factory manufactured loudspeakers), can equal or exceed the performance of exotic designs, and at much lower prices. Indeed, our very best designs are furnished as kits; and there are those who consider them superior to any of the very expensive designs that have come recently on the US market.

Irving M. Fried
Philadelphia, PA 19151

Contributing Editor Galo replies:

In order to give a complete response to Mr. Fried, I believe it is necessary to provide SB readers with some background information. In March 1985, Ed Dell and Mr. Fried agreed to have the Fried C/3-SM/3 loudspeaker system reviewed in SB. Because of my experience with transmission line designs, and my enthusiasm for them, the Editor asked me to take on the project.

Fried's top-of-the-line loudspeaker consists of two 2-way satellite speakers sitting on top of subwoofers. The subwoofers consist of 12" polypropylene drivers loaded into transmission lines. The loudspeakers were to be shipped to me as kits, which contain the drivers, crossovers and plans for enclosure construction.

After about two months the C/3 satellite kits arrived, without SM/3 subwoofers. Mr. Fried told me the woofers used in this system were on back-order from his supplier (Carbonneau Industries). In September 1985 I received a letter from Mr. Fried saying the woofers were still on back-order. He promised the delivery in three to four weeks, but sent the plans suggesting I could begin construction.

In April 1986 I received another letter from Mr. Fried saying the woofers were still back-ordered, this time due to a shortage of voice coils. He also said as soon as he heard from me "re: progress" he would try to ship the subwoofers. I wrote back, informing Mr. Fried I had not begun construction, since I was busy working on projects for which I had all the parts. I said July would be the best month to work on his speakers, during a break between two summer session programs. Mr. Fried wrote on May 6, saying I would have the subwoofers before the end of July.

Meanwhile, I called Carbonneau Industries and spoke with Gary Church. Mr. Church said there had been no shortages of woofers or voice coils.

By September 30, 1986, the kits had not arrived. I decided, and Mr. Dell agreed, that a year and a half of delay was quite enough. I returned the satellite kits with a letter of explanation.

I left open the offer to review the product if Fried was still interested, but told him not to ship anything unless the system was complete. My last letter from Fried, dated Oct. 1, 1986, was filled with apologies, and a promise to ship a complete kit "when we can pause." That was the last I heard from him.

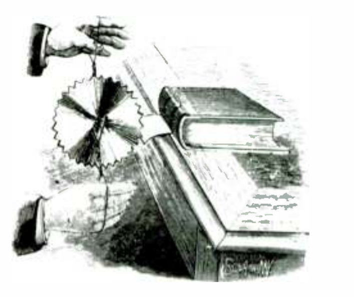
While I was in possession of Fried's satellite kits, I had a chance to closely examine the drivers and crossover. The Carbonneau 6½" mid-bass driver supplied by Fried was physically and visually identical to the 6½" Meniscus driver. When I spoke with Gary Church at Carbonneau, I asked whether the Meniscus drivers were, indeed, similar to those supplied to Fried. He stated they were, with minor differences in specifications. Our conversation left me with the impression that there were no significant cost or quality differences.

Throughout my year and a half of phone conversations and correspondence with Mr. Fried, I found it virtually impossible to get any concrete technical information from him. I think I made it clear that a review in SB would serve to educate both myself and readers on the technical workings of his loudspeaker system. Unfortunately, whenever I asked him any technical questions, he changed the subject. Whenever I tried to clarify a specific point, he would talk so fast I could not get another word in. I was usually given a lecture on how wonderful everyone thought his loudspeakers were, but my questions invariably remained unanswered.

I find Mr. Fried's letter regarding my review in the August issue to be absolutely typical. He says my assessment was not "precisely correct," but fails to give us any information which is "precisely correct." If his drivers differ in important ways from the Meniscus drivers, then why not tell us, specifically, what those differences are? If Mr. Fried wishes to pursue this further, he should give us a spec-by-spec comparison of the drivers, and specific reasons why his Carbonneau drivers are so much more expensive, including details on driver construction, quality control, and materials used in his units.

If he is willing to respond, and Carbonneau will verify what he says, I will be happy to change my statement from "virtually identical" to something like "physically similar."

I have heard many excellent comments on Fried systems from people whose opinions I respect. Mr. Fried had, and still has, a chance to allow SB readers to see for themselves, but he has failed to follow through on his promise. I am disappointed that he has given us yet another lecture on how excellent his loudspeakers are.



POLARITY PUZZLE

I thank Tom Cox for his interesting article/experiment ("A 2 x 4 Transmission Line" *SB*, 2/86, p. 9). I have three questions for him. I purchased the two other European loudspeakers of America models, the 5402/Audax and the MG102/Audax which have the same center post tweeter and polypropylene woofers. I tested the cone movement of each pair by connecting the positive terminal of a 1.5V battery to the positive marked terminal on the driver and the negative of the battery to the driver negative. The cone did not go outward but inward, which is opposite to all the other drivers I have tested.

Did you observe this same feature in the 4502/Audax and compensate for it by reversing the polarity of the other Audax pair?

I wonder why you did not use two pairs of the 4502s and not connect the tweeters on the top and bottom drivers? This would have given you four polypropylene cones and the same model drivers. I realize this would cost about $9 more per system but wouldn't the equal sensitivity and superior cone material be worth it?

The literature I have on the Audax 4502 specifies an F_S of 102Hz, not the 93Hz you mention in your article. Did you measure it to get the 93Hz figure?

If you like my suggestion to use the 4502 drivers throughout, what changes would you consider necessary? Thanks for your assistance.

James M. Tripp
Edmonds, WA 98020

Mr. Cox replies:

Your letter brought back memories of my own experiences shortly before the 2 x 4 article was finished. I was building another 2 x 4 driver panel and blithely wired it without checking driver polarity. The final system test was surprising, to say the least. Eventually I determined that the new shipment of ELA 4502s had been wired with a common positive lead rather than common negative as shown in the article schematic. I called Madisound and learned that a production lot had been wired incorrectly and common negative would be standard for this unit. It sounds like you have run into the same problem on other ELA models. Your excellent test method should be used to check the entire system as well as individual drivers. Madisound is an unusually helpful firm and I imagine you could arrange for replacement of the incorrectly wired units.

I agree with your comments about polypropylene's superiority to paper felt cone material, but I would not recommend using coaxial drivers exclusively. Even though tweeters in the top and bottom coaxial drivers would not be connected, they are obstructing the woofer cone's sound. At low frequencies (long wavelengths) the tweeters are acoustically invisible. As the radiated frequency increases (shorter wavelengths) the tweeters' physical dimensions become an appreciable part of the radiated sound wavelength and become an acoustical obstruction. For example, the ELA 4502 tweeter diameter is $\frac{1}{6}$ wavelength at roughly 1200Hz.

Since coaxial drivers have this disadvantage you may wonder why I didn't use separate tweeters and woofers for the 2 x 4. Using coaxial drivers where tweeters are needed permits all the space behind the driver panel to be used for TL (transmission line) loading. Both woofer and tweeter can be located on the same columnar center line. Finally, separate tweeters would have interrupted the uniform close-spacing of the woofer units I wanted to achieve cylindrical radiation pattern. Coaxial drivers are functionally effective as the column center units but would have significant functional disadvantages as top and bottom drivers.

Sensitivity and suspension differences between the Audax and ELA drivers may actually provide some advantages in the 2 x 4 application. One of my goals in speaker building is the best possible driver system to cover the musical fundamentals range (40–4kHz) without crossovers or cutoffs. At the current state of our understanding, however, we can expect some cone breakup effects when the ELA 4502s and Audax MHD10P25FSCs are allowed to run full range with no high-frequency cutoffs.

However, the ELA units with half-round foam surrounds and polypropylene cones should have significantly different cone break-

up characteristics than those of paper felt cones with butyl rubber surround. This non-coincident cone breakup should help reduce overall system effects. At the low end, to compensate for differences in fundamental resonant frequencies, I tuned individual transmission lines.

To answer your third question, I tested each driver using the method described in my article's driver testing section. The free-air resonance for these drivers ranged from 78–106Hz. Smaller drivers seem to be particularly hard to produce which operate in narrow free-air resonance limits. Specification sheet information is a general guide but measurement is a more accurate.

Good luck on your project. Let me know how it works out.

TAKING ADVANTAGE

I just received the 3/87 issue of *Speaker Builder* and this excellent magazine just keeps getting better. I stand up and cheer the 72-page issues and the newly added features. Fortunately, the cost won't increase for a couple more years; I'm taking advantage of the early renewal rates. Thanks!

Duke LeJeune
New Orleans, LA 70118

LOWTHER DIFFICULTIES

In July 1985, I contacted a manufacturing company in Great Britain, the Lowther Company, to determine whether certain products were still available. They assured me the equipment I needed was in stock and would be sent promptly on receipt of a bank draft. I ordered a Lowther PM-4 speaker and a replacement cone, and sent a bank draft for $350. The bank draft was promptly cashed, but the equipment was not sent.

I have confirmed by phone and by writing that the order and the funds were received and Lowther has repeatedly promised to send the equipment. Despite repeated inquiries they have not complied. In February 1987 they acknowledged their indebtedness yet they still have not sent the equipment, or reimbursed the money.

I am now entering suit to recover compensatory and punitive damages in a British court. I have spoken to several high fidelity dealers in the Philadelphia area, two who are familiar with the company, and one has informed me that they had exactly the same problems. I am writing primarily to warn prospective buyers in this country of Lowther products that they may expect similar problems and because of the difficulty in obtaining redress from a foreign country, if they place such an order may have a great problem in obtaining the merchandise.

Arthur J. Weiss, M.D.
Philadelphia, PA 19107

PENTAGONAL PROBLEMS

In his article "Building Pentagonal Enclosures" (*SB* 1/86, p. 28) Daniel Coyle says this type have few surfaces that will support standing waves. However, any standing wave generated will cause the entire cabinet to resonate as they will not be damped. These cabinets, then, have the potential to introduce considerable amounts of distortion. (Note *Stereophile*'s review of the Pentagram P10s, Vol. 7, No. 6). I therefore wonder whether I can brace a cabinet without any parallel sides just to make sure the few resonances that may be generated are damped. In his reply to the *Stereophile* review, the manufacturer states the P10 had in it "strategically placed internal braces" (Vol. 9, No. 4., p. 153).

Mr. Coyle also stated that two drivers will act as one below the frequency 8552/D where D is the distance between the outside edges of the drivers. Many manufacturers and designers (including Mr. D'Appolito) have not obeyed this "rule" and produced excellent designs. For those of us who do not have the back issues, how did Mr. Coyle arrive at 8552—magic numbers notwithstanding?

Arranging two woofers in the configuration Mr. Coyle suggests will not be sonically better than a vertical stack. I believe the early reflections caused by the "outside" woofer would distort the sound stage and imaging in the lower frequencies. I would suggest installing the bass drivers on the front and the midranges and tweeters on the side panels. This could broaden the sound stage. Note that higher frequencies are much more directional and hence will benefit from this design. Also the shorter wavelengths of the higher frequencies will tend to hamper distortion.

Navin M. Advani
Brooklyn, NY 11201

Mr. Coyle replies:

I deny having any interest in magic numbers. Every cabinet introduces distortion. The pentagonal shape is a means of diminishing one kind of coloration. Non-parallel faces will not support an internal standing wave. There remains the problem of energy stored in the cabinet being released after the musical impulse. More rigid walls (two layers of ¾" particle board) diminish the possibility the box

Audio Amateur Loudspeaker Projects

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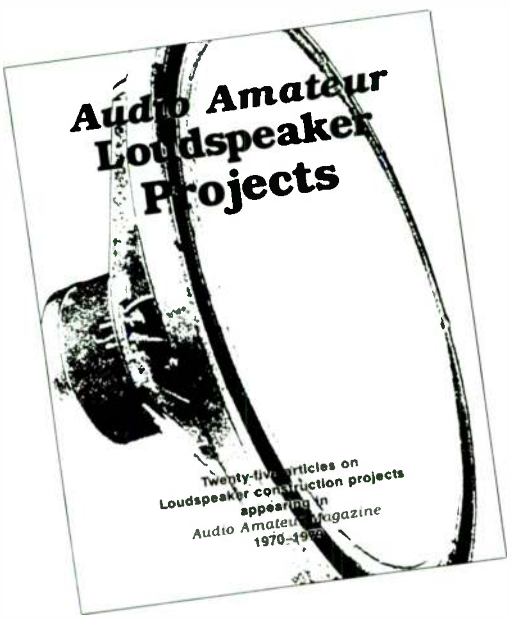

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OLD COLONY BOOKS PO BOX 243, Peterborough, NH 03458 *will flex. Damped walls (tar and sand or lead lining) absorb the energy. The best box is no box at all and for that reason, panel mounted speakers are very low in resonant coloration.*

I re-read the review of the Pentagram P-10s in Stereophile *and draw your attention to reviewer Watkinson's comment that "loose and boomy bass are problems I have noted in other passive radiator designs." All else being equal, I believe five sides offer several advantages over four, as I mentioned in the article.*

The best internal brace is a bulkhead which fits the inside the the cabinet and couples all the walls together, thereby diminishing the amplitude of resonances and raising the frequency of resonance to a place where it may be more easily damped.

The magic number 8552/speaker diameter describes the point at which Ka = 2. This is from Leo Beranek's Acoustics, *Fig. 4.12, where the directivity of the speaker is compared to frequency. In this system,* $Ka = 2\pi a/\lambda$ *(*λ *= wavelength of frequency of interest, a = speaker radius). At Ka = 2 a driver in a sealed, small frontal area enclosure is about 9dB down 90° to the side. As it is operated at higher frequencies the driver becomes more "beamy." While some beaming is probably good, too much is not. I chose Ka = 2 as an upper limit.*

Regarding your preference for driver placement, I agree that for bass drivers a vertical array is fine. The aim of the horizontal deployment was to decrease the image's vertical elongation. Olson, in Acoustical Engineering *in Fig. 2.8 gives a family of graphs that show a sound distribution pattern similar to Ka = 2 when the radius of the curved line source is less than* $\lambda/4$ *and the angle between the speakers is 120°. In the pentagon the included angle is 144°, so it may be better to use* $\lambda/5$ *as a guide to choosing the crossover frequency. For the enclosure in my article this would suggest lowering the crossover to 330Hz.*

The goal of low directivity counter-indicates using midrange or treble drivers displaced by much of an included horizontal angle. At 1kHz, $\lambda/5$ *= 2.2", so there would be a very ragged horizontal distribution of the mids and treble where positioned on even a very small cabinet's sides. I believe they are best when vertically aligned above each other.*

MIX 'N MATCH AMPS

Bob Ballard's letter, "In Praise of Active Crossovers," (*SB*, 4/86, p. 52) is an excellent summary of active systems' advantages. He notes that different amplifiers (i.e., presumably brand and topology) can be readily used together. Would he comment further on using different amplifiers together and specifically how phase and shift differences can be identified and minimized?

Mark A. Butcher
Norwood, Australia

Mr. Ballard replies:

You are correct in assuming that "different amplifiers" are those of different brand and topology.

Here are a few caveats about amplifiers to be used in each brand. All your amplifiers should be either non-inverting or inverting. They can't be mixed. Next, amplifiers used in adjacent bands must have their outputs in phase with each other, particularly at the frequency of crossover F_C*. Most audiophile quality amplifiers deviate very little from one frequency extreme to another. Most of the phase differences occur at the extreme ends of the audible band. Actually, I am unable to detect any anomalies in sound due to phase differences at the very low frequencies (30–90Hz). I attribute this mostly to the strange things that happen to low- frequency sound waves in a practical listening environment. Phases of low frequencies can be altered drastically just by someone entering the room, or if you move a piece of furniture. At the extreme high end (10k–20kHz), phase differences are again hard to detect by listening—at least, I can't. Frequencies in this range are all harmonics of the fundamental tone being played or sung, and I think phase differences in harmonics are difficult to detect. Therefore it is the midrange amplifier that is critical. It has been said by many that if you don't have smooth high quality sound in the midrange you don't have anything to build on for the overall quality sound.*

Phase differences can be identified by using an audio signal generator to feed a signal of selected frequencies to the amplifier under test in combination with an oscilloscope with separate inputs to the vertical and horizontal sweeps. The input signal should be applied to the vertical sweep and the output to the horizontal sweep. By proper adjustment of the vertical and horizontal gains a straight line trace is obtained on the scope which starts at the lower left hand corner and progresses along a 45° line to the upper right hand corner. If this condition exists, then the output is in-phase with the input. Please refer to SB, *3/82, p. 18 for an explanation of this technique in greater detail. Also in this two-part article you will find a discussion on the order of active filters which you should consider when building an active crossover or buy a kit. If you don't have any of the equipment to measure phase differences then you must rely on information from the manufacturer.*

I would recommend the Williamson 20/20 amplifier for the tweeter and the 40/40 for the midrange, and the Borbely 100W amplifier for the bass, all from Old Colony. These amplifiers used together have no phase problems. Another amp to consider for the bass is the Hafler 100W/channel. Others, even better, 200W per channel are also on the market. However, unless you want floor moving bass, the 100W categories do an excellent job. If you do elect to use 200W or more per channel, be sure your woofer driver is adequately rated.

You cannot minimize amplifier phase differences. You either live with them, get a different amplifier, or use my active crossover in SB, *3/82. Good luck and thanks for the nice things you had to say about my letter. I assume that you were or have become an active crossover advocate.*

ESL SUPPLIES

I have recently had my interest renewed in the design and construction of electrostatic loudspeakers. Of late, the do-it-yourselfer has had little new information on new developments in the ESL/DIY area. Certainly, the exceptional Roger Sanders articles (*TAA* 1975; *SB* 1980, 1982) represented the leading edge of thinking, but his letter, penned in late 1981, was the last significant update in ESL design and construction. I can't help but think others out there are still as interested as I in applying and advancing that art.

After numerous delays, TAB Books has released "Electrostatic Loudspeaker Design and Construction," by Ronald Wagner. I have not discussed the book with the author, but I hope it will be a winner. With current interest in ESL/DIY, and with the interest the book should generate, it occurs to me that builders would benefit from a sort of clearing house for ESL parts and information. I have already done some legwork for acquiring the parts to build my own variation of the Sanders design and I have been in contact with Mr. Sanders for advice, but he no longer supplies any of the necessary parts for construction.

I would be very interested to hear from other readers of any interest in my proposal for a network, or clearinghouse for ESL enthusiasts and I would even entertain the idea of coordinating information and parts purchase efforts.

Neil Shattles
Lilburn, GA 30247

Mr. Sanders reviews the Wagner book in this issue.—Ed.

SPICA-TC50

Most of my speaker building time is spent researching and studying loudspeaker designs. One that has amazed me is the Spica-TC50. Its simple two-way design, using an Audax HIF17 series woofer and an Audax HD100 series tweeter, has a depth of image some planar loudspeakers just can't match.

The drivers are inexpensive, about $30. The cabinet is a wedge shaped, sealed box. From what I have learned, the secret of this speaker must be in the crossover. It uses a Bessell summing low-pass with a Butterworth high-pass function and uses the wedge shape of the cabinet as a phase delay.

Every review I have read has been excellent on this simple loudspeaker, and from my listening experience, I agree. Peter Moncrieff of *International Audio Review* has written a few long, admiring articles about this speaker.

Perhaps *SB* could look into the crossover and the speaker as a whole. I am sure much of this crossover technology would help improve many home built designs. I believe Mr. Kantor's statements about crossover design (*SB* 4/86, p. 20) hit the nail on the head.

I am a new subscriber starting with the 1986 issues, but I have been involved in speaker building for about 10 years. I really enjoy the magazine and find it a refreshing change from the *Journal* of the Audio Engineering Society.

J.B.S.
Pittsburgh, PA 15228

STRATHEARN RETURN

In light of the recent interest in ribbon drivers (*SB* 4/86, p. 14) perhaps the Strathearn (now Stratec) ribbon deserves another look.

A frequency response graph of an unmodified Strathearn was presented by Mark McKenzie using FFT techniques in his article (*SB* 3/85, p. 22). After corresponding with Mr. McKenzie following my letter (*SB* 4/85, p. 54), it became apparent that about 4dB of the irregularities in his measurements could indeed be attributed to uncompensated diffraction effects (*Fig. 1*). The unenclosed rear radiation may also have affected the response, particularly after the authors removed the faceplate and dampers.

However, a letter by Dr. Eugene Zaustinsky points out that Mr. McKenzie's measurements have been made at a distance of only one meter, which could produce drastic response irregularities at high frequencies with a 20″ line-source driver such as the Strathearn. I was able to verify this (*Fig. 2*) with the formulas I had sent Mr. McKenzie in my original letter. With these formulas I had intended to indicate the effect of stacking a pair of Strathearns at a normal listening distance.

In conclusion, further measurements are required before the unmodified Stratec/Strathearn may be judged. This is not meant to detract in any way from the interesting ideas and methods presented in Mr. McKenzie's article.

Ralph Gonzalez
Philadelphia, PA 19143

ROUND ELECTROSTATICS

In reply to James Rice's question regarding round electrostatics *SB*, 4/85 p. 58, Hans Vogt submitted an article in *Wireless World* May 29, 1929, pp. 553-556. If Mr. Rice wishes a copy of that article I can supply it.

Alan Smith
Dunedin, New Zealand

LOWER RESPONSE AND EQUALIZATION

In recent years it has become increasingly common to attain low level frequency response from small, totally enclosed boxes by electrical equalization. Suppose I place transient response above all other design considerations and insist on a critically damped system (Q_{TC} = 0.05). Given the limitations of available drivers, I then find that I must either have an enormous box or allow f_3 to shoot upward, sacrificing low frequency response. I choose the latter, and equalize the amplifier to compensate.

Is this procedure self-defeating? In boosting the low frequency response, have I not also raised the Q of the system above the design goal? On one hand, it is hard to understand how electrical pre-emphasis can alter the damping of the mechanical and acoustical meshes. On the other hand it would seem that minimum phase shift networks having the same overall steady state transfer function must have the same transient response, regardless of how or where that transfer function is achieved. I am unable to think my way out of this problem.

Also, I would be grateful if someone could direct me to a treatment of the radiation load on a pulsating cylinder. It is not found in Olson or Beranek, yet it seems many wall-mounted or corner-mounted column arrays approach this model more closely than the conventional piston model.

Edward A. Fagen
Newark, DE 19711

Joseph D'Appolito responds:

We must first distinguish between the frequency response and the power response of a loudspeaker. Frequency response is measured at a low power level (generally 1W) where loudspeaker response is linear. Frequency response is often called "small signal" response. It is plotted on dB versus frequency axes relative to an arbitrary 0dB reference level, often selected at 1kHz.

Power response is a measure of the maximum acoustic output a speaker will produce at its maximum allowable input power. At higher frequencies maximum input power is limited by maximum voice coil throw. Power response is often referred to as "large signal" response and represents the input power level above which driver operation becomes nonlinear. Power response is displayed on a plot of dB SPL versus frequency. The maximum allowable input power that produces that sound pressure level may also be plotted.

Simply put, electronic equalization will alter the frequency response of a loudspeaker, but it cannot affect the power response which is solely a function of the driver. As long as driver operation is linear, the acoustic responses of a given driver in a large box and that same driver in a smaller box, equalized to produce the same overall transfer function, will be indistinguishable.

These points are made clearer by examining a specific example. Table 1 lists the parameters for a typical 8" driver. This driver has an F_{sa} *of 32Hz, a* Q_t *of 0.35, a* V_{as} *of 60 liters and a maximum thermal power rating of 200W.* Table 2 *lists the frequency response, maximum allowable input power and the maximum SPL in dB generated by that input power level using a box volume of 60 liters, corresponding to a driver/box system Q of 0.5.* Table 3 *lists the same quantities for a 12 liter system having a Q of 0.86; I shall call these System 1 and System 2, respectively. (The results in* Tables 2 *and 3 were generated with a program similar to BOXRESPONSE.)*

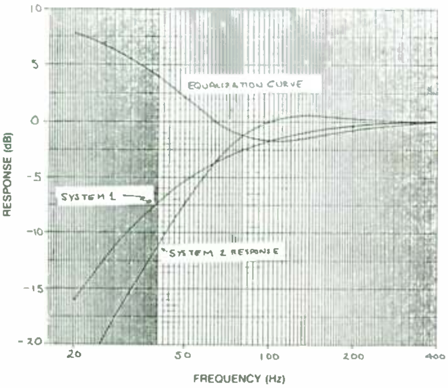

FIGURE 1: Frequency response curve.

Frequency response for the two box sizes are plotted in Fig. 1. *The response rolloff for System 1 is very smooth and slow, as one would expect from a system with a Q of 0.5 The response of System 2 is slightly peaked and falls off more rapidly, especially below 64Hz. To make the frequency response of System 2 look like System 1 we must apply electronic equalization to System 2, equal to the difference between the two curves. This equalization curve is also plotted in* Fig. 1. *This equalization does not alter the Q of System 2. What it does do is reduce the input to the speaker in the region of peaking by supplying a corresponding dip that eliminates the effect of the underdamped Q.*

What happens to power response? The power response and the maximum allowable input power that first produces that SPL are plotted for both systems in Fig. 2. *First notice the maximum SPL output capability of both systems is identical. As stated earlier, maximum SPL is only a function of driver power rating, cone area and maximum allowable throw; and must be the same for both systems.*

The input power that produces the maximum SPL is box dependent. Above 200Hz, input power is limited in both systems by the voice coil thermal power rating of 200W. Below 200Hz, power is limited by the allowable voice coil excursion of 4mm to less than 200W and consequently the maximum SPL falls off at 12dB/octave.

Between 64 and 125Hz, the allowable input power of System 2 is somewhat less than that of System 1. Below 64Hz System 2 will accept increasingly more power than System 1. This is due to the stiffer spring constant of the air mass in the smaller box which provides greater mechanical opposition to cone motion.

At even lower frequencies the peak power curves flatten out as both systems transition from mass controlled to stiffness controlled response. At these very low frequencies, opposition to cone motion comes solely from the spring stiffness of the driver/box combination. Since the smaller box supplies more stiffness, System 2 accepts more power than System 1 at low frequencies. Finally, notice that the power rating trend in the throw limited region is just the inverse of the frequency response differences.

Now, what is the effect of electronic equalization? Let's look at two frequencies. Take 32Hz first. From Table 2 *the difference in frequency response of the two systems is 5.55dB. Therefore we must apply 5.55dB of electronic boost to System 2 to reach the output level of System 1. The amount of boost represents a factor of 3.59 increase in input power. System 1 will accept an input power of 5.89W at 32Hz before its throw limit is exceeded. If we increase this power level by factor of 3.59 we get 21.13W. Amazing. This is exactly the maximum input power System 2 will accept at 32Hz to produce the same SPL.*

Now look at 100Hz. Here System 1 is down relative to System 2 by 1.71dB. Thus System 2 input power must be reduced by a factor of 1.49 to produce the same frequency response as System 1. System 1 will accept a power input of 88.85W at 100Hz. Dividing this power level by 1.49 results in an input power of 58.86W, which is the power limit on System 2 at the same frequency!

We see that the maximun power rating of System 2 increases or decreases in just the right amount to produce the same frequency response and power response (i.e. maximum SPL) as the unequalized system. In other words, System 1 and 2 are both small signal and large signal equivalent!

If the two systems are equivalent, why equalize? Well, the obvious answer is to reduce the box size. There is a less obvious reason.

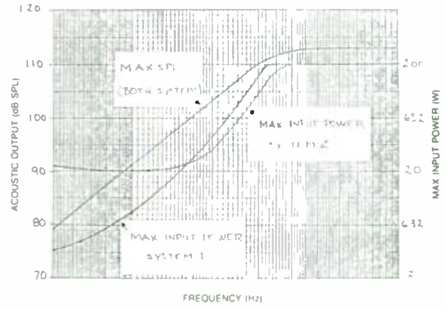

FIGURE 2: Power response curves.

Closed-box drivers have floppy, nonlinear suspensions. They rely on the much greater stiffness of the box air spring to linearize cone motion. The use of a smaller box will generally result in less low-frequency distortion. But isn't there a power penalty? Not really. If both systems are operated within their limits, midband power requirements will control amplifier sizing.

Note: we have examined only the case where speaker response is equalized to just offset the effect of reduced box size. This is the condition of Mr. Fagen's question. It is common practice to add greater levels of equalization than this to small boxes. In this case the speaker will be over-driven causing much higher distortion and degradation in transient response.

Regarding Mr. Fagen's question on column speakers, they are well modeled by line sources.

Mr. Bullock replies:

Any low-pass filtering of a signal affects its transient response negatively. Whether it occurs in the electrical, the mechanical or the acoustical state, or a combination thereof,

TABLE 1

EXAMPLE 8″ DRIVER PARAMETERS

Free air resonance	32Hz
Total Q (Qt)	0.35
Equivalent volume (Vas)	60 liters
Peak power	200W
Max. throw (Xmax)	4mm
Cone area	220 cm2

TABLE 2

8″ DRIVER RESPONSE IN 60 LITER BOX

FREQ. (Hz)	FREQ. RESP. (dB)	SPL (dB)*	POWER (W)**
10.00	−26.81	67.19	2.91
12.50	−23.15	71.06	3.06
16.00	−19.24	75.35	3.34
20.00	−15.88	79.23	3.77
25.00	−12.76	83.10	4.48
32.00	− 9.66	87.39	5.89
40.00	− 7.26	91.27	8.28
50.00	− 5.28	95.15	12.81
64.00	− 3.58	99.43	23.27
80.00	− 2.46	103.31	43.82
100.00	− 1.65	107.19	88.85
125.00	− 1.09	111.06	190.74
160.00	− 0.68	111.68	200.00
200.00	− 0.44	111.92	200.00
250.00	− 0.29	112.08	200.00
320.00	− 0.18	112.19	200.00
400.00	− 0.11	112.25	200.00

* Max Throw Limited SPL
** Throw Limited Power

TABLE 3

8″ DRIVER RESPONSE IN 12 LITER BOX

FREQ. (Hz)	FREQ. RESP. (dB)	SPL (dB)*	POWER (W)**
10.00	−35.72	67.19	22.70
12.50	−31.82	71.06	22.58
16.00	−27.49	75.35	22.36
20.00	−23.56	79.23	22.08
25.00	−19.61	83.10	21.68
32.00	−15.21	87.39	21.13
40.00	−11.24	91.27	20.68
50.00	− 7.38	95.15	20.77
64.00	− 3.60	99.43	23.37
80.00	− 1.17	103.31	32.57
100.00	0.06	107.19	59.86
125.00	0.44	111.06	134.03
160.00	0.44	112.80	200.00
200.00	0.34	112.70	200.00
250.00	0.24	112.60	200.00
320.00	0.15	112.51	200.00
400.00	0.10	112.46	200.00

* Max Throw Limited SPL
** Throw Limited Power

makes no difference. Using a high-pass filter to raise the low frequency response of any type of loudspeaker is going to degrade transient response. A closed box equalized with a second-order high-pass filter will exhibit a transient response similar to that of a vented box.

Theoretical Acoustics, *by Morse and Ingard, covers radiation from a cylinder, but I believe the results are too complicated for convenient use.*

KEF'S NEW BASS LOADING

Anyone interested in experimenting with the bass loading technique recently introduced by KEF where the woofer is contained *within* the enclosure and the only output is that of the vent, will find a description in "A Bandpass Loudspeaker Enclosure," L.R. Fincham, KEF Electronics Limited. Presented at the 63rd convention of the Audio Engineering Society, 1979, the preprint may be obtained from the Audio Engineering Society, Office of Special Publications, 60 E. 42nd St., NY, NY 10017.

This technique has the advantages of (1) high efficiency, and (2) no need for a passive low-pass crossover since the enclosure produces a band-pass effect acoustically. A subwoofer using this approach could have great potential.

Ralph Gonzalez
Philadelphia, PA 19143

A construction article using this technique is in preparation for SB.—Ed.

SHIELDING TV SPEAKERS

A *SB* reader has asked if anyone knows about shielding speakers for use next to a television set. As I'm planning to build a set of TV speakers, I've been anxiously waiting for a reply.

Robert L. Friend
Beverly Hills, CA 90211

Your speakers will cause problems if they are located within 5-6 inches of the TV set's sides. If they are located correctly for stereo separation (i.e. a minimum of 6 ft.) then the speakers' magnets will not affect the color mix of your TV picture. If you must have the speakers close to the set, line the inside of the cabinets with sheet steel.—Ed.

FILTER CALCULATIONS

continued from page 20

checked on a simple one-octave analyzer built into one of my equalizers, just to make sure they were functioning properly before going into service. So far, no anomalous results have cropped up.

As a final note, I limited the scope of this article to even-order filters not because I advocate them especially over odd-order filters, but because I used B4 Linkwitz-Riley crossovers.

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2. Hoenig, Stuart A., *How to Build and Use Electronic Devices Without Frustration, Panic, Mountains of Money, or an Engineering Degree*, 2nd Ed. Boston: Little, Brown & Co., 1980.
3. Linkwitz, S.H., "Active Crossover Networks for Noncoincident Drivers," *JAES*, Vol. 24, Jan./Feb. 1976, p. 2.

Also suggested reading

Jung, Walter G., *Audio IC Op Amp Applications* 3rd Ed., Howard W. Sams, 1986.

Lancaster,. Don, *Active-Filter Cookbook*, Howard W. Sams, 1975.

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TRADE

FALCON ACOUSTICS is one of the largest independent manufacturers of crossover networks and speaker system accessories in the UK, supplying manufacturers, retailers and export. We have outlets in both the USA and Canada. Your guarantee is the trust our manufacturing customers put in the quality and reliability of our networks. **Inductors:** Ferrite (up to 200W), air-cored and transformer; **Capacitors:** Reversible electrolytic 50V/100V, polyester, polycarbonate and polypropylene; **Networks:** choice of over 70 for different unit combinations; **Components & Accessories:** most except for the wood! If you have supply problems, please send for details, $2 air, $1 surface, to our mail order retail outlet. ***FALCON ELECTRONICS***, Tabor House, Norwich Rd., Mulbarton, Norfolk, England NR14 8JT. TTF

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FOR SALE

3D Acoustics 3-piece speaker system, walnut, $175; Belles DMC preamp, $175. Both low use/original cartons. *Audio Amateur* 70, 74 and 77, best offer. All plus shipping. Rafael Lopez, 3340 NW 4 St., Miami, FL 33125.

One pair Snell type, excellent condition, $1200. Thomas Dussault, (201) 906-8289 evenings, (609) 924-0599 days.

Dynaco PAT-5 preamp, unmodified, champagne front panel, good condition, asking $65 ($95 Canadian); (2) Cerwin-Vega speaker cabinets, model CDN, holes for 12″ woofer, midrange, tweeter, and port, asking $30 for pair. Call or write Greg Nawrocki, 21 Indiana St., Kitchener, Ontario N2H 2A4, Canada, (519) 745-1579.

Several pairs Focal 7N501 woofers (as used in Gold Ribbon System in *SB* 4/86), $60/pair unused, $50/pair used. Used, lonely Jordan Module seeks a good home, $20. Duke (504) 866-7381.

Nine (9) pairs of Jordan 50mm Modules. Seven pairs neither used or mounted. Project for which they were purchased never got under way. Paid $1450 for them. Will sell for $450. also two vertical array cabinets. Stan Foote, 11A Meriam St., Wakefield, MA 01880, (617) 246-4106 days.

Drivers. Like new, used only for testing. Dynaudio D-28AF, $60/pair; Dynaudio 17W75, $80/pair; (2 pair) Peerless TP165RDC (cast frame, 30 oz. magnet version of TP165R), $65/pair; KEF B110 used, $60/pair. Joe D'Appolito, 9 W. Knoll Rd., Andover, MA 01810, (617) 470-1346 evenings.

Linn Sondek LP 12 (pre-Valhalla) with SME board and Grace 707 adapter, $300; Grace 707, $50. Buyer pays shipping. Call Rion Dudley, (206) 285-7918.

Heath IG-1275 linear/log sweep generator, like new, $175; Heath IR-5204 strip chart recorder, with extra paper and pens, brand new, $200. Both for $350. Bill Hoffman, 1555 S. Virginia St., Reno, NV 89502, (702) 329-6597 days.

Magneplanar Timpani 1C, new white grille cloth, custom stands, $800; TEAC 3340S, 4-track, 4-channel, 10½″, 7½″ and 15 ips, $650; CM Labs Model 601 electronic crossover, $30. Bruce Beyer, 46998 Bartlett Dr., Canton, MI 48187, (313) 455-1728.

Pair Dynaudio 21W-54 subwoofers in eight-foot modified Bailey transmission lines, 25Hz response @160W, trimmed in solid oak, $495/pair. Bargain prices on raw drivers from Focal, Peerless, Scanspeak, Polydax and Panasonic. Call or write for details and prices. Bob Pickard, 1720 Lilac Dr., Walnut Creek, CA 94595, (415) 930-9104.

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JBL 15″ woofer model 2235H (8Ω, 300W, 4″ voice coil, SPL 93dB/W/M, FS 20Hz, Vas 16 cu. ft., Qt 25. In original box with warranty. List $219, sell for $125 includes shipping and COD costs in US. Field Mead, 14275 Sherwood Rd. NW, Seattle, WA 98177, (206) 363-4657.

Hitachi LC Intercon 1M, $25, 2M, $40; Sola Power conditioners, 60W, 250W, 1000W, 2000W; 5% metal film resistors, ½W, 25¢, 1W, 35¢, 2W, 50¢; MK II minus power transformer, ($?). WANTED: Working pair of Ionovacs with power supplies GSI mod for Lux MQ68C. Alan (201) 887-9426, leave message on machine.

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Tuning dial for Dynaco FM-3 tuner, original manual for Dynaco PAS-3X preamp. G. Embler, 732 College Ave., Palo Alto, CA 94306, (415) 493-1980 days, (415) 424-1483 evenings.

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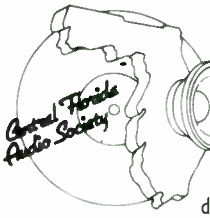

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