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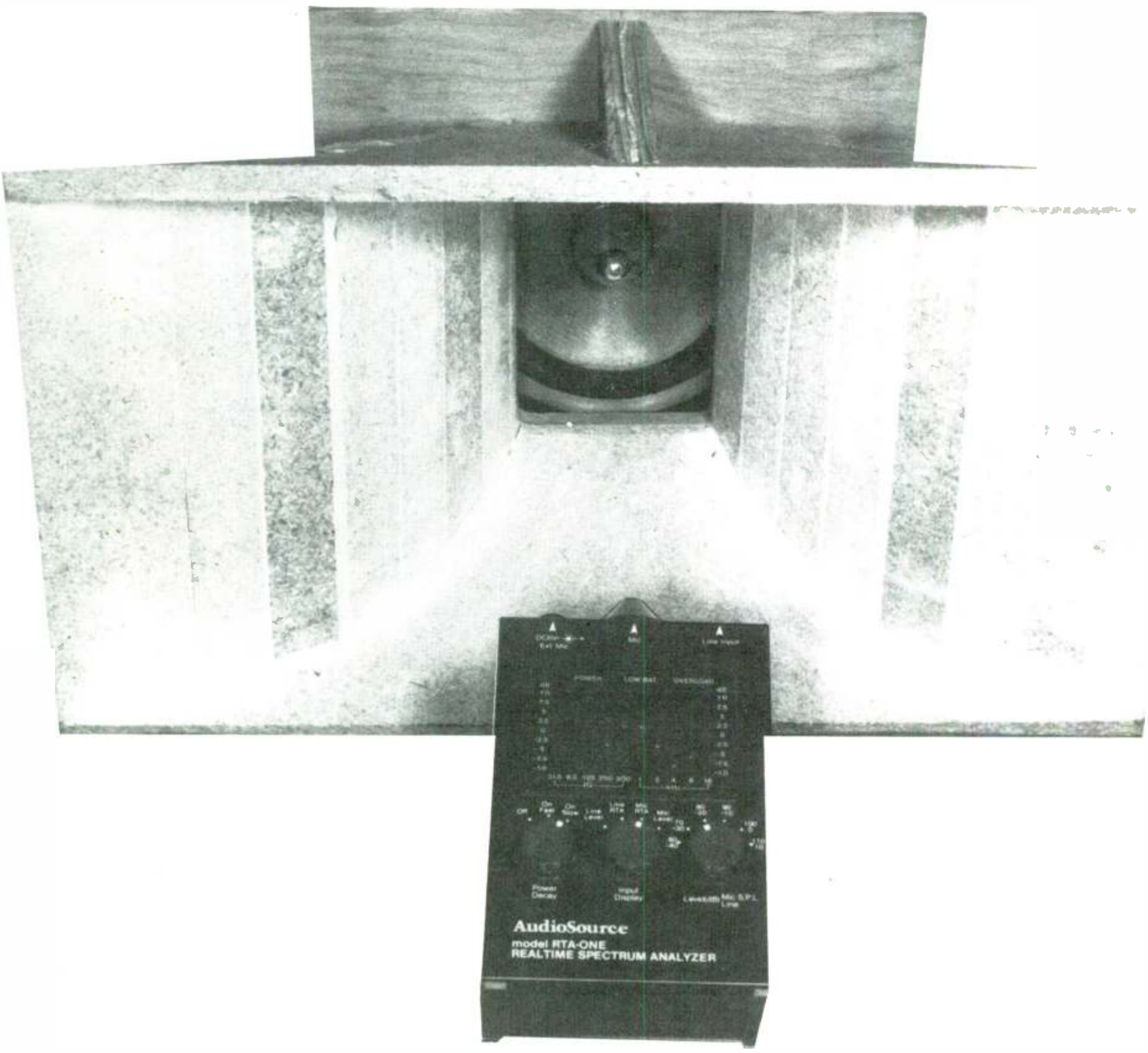
SPEAKER BUILDER



EDGAR'S MIDRANGE HORN

Good News

BOSTON ACOUSTICS announces two new additions to their automotive speaker line: the 705 and the 780LF.

The 705 is a 5½-inch double cone speaker that is said to handle 20W per channel with peaks up to 40W, and it is designed to fit all standard 5½-inch factory cutouts. The frequency response of the 705 is 58Hz–1.7kHz, ±3dB, and this 4Ω system produces 93dB at 0.5 meters and 1W input.

The 780LF eight-inch subwoofer is designed to extend the bass of any full-range car speaker system. It has a 4Ω sensitivity of 93dB at 0.5 meters and handles 75W continuous, 150W peak. The woofer cones are made of copolymer material that can withstand the heat, cold and humidity of an automotive environment and are completely waterproof.

For further information contact John C. Chermesino, Boston Acoustics, 247 Lynnfield St., Peabody, MA 01960.

Fast Reply #FI336

MADISOUND Speaker Components announces that their new Audio Projects bulletin board service is now operating: call 608 767-2585 from 11:00 p.m. to 8:00 a.m. CST, at 300 and 1200 baud, no parity, 8 bit word, and 1 stop bit.

The bulletin board is designed to exchange information and actively discuss subjects pertaining to amateur audio hobbyists and computer hobbyists. The board is open to all, and Madisound would like to encourage uploading and downloading of software and articles pertaining to audio speaker building and crossover design.

The FIDO software is said to be easy to use, and supports most modem protocols including xmodem, xmodem crc, modem7 batch and kermit. The system also provides on-line help.

For more information contact Madisound Speaker Components, 8982 Table Bluff Rd., Box 4283, Madison, WI 53711.

Fast Reply #FI20

AUDIO-TECHNICA has added two dynamic moving coil stereophones to its "Dynamic Capability" series, the ATH-M7 PRO and the ATH-18.

The ATH-M7 PRO is said to have extended frequency response and low distortion plus a design that features a self-supporting voice coil, a low-mass high-compliance diaphragm and a tangential surround for linear diaphragm motion. A soft semi-circumaural pad on each earpiece provides a seal against ambient noise without exerting undue pressure. Audio-Technica claims that frequency response is flat over 20–20kHz on both models. The 3.5 ounce ATH-18 features samarium magnets said to provide highly efficient, balanced sound.

A-T's term "Dynamic Capability" is one they use to describe a compatibility with various types of playback devices such as personal or portable cassette/radio players, home phono systems and compact disk systems.

For further information contact Audio-Technica U.S., Inc., 1221 Commerce Dr., Stow, OH 44334.

Fast Reply #FI22

ACOUSTIC SCIENCES has developed the Tube Trap: a modular line of small, efficient and patented bass traps. The traps can be used alone, stacked into columns or installed on walls or ceilings.

The Tube Trap is a pressure zone trap, best utilized when backloaded by a wall or corner. Tri-corner installations in rectangular rooms are said to effectively damp all major room resonant modes. Two models are in production: the M-90, a voice range trap that provides 10 sabines per three foot section absorption, and the M-45, a trap that ranges an octave below to 45Hz, providing 15 sabines per section.

Tube Trap claims to clean up transient distortions due to room acoustics. Low-end phase shifting from a tone burst is reduced by trapping the first, strong reflection. The result is a faster and more detailed low-end acoustic room.

Both models have a "midrange crossover diffusor panel" section to keep sound field balance even when listening in the nearfield.

For more information contact Acoustic Sciences Corp., PO Box 11156, Eugene, OR 97440, or call 1-800-ASC-TUBE.

Fast Reply #FI280

OVONIC Synthetic Materials Company has developed a "super" magnet, said to provide the highest performance per dollar of any high strength magnet currently available. The company claims it has up to ten times the power of Ferrite magnets and is therefore a commercial manufacturers' dream. Beside the obvious benefits this could have in loudspeaker manufacture, the magnet has wide applications in the electric motor, automobile and computer industries.

For more information contact Energy Conversion Devices, Inc., 1675 West Maple Rd., Troy, MI 48084.

Fast Reply #FI64

INFINITY SYSTEMS Inc. introduces their new RS loudspeaker line: a three-way, the RS4000, and three two-way units, the 8-inch RS3000, 6½-inch RS2000 and a 4½-inch RS1000. These new models are said to offer quality performance in attractively styled minimum diffraction enclosures with improved components. The RS Series replaces the current RS models 7b, 8b, 9b, 10b and 11b.

The RS4000 has an EMIT® tweeter, and the entire series uses new crossover networks with improved quality components. The two-way units use Polycell® tweeters, that claim a smooth, clean response with minimum distortion, and varying size polypropylene woofers for deep extended bass.

The enclosures are in contemporary style with smooth rounded edges, minimum diffraction floating grilles and finished in a vinyl simulated oak veneer.

For more information contact Infinity Systems Inc., 9409 Owensworth Ave., Chatsworth, CA 91311.

Fast Reply #FI354

Just out from **KLH** is their Model 862. The 862 designation describes an 8-sided enclosure using a 6-inch polypropylene woofer in a 2-way system. A ferrofluid-cooled soft dome tweeter handles the highs and is crossed over at 2.5kHz.

The unusual eight-sided shape is said to enhance sound and ease room placement. The three-sided back is designed to cancel rear reflections from the drivers while providing placement convenient to wall/floor, wall/ceiling and inside corners. The enclosure is finished in black ash vinyl veneer.

For more information write the KLH division of Kyocera International, Inc., 7 Powder Horn Dr., Warren, NJ 07060.

Fast Reply #FI58

SPEAKER BUILDER
MAGAZINE

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A Note To Contributors

We welcome contributions for possible publication in the form of manuscripts, photographs or drawings, and will be glad to consider them for publication. Please enclose a stamped, addressed return envelope with each submission. While we cannot accept responsibility for loss or damage, all material will be handled with care while in our possession. Receipt of material is acknowledged by postcard. Payment is on publication.

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About This Issue

Bruce Edgar's long awaited midrange horn appears this time (p. 7) after a long, herculean effort. Horn theory has been largely neglected in this country, abandoned after the introduction of stereo and the rise of the popular closed box systems. Edgar's contribution is significant because of his horn's smooth response, its relatively small size and efficiency.

Bob Bullock's fourth article in his comprehensive and outstanding crossover series begins on page 20.

Dan Coyle's guidelines for constructing five-sided enclosures (p. 28) gives speaker builders a brand new format to investigate. **Joseph Kmetz** shows us how to build sonically inert stands for the smaller speaker formats which work best at ear level (p. 18).

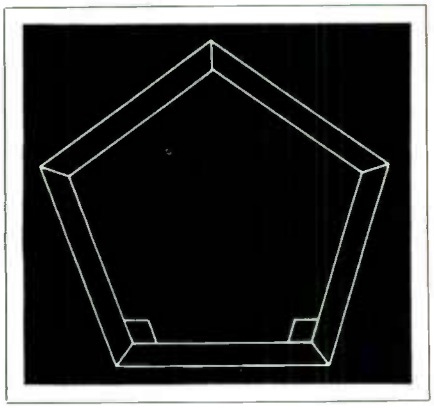
Dave Davenport (p. 30) reviews Audio Concepts' Model G system and evaluates its performance while **Dick Marsh**, from his own special angle, upgrades some of its features and gives his opinion of its sonic character.

We apologize to our subscribers for difficulties many have experienced with their subscriptions in past months. If you are still having a problem, please document it fully and our new staff will make adjustments for you.

We regret the delay in delivery of this first issue and thank all of you for your patience. We will do our best to catch up in the coming months.

SPEAKER BUILDER

VOLUME 7 NUMBER 1 APRIL 1986



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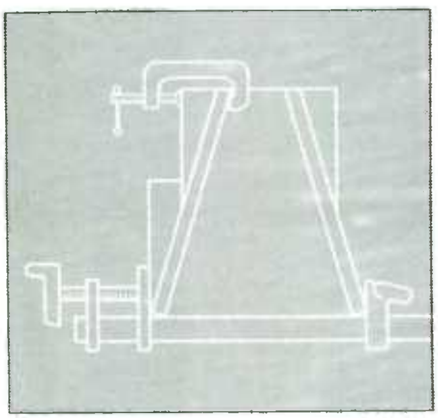

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FEATURES

DEPARTMENTS

Editorial

Hats and Hopes

An obvious newcomer to Boston, strolling in the Public Garden, expressed admiration for her friend's hat, asking where she had bought it.

Her friend replied, "My dear, in Boston we do not buy our hats, we *have* our hats." Audiophiles are often like Boston ladies, they *have* their equipment—and especially loudspeakers.

I am very pleased to report that you speaker builders are a very different breed. You are, by all evidence I can see, enthusiastically exploring the rich, exciting bypaths of loudspeaker technology. You are eagerly gathering the new information, coming now in a flood tide from manufacturers, on the steadily evolving and improving array of drivers and kits which are becoming available.

Our advertisers report that you are asking for information at a remarkable rate and that you are buying hardware to put into your own cabinetry at higher and higher levels.

The excitement really flows from the new sense of discovery and confidence I see in the articles we have been privileged to publish. The effect of the work of our dedicated band of authors and contributing editors has begun to make *Speaker Builder* a world class journal and a unique source of reliable information about loudspeaker behavior.

The other vital factor is your willingness, as readers, to react to authors with excellent, provocative questions and their consequent investment of their time and intelligence in discussing those questions in our *Letters* column. You will notice very lively interchanges in our columns this time, and none livelier than those replies from Joseph D'Appolito. I take great pleasure in welcoming him to the staff as one of our five Contributing Editors. In a conversation the other day he remarked that he had moved from actively building other kinds of audio hardware to loudspeakers just because the improvements were so much more evident and the challenges more interesting.

You are also giving the driver vendors feedback in quality and quantity unlike any they have hitherto enjoyed. Doubtless we have seen a shakeout in vendors, and some are more responsive than others. But your use of our "Fast Reply" cards produced over 4,000 requests for information during January of 1986 alone. And the relationship does not end with a literature request or even an order. Your field reports on performance, driver compatibility, reliability and ease of use are being relayed back to the manufacturers. Even though most of the popular devices are made in France, West Germany, Denmark, Sweden and the United Kingdom, the makers appear to be exceptionally responsive to what you are telling them via their dealers and distributors.

This field has attracted some formidably well-equipped devotees. And all of us who love good sound are benefitting from their investment. And the excitement about speaker matters is not confined to this magazine. The *Journal* of the Audio Engineering Society has, for the last two years, been heavily devoted to papers on loudspeakers. And there is still a lot of frontier land left unmapped in this world of loudspeakers. You are exploring the more and more refined questions and possibilities of loudspeaker technology.

Thus you are, collectively, building a solid foundation for a rewarding and exciting avocation. We are planning to do our best to spread the good news of what is transpiring as far and as wide as possible.

You can help even more than you are by spreading the news as well. "The more the merrier" is an old, worn aphorism, but it is certainly true of this avocation. The growth of the group of genuinely active speaker builders will work for a healthier range of product offerings, and a growing body of knowledge about this endeavor to reproduce sound realistically.

It is a great pleasure for us to watch all this happen and to play our part in providing the forum for this interchange. And I believe we are only on the threshold of something that will get better as it grows.—The Editor

THE EDGAR MIDRANGE HORN

BY BRUCE C. EDGAR
Contributing Editor
Photos by Manfred Buechler

> *"It remains hard to convince people, including acoustical engineers, that the midrange is where we live, and it is in the midrange that distortion is the most annoying and where amplitude response errors are most prominently evident. I have spent more man-hours of R & D time on the midrange than on the bass."*[1]
>
> *Paul Klipsch, 1971*

When I promised a midrange horn to go along with my 70Hz mini horn design (*SB* 2/83 p. 7), little did I realize how true Klipsch's words would be. In 1982 I had a midrange horn operating, but colorations in the sound quality were too objectionable to present to *SB* readers.

The colorations came from three main sources: driver limitations, throat and mouth configurations, and construction techniques and materials. In the process of researching, identifying and correcting each source of coloration, I found I was following the same path as many horn builders had followed over the last 60 years.

HORN HISTORY. When the radio boom hit the general public in the 1920's, a general need for sound amplification devices developed so that people weren't tied down with earphones. Since amplifiers were practically nonexistent, enterprising experimenters found that earphones could be fitted to phonograph horns to give the needed sound level in a room.

In one article[2] a pair of earphones was mounted in front of a wooden chopping bowl to give acoustic reinforcement. The "amplification" came from the resonant peaks in the horn response or, in the case of the chopping bowl, from cavity resonances. However, a horn does not "amplify"; it can only provide an efficient means for coupling the acoustic output from a small radiator to a free space.

The use of an earphone with a small permanent magnet only compounded the effects of inherent resonances in the old phonograph horns. Very often the use of a driver with a weak magnet introduces a nasty resonance at the lower end of the spectrum, even on an exponential horn. For a midrange horn with a resonance around 400-500Hz, such a peak gives a "horn like" nasal

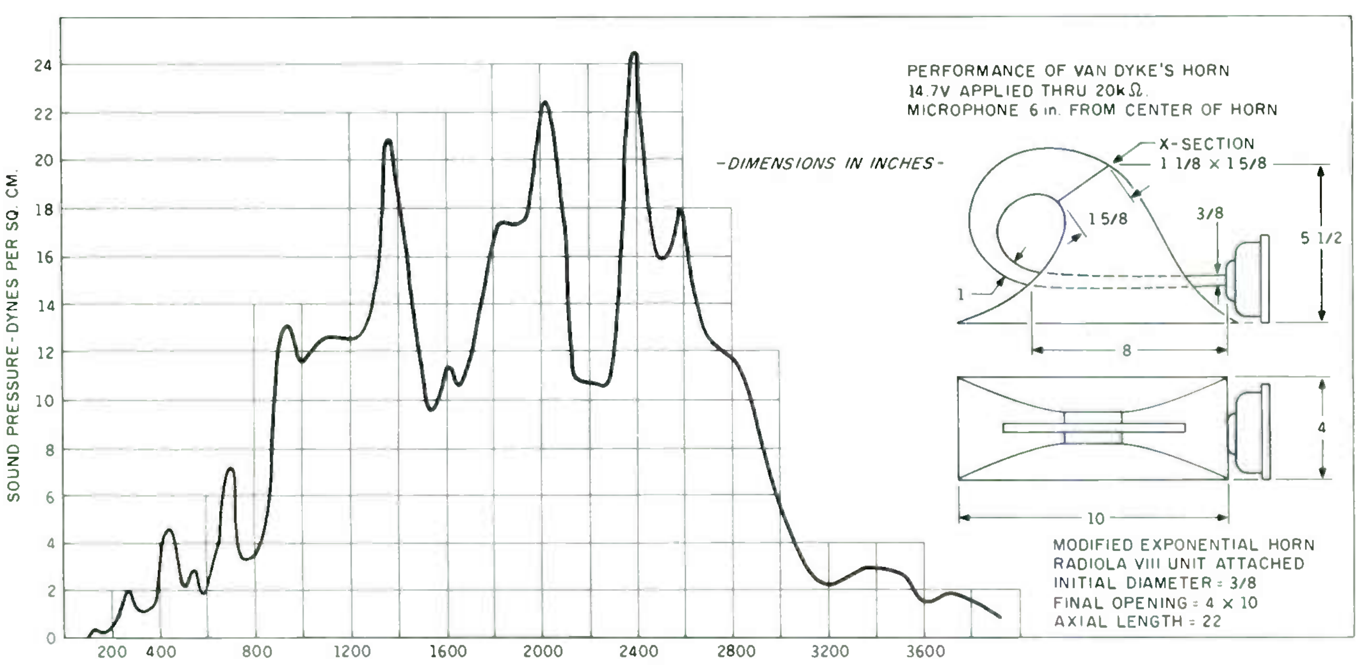

FIGURE 1a: The response of an experimental exponential horn (circa 1924) as measured by an RCA engineer. Notice the prominent response peaks. *Courtesy of the Smithsonian.*

sound quality that will make audiophiles shudder.

To give the reader a view of how those old horns sounded, *Fig. 1a* and *1b* are plots of the response of two sample horns as measured by an unknown RCA engineer in 1923 and 1924. I found these response graphs in the Smithsonian Museum archives through the aid of electro-acoustic devices curator Mr. Ed Sivowitch.

Figure 1a shows the response of a folded exponential horn with a series of three response peaks (peak to trough ratio of approximately 6dB) typical of an exponential horn with a too small mouth. In fact, from the given dimensions, the flare rate is 125Hz. The optimum mouth area should be 928 square inches; whereas the mouth area in this case is only 40 square inches. The Radiola driver probably has an effective response from only 1 to 3kHz, which implies that the designer failed to optimize the horn parameters to the driver capabilities.

The horn in *Fig. 1b* shows a decent response curve with a ripple of 3dB or less, ignoring the peak at the bottom of the spectrum. The initial peak is probably due to the effects of throat reactance and a weak magnet in the driver. Even though the horn in *Fig. 1b* has a much smoother response than the *Fig. 1a* horn, the sound quality would still be "tinny" due to the absence of frequencies below 500Hz.

It makes one wonder when you read the advertising hype of the period that told of the "wonderful realistic sound" that came from such horns. Of course, when wide range dynamic loudspeakers became available in the late 20s, the public quickly converted over to the new dynamics. The old gooseneck horns of that period are now prized as collector's items.

The reader may well ask, "Why look at these old response graphs made with antiquated measuring gear of dubious calibration standards?" The answer is horn acoustics have not changed in 60 years, and many of the same problems designers encountered in the 20s are still faced by horn builders today.

MID HORN SPECS. As I pointed out in my mini horn article, most bass horns have a mass cutoff in the 300-500Hz region. So a properly designed midrange horn should go down to a least 400Hz in order to mate with a bass horn.

A quick survey of commercial midrange horns available to the constructor shows that most have cutoffs of 600-800Hz. The reason for the higher cutoff frequencies is the market for midrange horns lies primarily with direct radiator woofers and not with bass horns. But if you have a bass horn, you still have a problem if you try to use one of these midrange horns due to a significant hole in the response around 500Hz. Some professional horns, i.e., JBL, with two inch diameter throats will go down to 500Hz, but they are expensive and really do not belong in a home environment.

My solution to the problem uses a midrange cone driver on a wooden midrange horn. This approach has been tried by several authors with varying success.[3,4,5]

The problems with this approach are manifold, as pointed out to me by reader R.J. Feeser.[6] Most cone drivers have too much mass to provide response above 1–2kHz, where they must mate to a tweeter horn. In addition, very complex interactions take place between cone stiffness, mass, size, the air chamber about the horn throat and the throat reactance, that can cause severe irregularities in the response. However, I felt with the profusion of midrange drivers on the market today, and with proper testing, I could find a few good candidates for a midrange horn.

TESTING DRIVERS. Over the years I have assembled a collection of oddball speakers, including several midrange drivers. Through the aid of a friend, who managed a local electronics store stocked with raw drivers, I was able to test a representative sample of midrange cone and dome drivers. The total collection, between my own and the store's contribution, was about 20. A sample illustrating the variety is shown in *Fig. 2.*

A number of years ago I was given a pair of Western Electric (WE) type 31 PA horns with two inch diameter throats, but minus the drivers. Since the proper drivers for these horns are expensive, the WE horns collected dust in my closet for several years. In fact, this midrange horn article was stimulated by my desire to find a proper driver for them. In the process the horns became a useful test stand for screening drivers.

I used an old Scott receiver (*Fig. 3*) as a white noise generator (i.e., interstation hiss although a GR noise generator was used in final testing) and amplifier. A Sennheiser MD421 microphone was poked into the mouth of the horn for a near field response test. I used an HP sweeping spectrum analyzer for the initial spectrum meas-

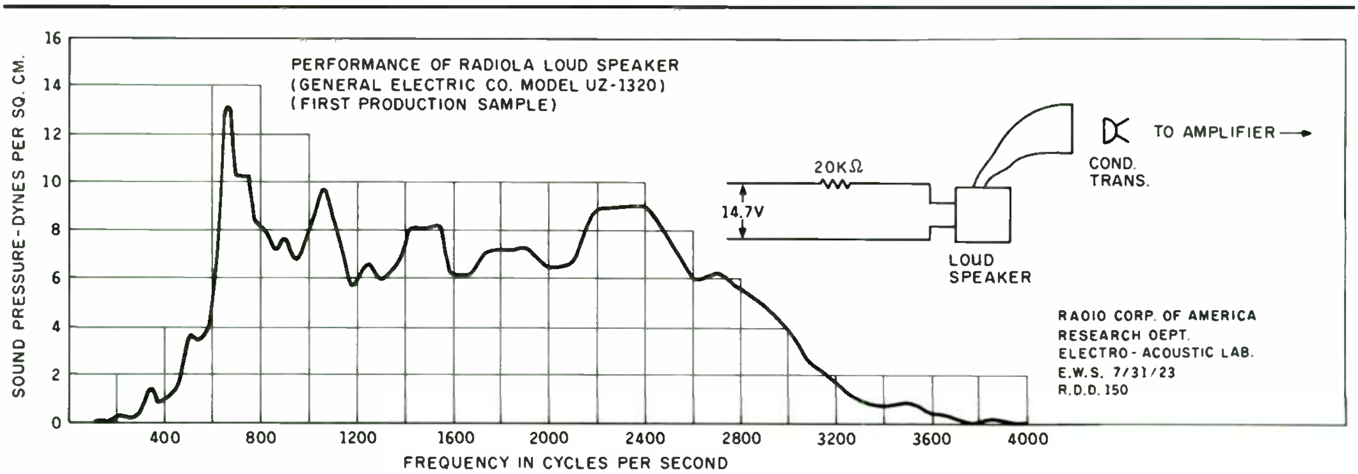

FIGURE 1b: The response of a midrange horn as measured by an RCA engineer (circa 1923). Other than the peak at the low end, the response is fairly smooth (less than 3dB ripple). *Courtesy of the Smithsonian.*

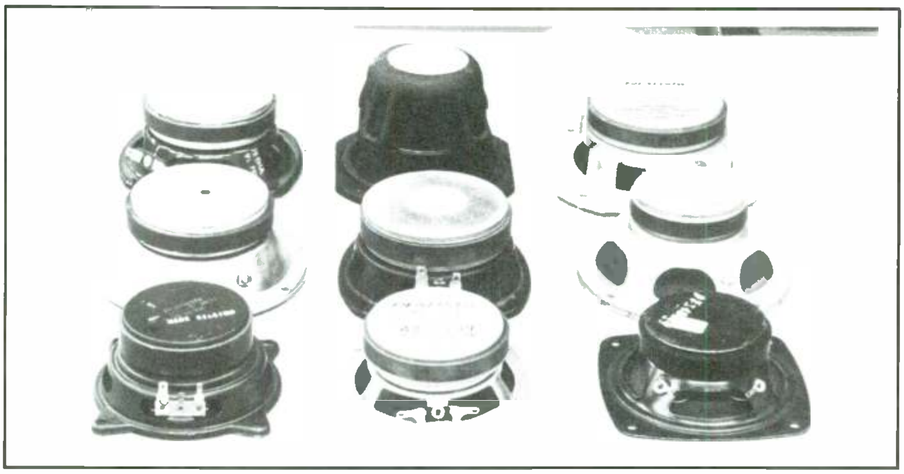

FIGURE 2: A sample of the drivers used in my initial tests.

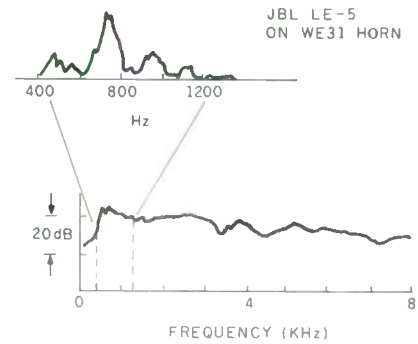

FIGURE 4: The spectrum response of a JBL LE-5 driver on the WE 31 horn. The blown up spectrum on a linear amplitude scale emphasizes the response peaks between 400 and 1200Hz.

urement, although later I substituted an FFT (Fast Fourier Transform) analyzer for greater resolution in the final tests.

I tried the WE horn with the 70Hz mini horn for a while with several drivers. My favorite driver was the JBL LE-5, a five-inch midrange cone driver that had long been the mainstay of the JBL speaker line. As a direct radiator midrange, it had a rising amplitude character with a frequency responsible for bright "west coast" sound. On a horn, the extended response of the LE-5 seemed to cancel out the natural tendency for the WE horn to attenuate the frequencies above 1kHz.

On many types of orchestral music the sound quality was outstanding, but when listening to a male announcer's voice on an FM station the midrange horn added an annoying distinct nasal quality to the sound. At the time I accepted it as something inherent in midrange horns as are their good qualities of increased dynamic range, lower distortion, etc.

When I first started white noise testing with the HP sweeping analyzer, I noticed an enhancement in the amplitude response around 500Hz with the WE horn. It was there with most drivers I tried, so I did not pay much attention to it. Later, with a time averaging FFT analyzer, the midrange enhancement resolved into distinct resonances, as shown in *Fig. 4*. Other drivers exhibited the same resonance structure so I began to question the design of this old WE horn as a suitable midrange horn design model.

OLSON'S CALCULATIONS. In the midst of my midrange horn quandary, a fellow scientist, Dave Rowe, walked into my office, introduced himself as a fellow speaker builder, and asked me to tell him all about horn loudspeakers. After some long discussions, he suggested that we take Olson's[7] horn acoustic impedance calculations and model the response variations of the various horns I had designed and built.

Acoustic impedance is defined as the ratio of the air particle pressure to the particle velocity. To most speaker builders this definition doesn't mean much. The following illustration may help.

If you place a tube of certain length over the front of a loudspeaker, you will notice the tube seems to reinforce certain frequencies. At these frequencies, determined by the length of the tube, a standing wave is created by the reflection of sound at the end of the tube. The speaker sees a high resistance of high acoustical impedance to work into at these resonant frequencies, and a low resistance in between. In a resonant horn, such as a trombone, the tube has a very slow taper until the horn's end flare, or bell. The rapid flare at the end creates a discontinuity or reflection point to create standing waves. A percentage of the sound energy leaks out and radiates as a musical note, while the rest remains in the standing wave inside the horn.

In contrast to resonant horns, a "transmission" horn, with an exponential, tractrix or conical flare, tries to minimize the reflections and match the high impedance a driver likes to see, to the low impedance of air. In a transmission horn the designer tries to minimize the ripple in the acoustic impedance, which the driver sees at the throat of the horn, and eliminate all resonant peaks (see Benade[8] for more discussion).

ACOUSTICAL IMPEDANCE. Olson gives the following expression for the

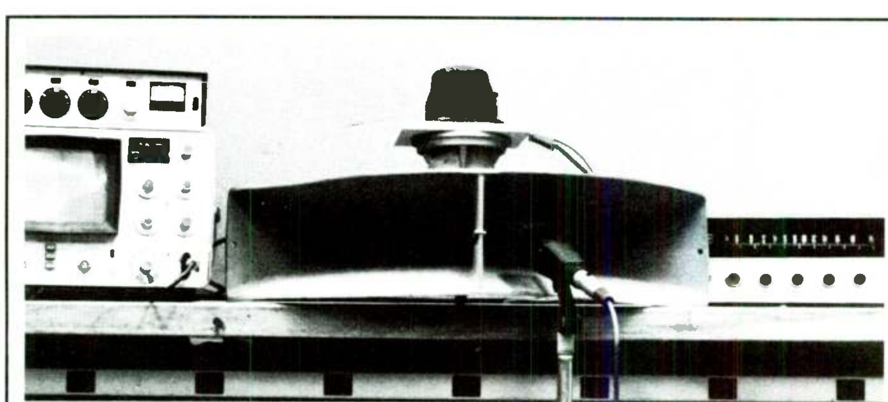

FIGURE 3: The initial test setup using a Western Electric Horn (Model 31), a HP sweeping analyzer, an old Scott receiver, a Shure microphone preamp and a Sennheiser MD-421 microphone.

acoustic impedance of a finite exponential horn:

$$Z_T = \frac{\rho C}{S_T}\left[\frac{S_m Z_m\,[\cos(bl+\theta)] + i\rho c\,\sin(bl)}{i\,S_m Z_m\,\sin(bl) + \rho c\,\cos(bl-\theta)}\right]$$

where:

S_T = area of throat
S_m = area of mouth
l = length of the horn
Z_T = acoustical impedance of throat
Z_m = acoustical impedance of mouth
a = $m/2$
b = $\frac{1}{2}\sqrt{4k^2 - m^2}$
θ = $\tan^{-1} a/b$
k = $2\pi/\lambda$
m = $4\pi f_c/c$
f_c = flare cutoff frequency
c = speed of sound
ρ = density of air
f = frequency
λ = wave length
i = imaginary unit $= \sqrt{-1}$

For the mouth impedance, Olson approximates Z_m as the air load upon one side of a vibrating round piston set in an infinite wall. However most horn mouths are not round, but rectangular for practical reasons. For a rectangular mouth, we used the rectangular piston radiation impedance functions as tabulated by Burnett and Soroka.[9] They showed acoustical impedance to be a function of the aspect ratio of the sides. (If the rectangle has sides of length a and b, then the aspect ratio R=a/b, where a>b.)

As an experiment we simulated the throat reactance of the 1920s exponential horn in *Fig. 1a* and the WE 31 horn. Parameters of the WE horn were physically measured to be: flare rate = 250Hz, throat size = 2 inches diameter, mouth size 5 by 26 inches, and length = 16 inches.

Figure 5a shows the calculated acoustic impedance versus the measured response curve for the 1920s horn. The two response peaks below 800Hz do not correlate well with the acoustic impedance peaks. The higher frequency response peaks are probably due to resonances in the metal diaphragm of the driver. *Figure 5b* shows a similar comparison between the response curve and the calculated acoustic impedances for the WE horn.

By now the reader will agree with me that the old design method for midrange horns has many faults. The acoustic impedance calculations show they have many resonances that are unacceptable by today's standards. The basic problem arises when the flare rate frequency is set well below the mouth cutoff frequency, and the mouth is too narrow (i.e., high aspect ratio).

The latter fact is demonstrated by the calculations of *Fig. 6*. We take the case of a midrange horn with a 350Hz flare rate and mouth size, and vary the shape, keeping the mouth area constant. Four aspect ratios, R=1, 2, 3 and 4 were used. The case of R=2,(i.e., mouth width=twice the mouth height), appears to have the minimum ripple in the acoustic impedance.

TRACTRIX HORN DESIGN. When I wrote my paper on tractrix horn design (*SB* 2/81, p. 9), I must confess a certain *naivete* about horn design. It was pretty much magic to me as it was to many others.

The tractrix horn expansion was a case in point. Along with many others,

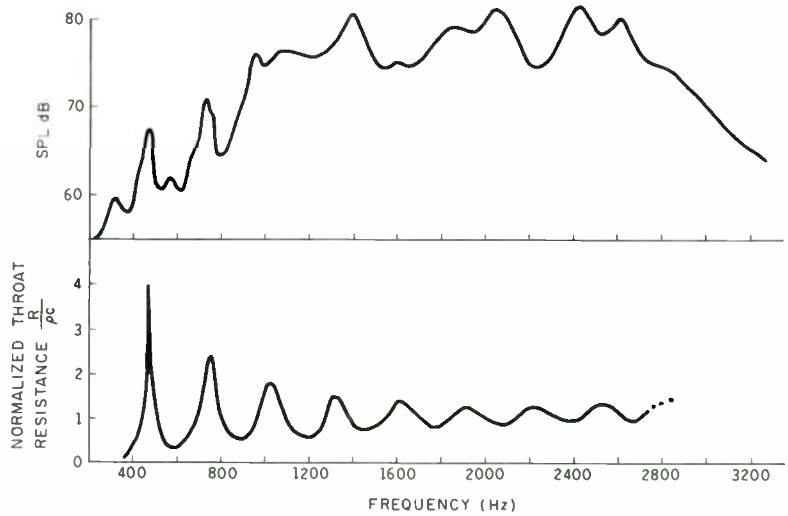

FIGURE 5a: The measured response of the old midrange horn of *Fig. 1a*, plotted on a log scale, compared with the calculated throat resistance.

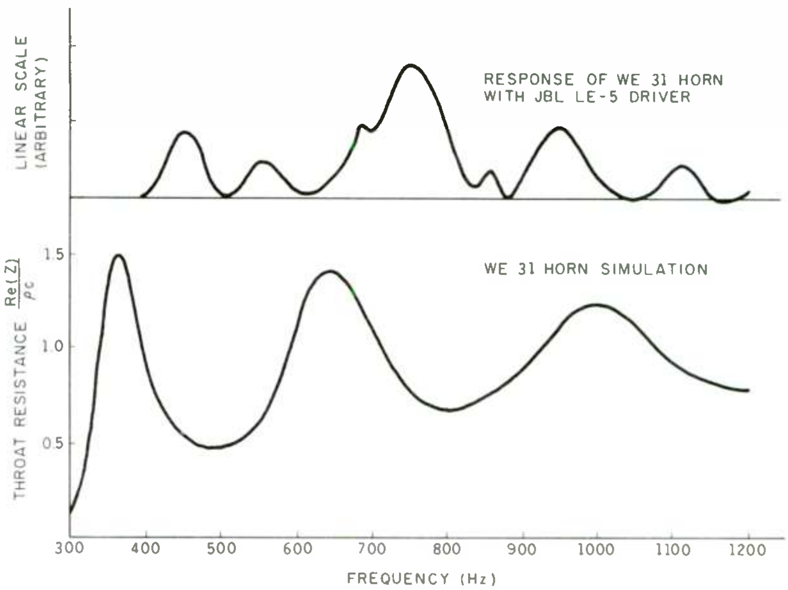

FIGURE 5b: A comparison of the measured response of the WE 31 horn, with the calculated throat resistance.

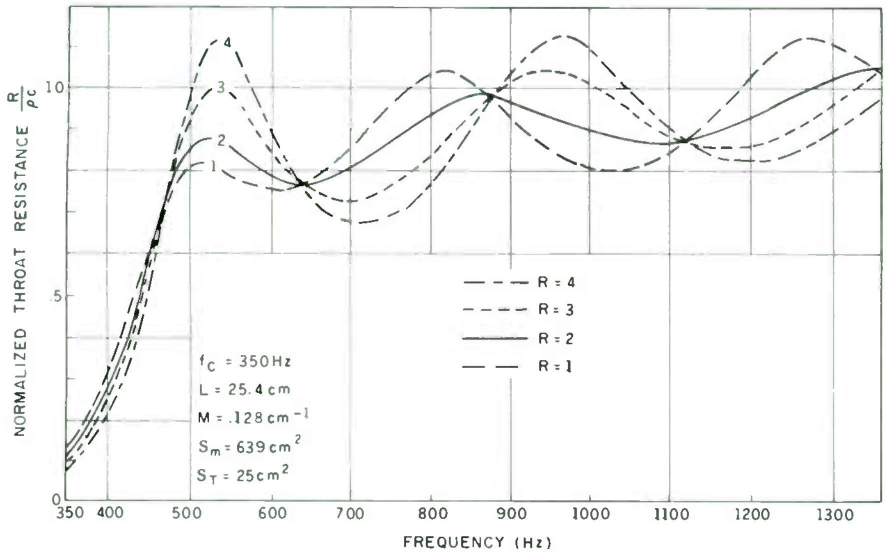

FIGURE 6: The effect of mouth shape on midrange horn throat resistance. R = 1 is a square mouth, and R = 4 is a long narrow mouth.

TABLE 1

300Hz TRACTRIX EXPANSION

2″ x 2″ Throat x (inches)	9″ x 18″ Mouth W/2 inches
0	1.0
1	1.05
2	1.10
3	1.21
4	1.38
5	1.60
6	1.92
7	2.30
8	2.80
9	3.55
10	4.65
11	7.0
11.5	9.0

I thought that the horn shape was the key to everything. However, at this writing, a great many other factors, i.e., mouth shape, throat size and coupling and driver choice, have just as much bearing on achieving a good horn loudspeaker. Having said all this, the tractrix horn is still a good choice for midrange horns because it launches spherical waves that can yield excellent stereo imaging effects.

CONSTRUCTION. For our midrange horn I chose a 300Hz tractrix expansion, with a 9 by 18-inch mouth and 2 by 2-inch throat, that would mate with my JBL LE-5 driver. This gives a horn length of 10 inches. For construction I followed the suggestion of Babani[5] to make the horn top and bottom as a wedge shape, and to construct the curved sides with strips of wood.

To start construction of the form, laminate together with nails or screws several 9 by 10-inch pieces of plywood or particle board until the thickness is 2-inches. It can be done with two 1-inch plywood pieces, or one ½-inch and two ¾-inch plywood sections. Cut the edges off as shown in *Fig. 7.*

Mount this trapezoidal piece with screws on a 10 by 18-inch board, as shown in *Fig. 8*, to make the jig for the horn. The side rails help keep horn parts rigid while building the unit. To cut out the top and bottom pieces, first cut out a template as shown in *Fig. 9.*

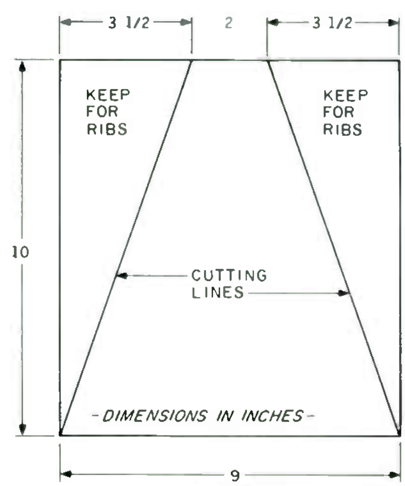

FIGURE 7: The cutout diagram for the trapezoidal form used in the horn jig. The form has to be 2 inches thick, and can be made from a sandwich with different thickness boards.

I plotted out the curve from *Table 1* on graph paper, with a 1 by 1-inch grid pattern that subdivides into 0.1-inch increments. With spray glue or rubber cement, mount the graph on a poster board or ⅛-inch masonite. Then cut out the template with a saber saw or an exacto knife. Take two 11½ by 18-inch pieces of ½-inch plywood or particle board, and nail them together with 1-inch brads. Place the template on the one side of the 11½ by 18-inch boards and trace out the tractrix curve.

With a saber saw or a band saw cut out the curved horn section. If your saw balks at cutting out 1 inch thickness, then cut the two horn sections separately. By cutting the sections together, you obtain a symmetry of the two sides that accommodates any slight irregularities in sawing out the pattern.

I suggest you start a saw cut at the mouth. If you start from the throat, by the time you get to the narrow section of the mouth, the boards will tend to break off due to their own weight.

Separate the two curved sections and remove the brads. Place them on either side of the trapezoid form (as shown in *Fig. 10*) so they are properly centered. Attach them temporarily with a nail or screw to the trapezoid form so the nail or screw can be removed later.

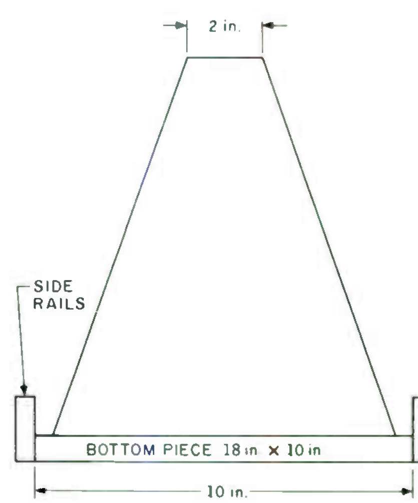

FIGURE 8: The trapezoidal form mounted on the jig. The bottom piece (18 by 10 inches) can be a ½ inch or greater in thickness.

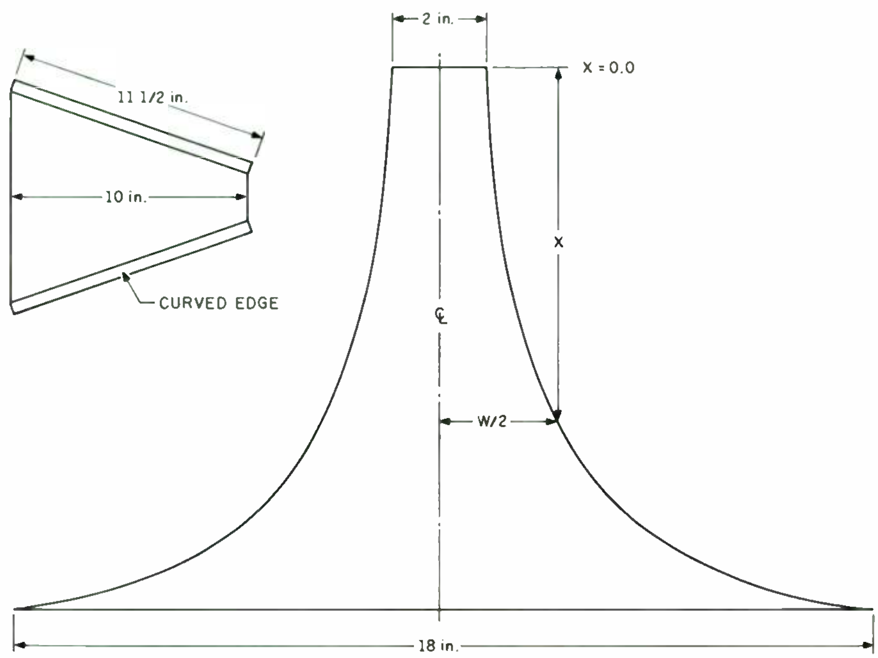

FIGURE 9: The scaled template for the top and bottom horn pieces. Plot *Table 1* on 1-inch grid engineering paper for a full size template.

Now rip a number of 1-inch (approximate) wide strips of ½-inch particle board. My strips are 1⅛ inches wide only because I found them in a scrap bin at a local woodshop (I'm a serious scrounger). Cut ten strips into 10 inch lengths. Place one across the curved sections so it straddles both. See that it touches the outside edge of the curved sections. Glue must be placed here for bonding. Spread a thick bead of carpenter's glue along the outside edge, starting at the mouth end and going up about 6 inches along the edge.

Place one of the 10-inch strips at the bottom, making sure the glue bonds between the strip and curved edge. Take another 10-inch strip, spread a bead of glue the length of the long edge where it comes in contact with the first strip. Repeat this process for five strips until it looks like *Fig. 11*.

Follow the same process for the other side. Let the glue set overnight, or for several hours. The reason for doing only five strips at the bottom is they tend to slide off the form if stacked up while the glue is wet. I have nailed the strips with brads, but the board tended to split or break off on the narrow curved end.

After the glue has set on the first strips, you can start cutting, fitting and gluing the succeeding strips until you reach the end of the throat. When the glue is dry on the rest of the strips, you may drive some 1-inch brads to add more strength. However, the structure is extremely rigid and non-resonant with only glue joints so that nailing is optional.

Remove the nails or screws that held the curved sections to the trapezoidal jig and lift off the horn. You may have to gently pry it off because sometimes glue will run down and stick the horn to the jig.

Looking into the horn you may notice some ⅛ inch gaps between the strips and the curved sections. I used Fixall or Water Putty to fill these. (Fixall is a coarse material and a fast drier. Water Putty is finer and takes longer to set.) After the gaps are filled, smooth off the excess filler with damp paper towels. Let the filler set and dry.

CONSTRUCTION TIPS. Some of you may ask, "Why can't I set my saber saw at the proper angle so the strips will be flat on the curved sections?" Well, if you notice after taking the horn off the jig, the gap between the strips and the curved section varies from being wide at the mouth to narrow at the throat, so there is no constant angle.

If you have access to a band saw, you can take a wedge (use one of the wedge cutoff pieces from making the jig) and mount it on one of the 9 by 18-inch rectangular pieces where the throat outline is drawn. When the tractrix curve is cut out, in the manner shown in *Fig. 12*, the correct angle is achieved along the curved section to allow flush mounting of the strips.

An alternative to using strips for constructing the curved side is to steam and bend wood. I tried this with ⅛-inch plywood (shown in *Fig. 13a*), but the thin walls have resonances (*Fig. 13b*) that show up in the response.

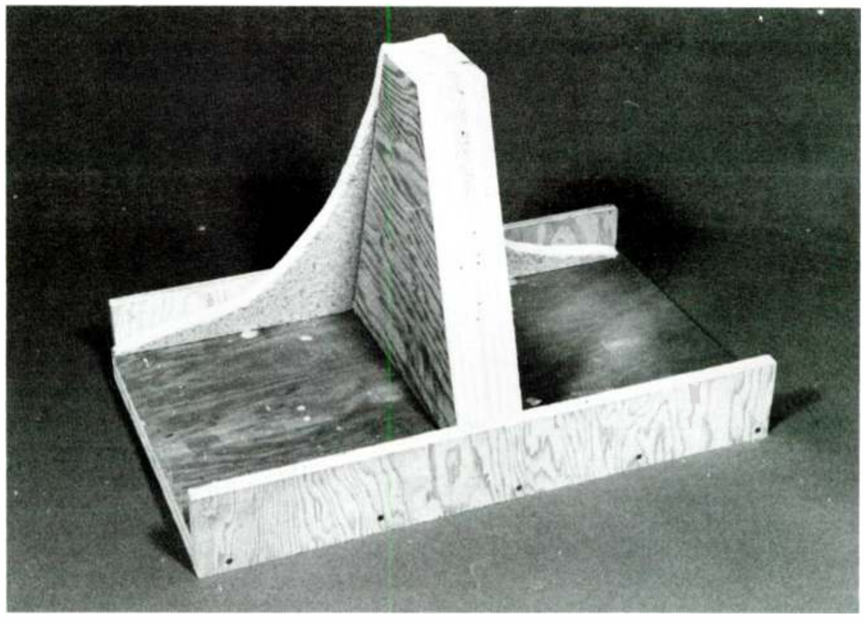

FIGURE 10: The horn jig with one top piece in place.

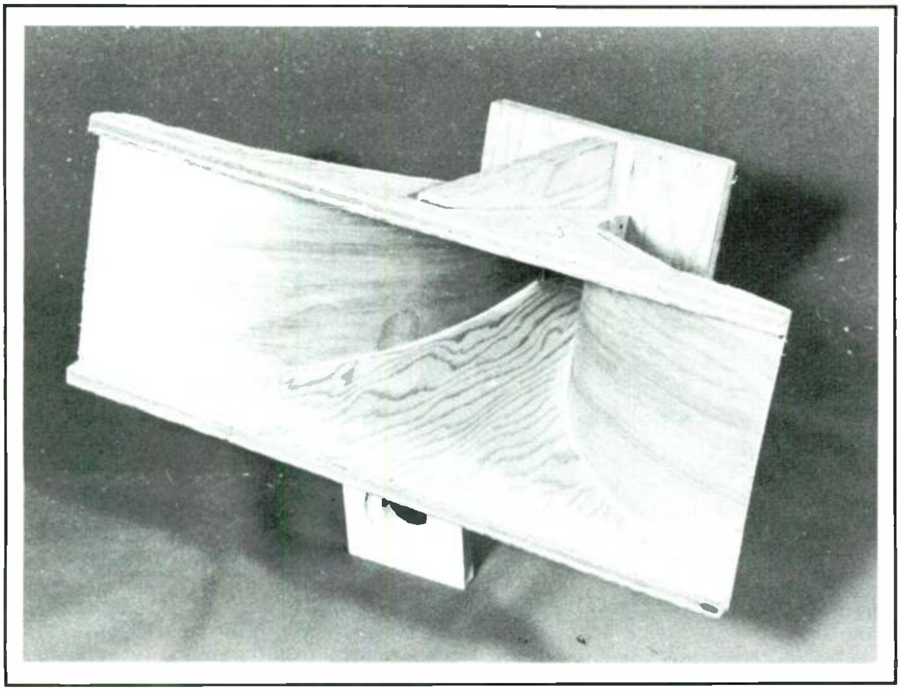

FIGURE 13a: An early experimental horn of mine with thin walls of 1/8-inch bent wood.

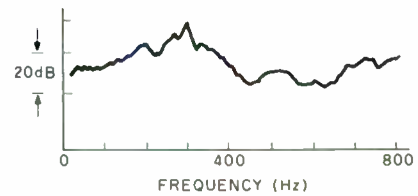

FIGURE 13b: The resonant nature of the thin wall horn produced by thumping the horn in front of a microphone. Notice the peak at 300Hz that will color sound.

Thicker plywood, ½ inch or greater, may give better results. But for the speaker builder with normal skills, the particle board strips are the easiest method.

FINISHING THE HORN. After the gaps have been filled, take a wood rasp or a Stanley Surform file and round off the corners of the 1-inch particle board strips where they join. Work off the excess material with rasps, files and sandpaper until you can run your hand over the curved surface and feel no bumps or joints. It's amazing how this relatively simple but labor intensive procedure can generate such a nice looking curve.

Now decide the thickness of your driver mounting plate. In my case, it depended on what scrap piece of ¾ or ½-inch plywood I had to give me a 10 by 10-inch square piece. Cut off the neck by the thickness of your mounting plate. I usually do not build up the strips on the curved side to the end of the throat, so that I can easily square off the throat's end with a belt sander.

Once the throat is faced off, take two of the end wedges left over from making the trapezoidal form. Center and glue them with clamps on the top and bottom flat sides to form the ribs (see *Fig. 14*). Now attach the speaker mounting board with screws to the ribs. I usually center the mounting board over the throat, then drill and drive in one screw. Then I place it on the side on which it normally rests in a system, and adjust the mounting board slightly so the whole assembly does not tip. Now you can drill and drive the other mounting screw.

Cut the throat opening next. With a pencil, mark the outline of the throat on the mounting board. Remove the mounting board. Center and draw a 2 by 2-inch square inside the throat outline. With a saber saw and wood rasp, cut out a 2 by 2-inch conical square throat opening on the speaker side, that matches with the throat opening on the other side.

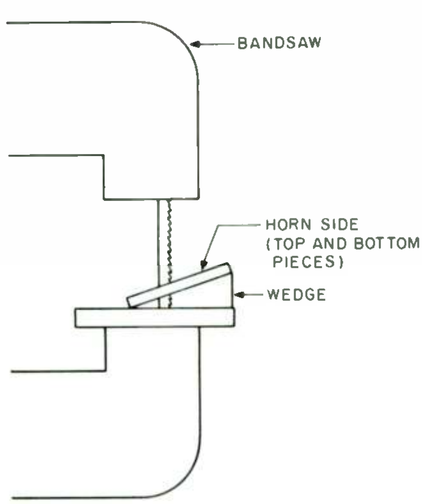

FIGURE 12: An alternative method for correctly cutting the beveled edges for the top and bottom pieces.

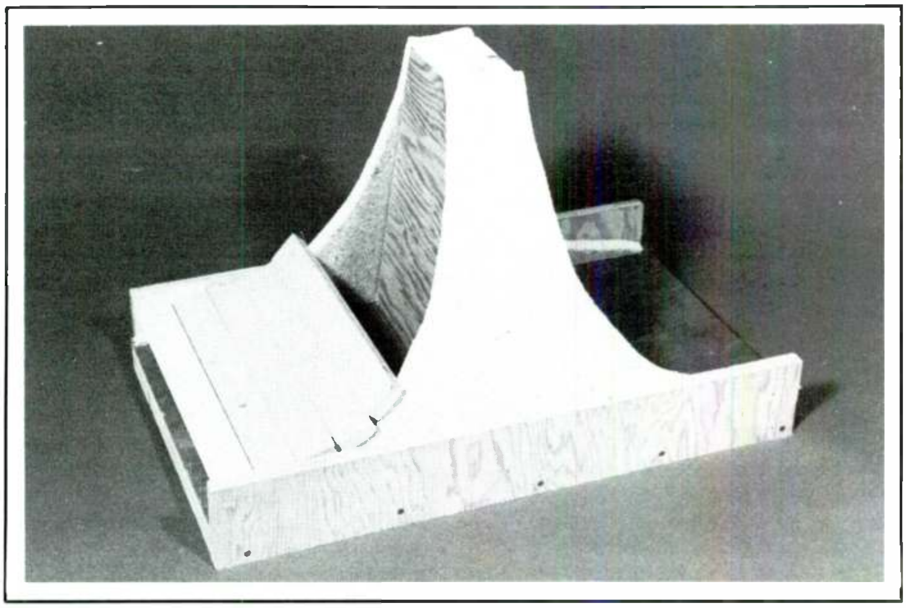

FIGURE 11: The initial placement of 1-inch wide strips to build up the horn's sides.

Now I could say you are finished, but the next problem is to match the speaker to the horn throat. The JBL LE-5 matched the tractrix horn with the 2 by 2-inch throat well, with a fairly flat frequency response from 400 to 4kHz as shown in *Fig. 15*. It also had a 105dB sensitivity at 1kHz (1W input measured with an Audioquest sound level meter at 1 meter away from the mouth), which is comparable to commercial horns.

Unfortunately, JBL no longer sells individual drivers from their commercial lines. But if you have a JBL 3-way system from the 1970s, it may have a usable LE-5. A possible improvement would be to make the tractrix horn, mount the LE-5 on it with an L-pad and hear an improvement in sound clarity and crispness.

AN ALTERNATE DRIVER. Because the JBL LE-5 was not available, I began to look at other candidates. The characteristics of the LE-5, which seemed to work well with horn loading, were a high BL factor and a small light cone. These characteristics give a radiation response which rises with frequency.

The first alternative candidate I found was a SIARE 16VR. It had a 40-ounce magnet and a 6-inch figberglass cone. However when I tried it on the horn with a 2 by 2-inch throat its frequency response, as shown in *Fig. 16*, was somewhat disappointing because of the 1.5kHz rolloff.

I concluded after some thought that the high frequency response could be improved if I used a larger throat to couple 1:1 to the cone size. When I sawed a length off the throat, as shown by *Fig. 17* and *19a*, the response improved to 3kHz. But a response peak at the low end appeared, as shown in *Fig. 18*.

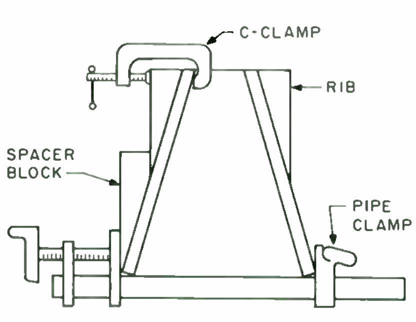

FIGURE 14: The clamp arrangement for mounting the ribs.

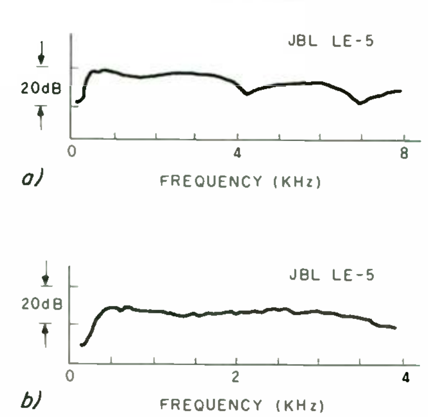

FIGURE 15: Response *(a)* is of a JBL LE-5 driver on 300Hz tractrix horn. Note the nulls at 4 and 7kHz. Response *(b)* is of *(a)* on an expanded scale.

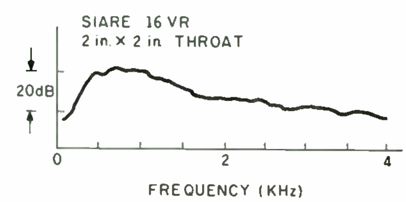

FIGURE 16: The response of an SIARE 16VR driver on a horn with 2 by 2-inch throat. Note the restricted response.

I accidentally found that if I introduced a gap in the throat-speaker interface, by offsetting the driver with ½-inch spacers (see *Fig. 19b*), the peak went away, as shown in *Fig. 20*. The gap's effect on the sound quality was very evident as the horn was lowered on the SIARE 16VR driver with white noise excitation. Without the gap the sound quality changes from an "open" sound to a "tunnelly" resonant quality.

The physics of going from a small throat to a large one are that phase cancellation occurs in the small throat, when sound paths to the throat from the cone are unequal. In compression driver horn systems, a phasing plug is used to solve this problem, but the complexity of a phase plug puts it out of reach for the amateur builder. By going to a larger throat, of a size comparable to the cone size, phase cancellation is reduced significantly. You will lose some efficiency, but I found the losses are small compared to the added octave of response gained.

The physics of the gap are that the gap behaves as a high pass filter, which effectively clips off the peak. The gap is equivalent to a short open tuning stub such as the one treated by Olson.

OTHER DRIVERS. I began to try other drivers that had the large BL factor and a rising radiation characteristic. The Polydax dome HD13D37 1¼-inch dome exhibited these characteristics. However, on a 2 by 2-inch throat horn it exhibited an annoying peak at 500Hz; but the rest of the response

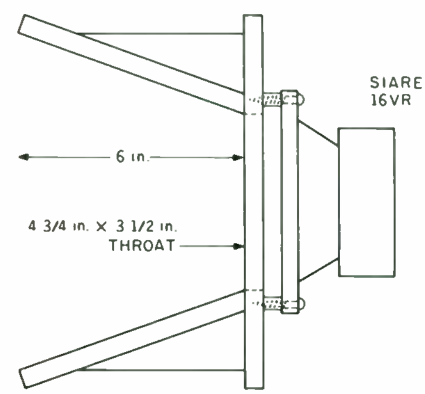

FIGURE 17: A shortened horn with a larger throat to match the SIARE 16VR driver.

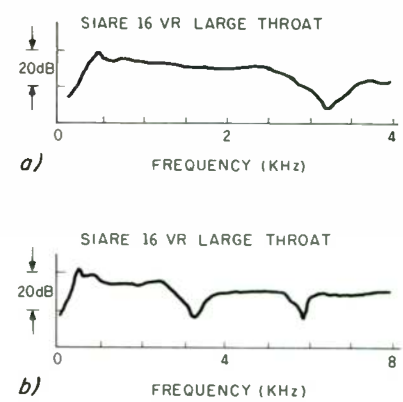

FIGURE 18: Response *(a)* is of an SIARE 16VR on a large throat horn. Note the response peak at the low end (500Hz). Response *(b)* is of *(a)* on an expanded scale. Note the nulls in the response are similar to those in *Fig. 15*.

Putting the Polydax dome on ½-inch standoffs helped somewhat, but filling was quite smooth and free from stricture, as shown in *Fig. 21a*. When I shortened the horn to a 4 by 2½-inch throat (7½ inches along the center axis), the response changed to a nice flat characteristic response of 4kHz, but a peak was still evident as demonstrated in *Fig. 21b*.

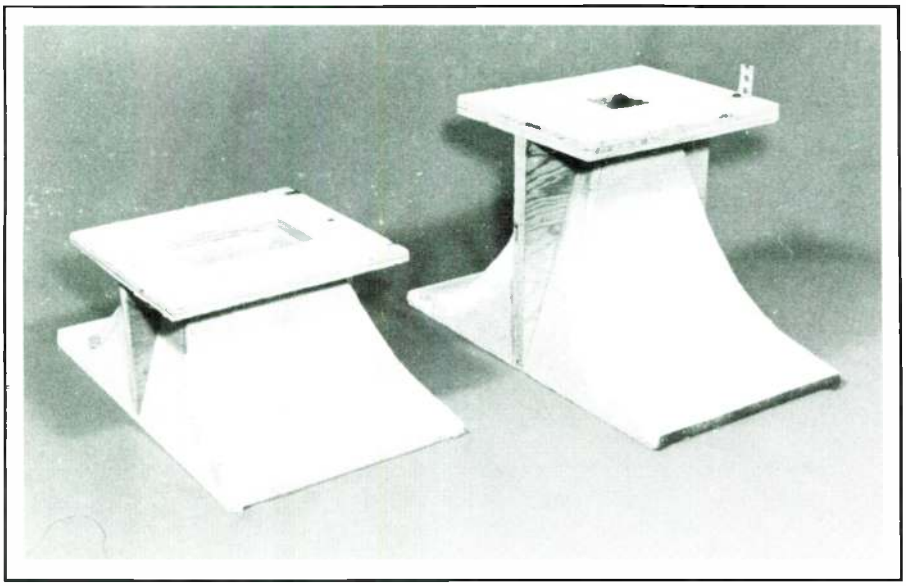

FIGURE 19a: A comparison of the two horns with differing throat sizes and horn lengths.

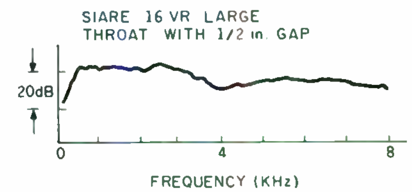

FIGURE 20: The response of an SIARE 16VR mounted on the shortened horns with ½-inch spacers. Notice the response has smoothed out from those in *Figs 16* and *18*.

the gap with ½-inch thick open-cell foam helped smooth out the response, as shown in *Fig. 21c*. I cut out a 1¾-inch diameter hole in the foam, to accommodate the dome and holes for the standoffs and mounting screws, as shown in the schematic *Fig. 22*. The foam acts as a resistive gap smoothing out the response above 3kHz and nicely extending it to 6kHz. The sensitivity is typically 100dB.

I found packing foam, typically found in shipping boxes, works the best. It is so porous you can hold it to your mouth and blow air through it. The foam, however, will deteriorate with time. I found using two layers of Scotchbrite® abrasive pads (found in hardware stores) will also work, and remain stable over time.

Searching the Polydax catalog, I found other driver candidates. One a 4-inch cone driver (HD12P25FSM) exhibited the best response (*Fig. 23*) on the same horn (4 by 2½-inch throat) as the Polydax dome, but with a ¼-inch gap and strips of Scotchbrite around the throat gap (but not covering the opening), it displayed 100dB sensitivity.

One person recommended the Polydax 6-inch professional midrange (PR17HR37TSM) used by several professional sound companies on their horns. *Figure 24a* plots the response on the large throat horn used with the SIARE 16VR. It shows the now familiar peak at 500Hz but, even worse, some irregular structure above 3kHz. Introducing a ½-inch gap at the throat, with foam strips around the edge, smooths out the response as shown in *Fig. 24b*.

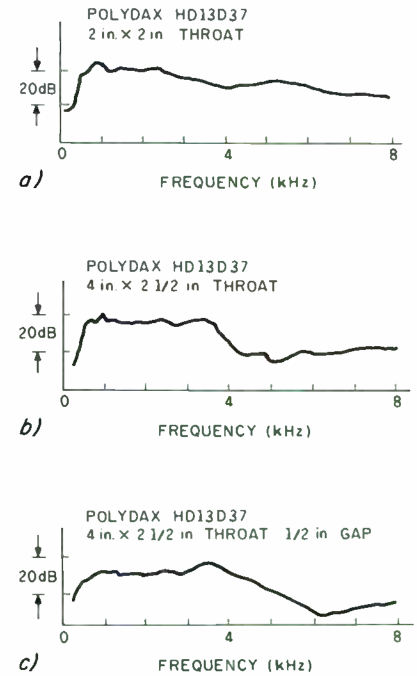

FIGURE 21: Response *(a)* is of a Polydax HD13D37 dome on a tractrix horn with a 2 by 2-inch throat. The response has an annoying peak below 1kHz. Response *(b)* is of a Polydax dome on the midsize throat tractrix horn. The response peak, although reduced, remains. Response *(c)* is of a Polydax dome with a ½-inch gap at the throat filled with porous foam rubber. The response is smoothed.

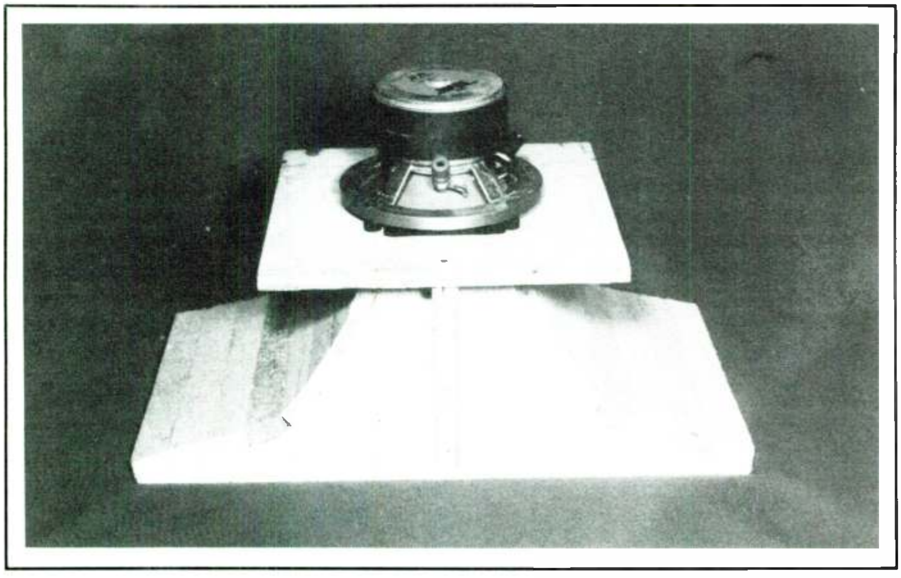

FIGURE 19b: The SIARE 16VR driver mounted with spacers to remove peak at 500Hz.

As I was trying to wrap up this article, I received a catalog from Focal Loudspeakers that described a number of drivers with possibilities for horn loading. Of the four models I tried, one 7-inch model (7N303) gave a very flat response (one of the best I've seen so far) on the large throat horn with about 100dB sensitivity. It also has some possible midbass horn applications with a resonant frequency of 70Hz.

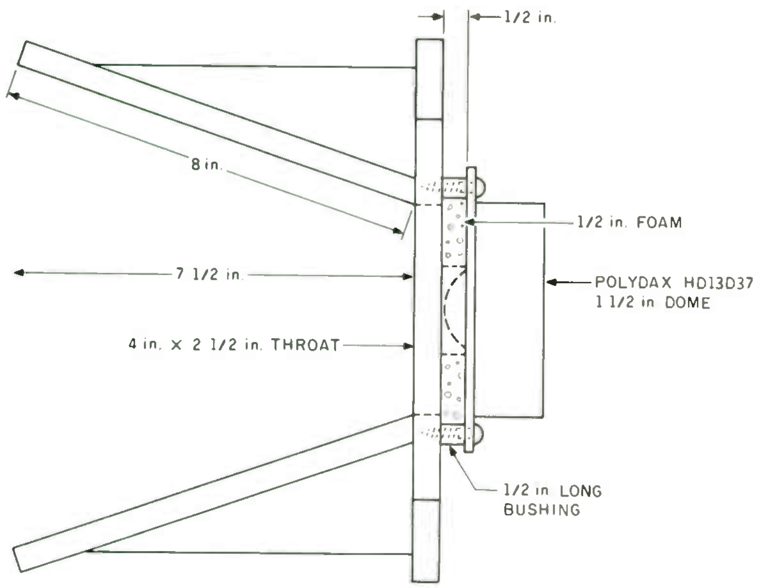

FIGURE 22: A cut away of the midsize tractrix horn with the Polydax dome driver.

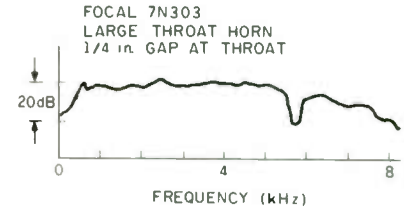

FIGURE 25: The response of a Focal 7N303 7-inch driver on a large throat horn with a ¼-inch gap at the throat.

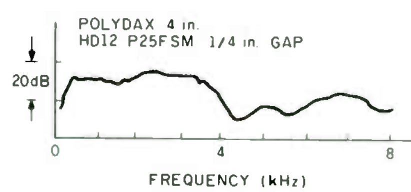

FIGURE 23: The response of the Polydax HD12P25FSM 4-inch driver on the tractrix horn with a ¼-inch gap at the throat and the gap edges filled with foam.

A NEW THEORY. In the classical theory of horns the upper mass cutoff $(2f_s/Q_{es})$ usually defines the effective upper frequency range. For example, in the case of the LE-5, I measure a Q_{es} of 2.74 and f_s of 316Hz, which would mean an upper mass cutoff of 230Hz, obviously well below our 4kHz measured response. I did not understand what was going on until I interviewed Ted Jordan (*SB* 2/84) and heard his ideas of suspension control of diaphragms.

Jordan,[10] and others who design full-range speakers, rely on stiffness and damping of the suspension to extend the high frequency response of the driver. This stiffness will overcome the control of the cone mass upon the power radiated from the speaker. Suspension control also gives a rising radiation characteristic, which eventually rolls off at some upper frequency, where the mass of the cone begins to exert influence.

A horn loads a driver by replacing the characteristic radiation resistance of a direct radiation (proportional to f^2, where f is frequency) by a constant radiation resistance determined by horn parameters. This phenomenon is schematically shown in *Fig. 26*, where the rising portion of a piston's radiation resistance is replaced by the horn's higher radiation resistance.

The mathematics of this phenomenon are somewhat complex, as demonstrated by Brociner[11], and I have yet to fit it all together. But, as demonstrated here, it will suffice to say cone drivers can and do work on large throat horns with good efficiency and frequency range.

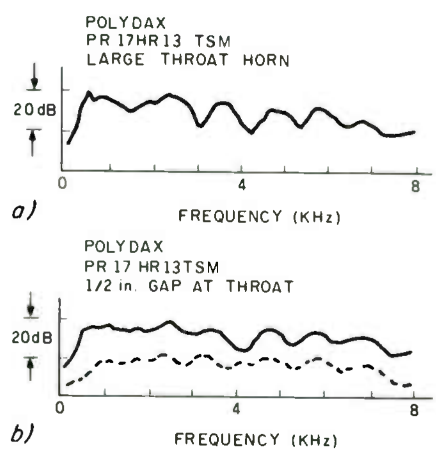

FIGURE 24: Response *(a)* is of a Polydax PR17HP37TSM 6-inch driver on the large throat horn. Response *(b)* is the same driver with a 1-inch gap at the throat stuffed with foam at the edges. The free air space response is shown for comparison.

MATCHING OTHER SPEAKERS. With closed box speakers the sensitivity of most components, (i.e., woofers, midranges and tweeters), usually ranges around 90dB. However, the sensitivity of horn systems can range from 100 to 110dB; so some attention must be paid to the proper integration of midrange horns, with horn and closed box systems.

Any of the midrange horns described earlier can be used with a closed box system. However, the midrange driver must be attenuated by an L-pad to bring it down to the level of the closed box woofer. Usually this attenuation is on the order of 10 to 15dB. I have used several of the midrange horns, with both sealed box and bass reflex systems, and midrange detailing from the horn significantly increases the resolution and imaging qualities of the total system. It pays to move the tweeter driver back so it is in proper alignment in regard to time with the midrange driver. When this modification is made female singing voices usually change from a rough character to a smooth blend.

In horn systems it is usually best to select a midrange horn with the same sensitivity level as the bass horn. If the bass horn is rated at 105dB, use one of the 105dB driver midrange horn combinations. For the Klipschorn, my midrange horn size should fit in the top volume for the mid and tweeter horns, although you may have to enlarge the cutout for the mouth. The Klipschorn is usually rated at 104dB sensitivity, so a good choice is one of the 105dB rated horns with some slight attenuation.

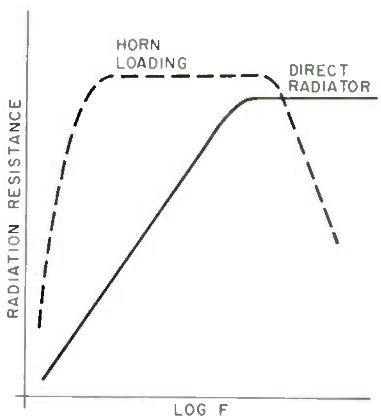

FIGURE 26: The radiation resistance of a direct radiator versus that for horn loading.

Some readers may have noticed I did not use back chambers on the open back drivers. I did experiment with back chambers and found the box sizes were usually comparable to the half or quarter wavelengths, at some midrange frequency, so either resonances or nulls would show up in the response.

If you do place the midrange horn in the Klipschorn, isolate the back with fiberglass insulation. If you isolate the speaker with a closed back, make it as large as possible and stuff it with fiberglass insulation or other good absorbent material. However, the best response is usually obtained with the back open because the drivers are suspension dominated and don't need a restoring force from an air chamber.

CONCLUSIONS. Although this midrange horn construction project may seem out of the mainstream for some speaker builders, it offers a new and different path for upgrading your old horn system or closed box system. Most of the comments I have received from listeners are on the clarity and crispness of the sound. The transient detail of the violin bowing or guitar pick is evident without harshness.

I think it is also evident from this article that one can easily build a bad-sounding midrange horn system. In the literature you will not find any discussion of horn mouth shape influence on sound quality. Although some large PA horns with square or similar mouths are on the market, most of the horns available to the speaker builder have wide narrow mouths that may lead to a peaky response.

The route I have shown for a good horn design is to choose a rectangular mouth (2:1 aspect ratio), or a square mouth, and a mouth size frequency cutoff equal to the flare rate frequency. Deviations from this design philosophy will introduce peaks in the response.

The most important ingredient in this approach is good driver selection. You may have noticed that I mentioned only drivers that work well with horn loading (see *Table 2*). If I listed the ones that do not work with horns, this article would run on for several more pages, because most drivers are designed as direct radiators, not horn drivers. However, I have shown that a small group of drivers, with larger than normal magnets and with stiffness control of the diaphragm, work well with horn loading. With stiff diaphragm drivers a throat size equal to or slightly larger than the driver will give the most extended range. Also a gap at the throat, either left open or stuffed with foam, will tame resonances at the low end and remove the "horn like" sound coloration.

This article has taken a number of years to assemble and digest the research results, and I think the wait has been worthwhile. It has been a labor of love with countless evenings and weekends spent with numerous drivers and horns in front of a spectrum analyzer. So for all the horn enthusiasts out there, I give you the *Edgar Midrange Horn*.

ACKNOWLEDGEMENTS

I thank Manfred Buechler for taking photos of the midrange horns through their many stages; Dave Rowe for making the acoustic impedance calculations and helpful advice; Ed Sivowitch of the Smithsonian for making available the old horn data; Evan Struhl for the loan of several Polydax drivers; Kimon Bellas for the loan of several Focal drivers; ITC Electronics for the loan of many other midrange drivers of assorted makes. And finally I thank my wife Nancy, for putting up with the endless progression of midrange horns dotting the living room.

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ABOUT THE AUTHOR

Dr. Bruce Edgar is a space scientist currently involved with planning space shuttle experiments for a Los Angeles based aerospace company. His hobbies, other than horn building, include woodworking and cycling. He also serves as president of his company's audiophile club. He is married and has two teenage sons.

TABLE 2

DRIVER	PRICE CLASS	HORN	RESPONSE	SENSITIVITY
JBL LE5	$60 (used)	Long	400–4kHz	105dB
SIARE 16VR	$60	Short	400–4kHz	105dB
Polydax HD13D37	$20	Mid	400–5kHz	100dB
Polydax HD12P25FSM	$20	Mid	400–4kHz	100dB
Polydax PR17HR37TSM	$40	Short	400–4kHz	105dB
Focal 7N303	$60	Short	400–5.5kHz	100dB

Note: Any horn (less driver), as built by the author, may be purchased at a cost of $60. Contact: Bruce Edgar, Box 1515, Redondo Beach, CA 90278, or use Fast Reply # FI16 for more information.

SAND STANDS

BY JOSEPH P. KMETZ

More and more researchers have been experimenting with the effect of stands on speaker performance. Objective and subjective evaluations by top British designers such as B&W and Celestion have indicated that marked improvement in sound quality results when speakers are placed on rigid, damped stands. This type of stand decouples the speaker from its surroundings, hence producing less "motion pollution," which can adversely affect the rest of your system. Celestion recommends using its superb SL-6 on a sand-filled stand to extract the best speaker performance.

In this article, I will explain how to construct similar stands that are adaptable to most speakers, using common hardware supplies. You will need scraps of ¾-inch particle board, 3 or 4-inch-diameter PVC plumbing pipe, a supply of long-drying epoxy (PC-7 works well), a bag of play sand and a can of spray paint. *(Editor's Note: To ensure that the sand is completely dry, try oven-drying it in shallow pans before use.)* The total cost of the materials should be only $10 to $15, and when you are finished, you will have a set of damped speaker stands that cost considerably less than those available from retail stores.

CONSTRUCTION. Start by measuring the width and depth of your cabinets. Cut two pieces of particle board 1 inch bigger than your cabinets' depth and width dimensions (i.e., if your cabinets measure 12 by 14, make the two pieces 13 by 15). These will be the tops of the stands *(Fig. 1)*. Now cut two more pieces of particle board, this time making them 4 inches bigger than your depth and width measurements

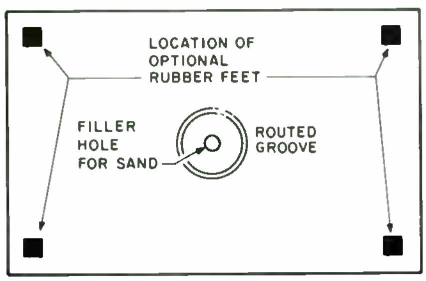

FIGURE 1: Cut the stand tops 1 inch bigger than your cabinets' depth and width dimensions.

(i.e., if the cabinets measure 12 by 14, make these pieces 16 by 18). These will be the base pieces for the stands *(Fig. 2)*.

Next, determine the height of your stands. The mid/high-frequency drivers should be at ear level when you are sitting in your favorite listening chair. Cut two pieces of PVC pipe ¾ inch shorter than this measurement. I recommend 4-inch OD pipe if you can find it, with ⅛-inch walls. If you

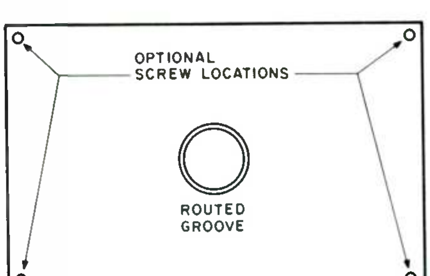

FIGURE 2: The base pieces should be 4 inches bigger than your cabinets' depth and width measurements.

cannot find 4-inch pipe, 3-inch will work well for all but the heaviest cabinets. Make the cuts as square as possible. I recommend using a table saw with a guide for this. If you do not have access to a table saw, careful cutting with a hacksaw will do.

Set the PVC pieces aside and locate the center point of one particle-board piece. Using your router and router guide, route a ⅜-inch-deep circular groove of the same diameter as your PVC pipe *(Figs. 1 and 2)* in the particle board. If you do not own a router guide, Michael Boulais explains how to make one inexpensively in his Dynaudio construction article (*SB* 1/83, p. 7). If you are using ⅛-inch-wall pipe, a ⅛-inch rabbet bit makes a nice groove into which the tube will fit nicely. Repeat this procedure for the remaining three pieces.

ASSEMBLY. Prepare enough 24-hour epoxy to fill the grooves you routed in the particle board. Place the two larger pieces (the bottoms) on a level surface. Make sure you check the surface with a carpenter's level, or you might end up with lopsided stands. Fill the grooves with epoxy and press the tubes into the grooves. Wipe away any excess epoxy and make sure the tops of the pipes are level (that is why the pipes must be cut square in the first place). Allow the epoxy to harden, then apply a small bead of epoxy around the outside of each pipe where it meets the particle board to make a stronger joint. Smooth the epoxy around the pipe for a one-piece look.

After this dries, drill a ⅜-inch hole through each of the smaller pieces of board (the tops) at its center *(Fig. 1)*. Fill the grooves (but not the hole) with

MATERIALS LIST

(4) pieces ¾-inch particle board (cut to suit)
(2) pieces 4-inch PVC pipe (cut to suit)
long-setting epoxy
bag of play sand
can of spray paint
silicone seal
(8) self-sticking rubber feet (optional, see text)
(8) screws to suit (optional, see text)
tools: router and router guide, funnel, drill and drill bits, carpenter's level, screwdriver, sandpaper (for finishing)

epoxy and press them onto the tops of the tubes. Again, make sure they are level and adjust them before the epoxy sets. Apply a small bead of epoxy to the top joint of each as above. Let the entire assembly dry overnight.

Now comes the return to your childhood. Place a funnel of the appropriate size in the ⅜-inch hole in one of the top boards. Slowly fill the pipe with sand, gently tapping it to make sure the sand is settling properly. Be patient: the sand will settle quite a bit, and it must fill the pipe completely for the stand to be totally effective.

When you are confident that no more sand will fit in the pipe, fill the hole with silicone sealer. After the silicone dries, it will form a plug you can remove to check for further settling of the sand. Repeat this procedure for the other stand.

I recommend that you recheck the sand after a few weeks of use to make sure that the pipes are still full. Then finish the edges of the particle board and spray paint the stands.

REFINEMENTS. You can refine your project by following two suggestions. I strongly recommend the first and leave the second to your discretion.

Ideally, your speaker cabinets should be seated firmly on top of the stands without any fore and aft motion. If you simply place the cabinets on top of the stands, the absence of friction between the two mating surfaces will allow the cabinets to slide back in reaction to the woofer's forward motion (like the thrusting of a jet, only on a smaller scale). This will cause smearing of the sound because the piston motion is not rigidly coupled to the air. To solve this problem, you can buy self-sticking rubber feet (Radio Shack part #64-2342), which provide enough friction so that the speaker will not slide and also provide additional decoupling for the cabinets. Put one rubber square in each corner of the stand tops *(Fig. 1)* and place the cabinets on them.

The second suggestion is of real value to those placing the stands on a carpeted floor. You will need eight screws that are long enough to penetrate the bottom of the stand as well as the carpet so that their points come in contact with the floor underneath. Simply place one screw in each corner of the bases *(Fig. 2)*. These "spikes" prevent the stand and speaker assembly from sliding around on the carpet, making it more rigid. The spikes will work equally well on bare floors; just make the screws shorter.

That's all there is to it. I look forward to hearing from anyone who tries this project and would appreciate any suggestions you might have for further improvements.

ABOUT THE AUTHOR

Joseph Kmetz works as a marketing consultant and has been interested in audio for more than ten years. He has built most of his audio system from scratch, with the help of SB *and* Audio Amateur, *and he hopes to open his own hi-fi shop sometime this year.*

PART IV

PASSIVE CROSSOVER NETWORKS

ACTIVE REALIZATIONS OF TWO-WAY DESIGNS

BY ROBERT M. BULLOCK III
Contributing Editor

I am going to show you, as promised in Part III (*SB* 3/85, p. 14), how to design active three-way all-pass and constant-power crossover networks, using only the 30Hz Rumble Filter Kit from Old Colony (KF-6, $19.75; PO Box 243, Peterborough, NH 03458).

THREE-WAY NETWORKS. The filter complement of a three-way crossover network consists of a pair of two-way crossover networks, one each centered at the low and high crossover frequencies, f_L and f_H ($f_L < f_H$) respectively. The two-way stages are usually interconnected in "cascade" as in *Fig. 1*, or in "parallel" as in *Fig. 2*. The parallel connection is better when realizing the networks by resistance terminated LC ladders, because it requires fewer components. The passive networks in Part II (*SB* 2/85, p. 26) are parallel for this reason.

If active filters are used, neither connection has a component count advantage, and the cascade connection becomes the better choice, for two reasons. First, its high-pass channel has a sharper transition to the stopband because the signal passes through two high-pass sections, rather than one as in the parallel connection. Second, it works better than the parallel connection in a constant power crossover (CPC). I use a cascade connection for those reasons.

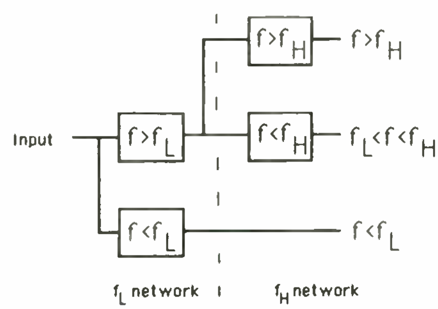

FIGURE 1: The block diagram of the cascade connection of a pair of two-way crossovers, forming a three-way network.

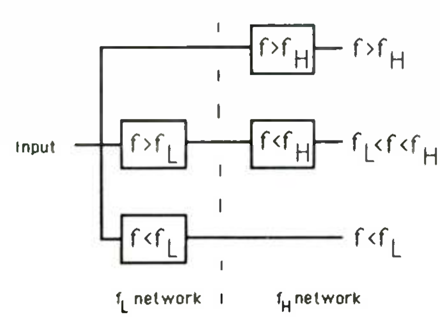

FIGURE 2: The parallel connection of a pair of two-way crossovers to form a three-way crossover.

ACTIVE CPC DESIGN. It is amazingly easy to design a three-way active CPC. First, choose the desired crossover frequencies f_L and f_H. Next refer to Part III of this series (*SB*, 3/85 p. 14) and design one two-way CPC with crossover frequency f_l, and a second with crossover frequency f_H. Finally, complete the three-way design by connecting these two-way stages as in *Fig. 1*. The result is always a three-way CPC, regardless of the orders chosen for each two-way stage, or the polarities observed in connecting the loudspeakers.

ALL-PASS CROSSOVERS. The rest of this article is devoted to the more complicated problem of actively implementing three-way all-pass crossovers (APCs). If you try the CPC procedure, but use two-way APC stages instead, the resulting three-way network is not an APC for either the cascade or the parallel connection. The cascade connection of two-way APC stages can be made to yield a three-way APC if additional filtering is used. But such filtering is difficult to implement using only the rumble filter kit. Rather than deal with extra circuitry, I am going to use a new three-way APC that is formed by cascading two-way stages which are not themselves APCs. It can be easily implemented using only the rumble filter kit.

CROSSOVER ORDER. I am going to deal only with networks of order less than or equal to four, as has been usual in this series. By this I mean each two-way stage of the three-way network is of an order less than or equal to four. Also, to keep the number of possibilities manageable, I deal only with networks where both two-way stages have the same order. This common order I call the order of the overall three-way network. The nth order network has an asymptotic roll off rate of 6ndB per octave for the lowpass and bandpass channels, and 12ndB per octave for the high-pass channel.

Each even order has only one three-way APC, but each odd order has four different networks: therefore one set of design formulas for order two, one set for order four, and four sets of formulas for orders one and three.

RUMBLE FILTER USE. Because each three-way network is thought to consist of a pair of two-way stages, it is sufficient for design purposes to tell you how to do each stage using the rumble filter kit. I do this as in Part III, by giving formulas for the filter shaping component values C1L, C2L, C3L, R1L, R2L, R3L, C1H, C2H, C3H, R1H, R2H, and R3H as shown in *Fig. 3*. The component sizes for each two-way stage depend not only on the crossover frequency involved, but also on the spread S between the two crossover frequencies as defined by

$$S = \frac{f_H}{f_l} \qquad (1)$$

I use the same filter alignment procedure here as I did in Part III. It involves starting with a convenient stock capacitor value (C), which becomes the value for one of the filter shaping capacitors. All other filter shaping capacitors are easily obtainable multiples of C. The resistor values are found by formula from this C, and the radian crossover frequency of the two-way stage under design defined by

$$\omega_l = 2\pi f_l \qquad (2)$$

or

$$\omega_H = 2\pi f_H \qquad (3)$$

GAIN ADJUSTMENTS. To obtain the correct response shape it is sometimes necessary to adjust the gain of some of the filter sections to a value other than one. This is done on the rumble filter board by means of the resistors RIL, RFL, RIH and RFH shown in *Fig. 3*. A convenient value is chosen for the RI, and the RF is then found by formula from RI.

The gain adjustment is needed to change the level of the bandpass channel relative to the low and high-pass channels. If the bandpass gain must be higher, RIL and RFL in the *second* stage are used. If it must be lower, then RIL and FRL in the *first* stage, and RIH and RFH in the second stage, are used. Two different procedures are necessary because the kit's circuit topology permits only gains greater than or equal to one.

FIRST-ORDER APC. First order APC has four possible networks.

Network 1:

STAGE 1

Choose a convenient C, and calculate ω_L from (2). Then use:

C1L = omit	C1H = jumper	(4)
C2L = omit	C2H = jumper	(5)
C3L = C	C3H = C	(6)
R1L = jumper	R1H = omit	(7)
R2L = jumper	R2H = omit	(8)
R3L = $1/(C\omega_l)$	R3H = $1/(C\omega_L)$	(9)

STAGE 2

Choose a convenient C, not necessarily the same as the one chosen for stage 1, and calculate ω_H. Then use formulas (4) through (8) above, together with:

$$R3L = 1/(C\omega_H) \qquad R3H = 1/(C\omega_H) \qquad (10)$$

Now connect the two stages as in *Fig. 1*. Observe polarity in all channels when connecting the loudspeakers.

Network 2:

Proceed exactly as for network 1, but reverse midrange and tweeter polarity when connecting the loudspeakers.

Network 3:

Proceed as for network 1, but use gain setting resistors RFL and RIL in stage 2. That is, choose a convenient value for RIL and let

$$RFL = \frac{2}{S}\ RIL \qquad (11)$$

where S is given by formula (1).

When connecting loudspeakers, reverse the midrange polarity.

Network 4:

Proceed as for network 1, but use the gain setting resistors RIL and RFL in stage 1, and, RIH and RFH in stage 2. For this, choose convenient values for RIL and RIH, and let

$$RFL = \frac{2}{(S-2)}\ RIL \qquad (12)$$

$$RFH = \frac{2}{(S-2)}\ RIH \qquad (13)$$

Finally, reverse the tweeter polarity.

SECOND-ORDER APC. Use S to calculate the first and second stage damping factors A_1 and A_2 from the formulas:

$$A_1 = \sqrt{\frac{4(S^2-1)}{(S^*-2)}} \qquad (14)$$

$$A_2 = S\sqrt{\frac{2(S^2-1)}{(S^2-2)}} \qquad (15)$$

Also calculate the gain factor

$$k = \frac{S^2}{(S_2-2)} \qquad (16)$$

and the two component value factors M_1 and M_2 from:

$$M_1 = \frac{A_1 + \sqrt{A_1^2 - 4}}{2} \qquad (17)$$

$$M_2 = \frac{A_2 + \sqrt{A_2^2 - 4}}{2} \qquad (18)$$

STAGE 1

Choose a convenient C value and calculate ω_L. Then

C1L = omit	C1H = jumper	(19)
C2L = C	C2H = C	(20)
C3L = C	C3H = C	(21)

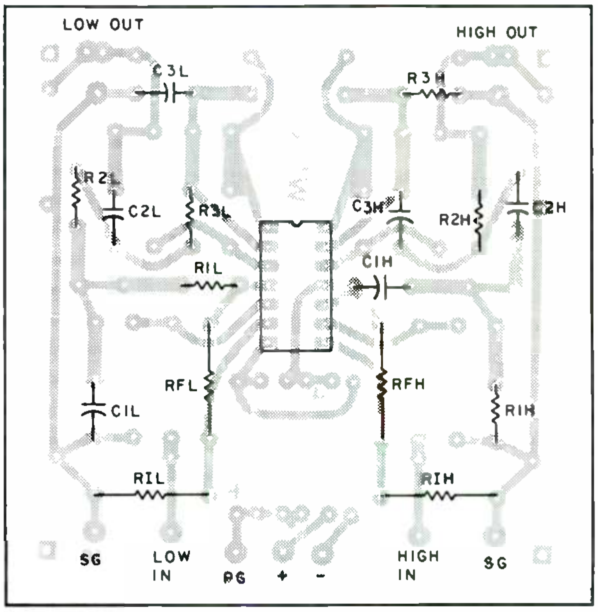

FIGURE 3: The rumble filter board stuffing guide, showing only the filter shaping and gain setting components. The left circuit is low-pass and the right is high-pass.

$R1L = \text{jumper}$ $\quad R1H = \text{omit}$ (22)
$R2L = 1/(C\omega_L M_1)$ $\quad R2H = A_1/(2C\omega_L)$ (23)
$R3L = M_1/(C\omega_L)$ $\quad R3H = 2/(A_1 C\omega_L)$ (24)

STAGE 2

Choose a convenient C and RIL, and calculate ω_H. The component values are:

$C1L = \text{omit}$ $\quad C1H = \text{jumper}$ (25)
$C2L = C$ $\quad C2H = C$ (26)
$C3L = C$ $\quad C3H = C$ (27)
$R1L = \text{jumper}$ $\quad R1H = \text{omit}$ (28)
$R2L = 1/(C\omega_H M_2)$ $\quad R2H = A_2/(2C\omega_H)$ (29)
$R3L = M_2/(C\omega_H)$ $\quad R3H = 2/(A_2 C\omega_H)$ (30)
$RFL = (k-1)RIL$ (31)

Connect the stages as in *Fig. 1* and reverse midrange polarity.

TABLE 1

THIRD-ORDER NORMALIZED ELEMENT VALUES: (STAGE 1) (NETWORK 1)

S	cm	r1L	r2L	r3L	r1H	r2H	r3H	k
3	.033	1.3181	4.2931	5.3549	.7830	.0571	22.3825	1.0136
4	.033	1.3830	2.6375	8.3076	.7660	.1289	10.1265	1.0021
5	.033	1.4078	2.2712	9.4772	.7561	.1648	8.0231	1.0005
6	.033	1.4219	2.1008	10.1448	.7497	.1868	7.1424	1.0002
7	.033	1.4311	2.0010	10.5822	.7451	.2016	6.6557	1.0001
8	.100	1.1742	4.0807	2.0870	.7417	.2124	6.3461	1.0000
9	.100	1.1400	4.8439	1.8108	.7391	.2207	6.1315	1.0000
10	.100	1.1242	5.2085	1.7078	.7370	.2271	5.9740	1.0000
11	.100	1.1132	5.4635	1.6443	.7353	.2324	5.8533	1.0000
12	.100	1.1047	5.6581	1.5998	.7338	.2367	5.7579	1.0000
13	.100	1.0980	5.8134	1.5666	.7326	.2403	5.6805	1.0000
14	.100	1.0925	5.9411	1.5407	.7316	.2434	5.6166	1.0000
15	.100	1.0878	6.0484	1.5199	.7307	.2460	5.5628	1.0000
16	.100	1.0838	6.1399	1.5027	.7299	.2483	5.5169	1.0000
17	.100	1.0804	6.2190	1.4884	.7292	.2504	5.4774	1.0000
18	.100	1.0773	6.2882	1.4761	.7286	.2522	5.4429	1.0000
19	.100	1.0746	6.3493	1.4656	.7280	.2538	5.4125	1.0000
20	.100	1.0723	6.4036	1.4564	.7275	.2552	5.3856	1.0000

HIGHER ORDERS. For orders three and four, the filter shaping component values cannot be expressed directly in terms of S because the relevant equations cannot be solved in closed form. What I have done is to numerically solve a normalized version of the system of equations, for a range of S values, and to tabulate the normalized component values in *Tables 1* through *10*. These normalized values are then used to calculate the actual component values.

The limitation of this procedure is that only those spreads S that appear in the tables can be accommodated, namely integer values between 2 or 3, and 20. Thus, you may have to shift your intended crossover frequencies slightly. For example, if you had planned to use $f_L = 500$ and $f_H = 2250$, you are out of luck, because $S = 5.625$ is not an integer. But you could use $f_H = 2000$ or 2400 instead, and obtain an S of 5 or 6, respectively. The latter choice would be closer to your intended value.

The table entries are labeled the same as the filter shaping components, except that lower case "r"s and "c"s are used. The actual component values (upper case) are obtained by "denormalizing" these lower case values through multiplication or division by certain factors. The formulas presented below show you explicitly what those factors are.

Each table has a column labeled "cm," for "capacitor multiplier." It is used to calculate one of the capacitor values from the selected value of C. Also, some tables have a column labeled "k." This column appears only when gain is needed in the stage, and is used to calculate RF from the chosen value of RI (ie., $RF = (k-1)RI$).

THIRD-ORDER APC. Third-order APC has four possible networks. Start with an integer S value between 3 and 20.

Network 1:

STAGE 1

Choose convenient values for C and RIL, and determine ω_L. Then use *Table 1* to calculate:

$C1L = C$ $\quad C1H = C$ (32)
$C2L = C$ $\quad C2H = C$ (33)
$C3L = (cm)C$ $\quad C3H = C$ (34)
$R1L = (r1L)/(C\omega_L)$ $\quad R1H = (r1H)/(C\omega_L)$ (35)
$R2L = (r2L)/(C\omega_L)$ $\quad R2H = (r2H)/(C\omega_L)$ (36)
$R3L = (r3L)/(C\omega_L)$ $\quad R3H = (r2H)/(C\omega_L)$ (37)
$RFL = (k-1)RIL$ (38)

STAGE 2

Choose convenient values for C and RIH, and calculate ω_H. Calculate component values using *Table 2* and formulas (32) through (37) above, with ω_H replacing ω_L, and

$$RFH = (k-1)RIH \qquad (39)$$

Connect the two stages as in *Fig. 1* and observe all loudspeaker polarities.

Network 2:

STAGE 1

Choose a C, calculate ω_L and use *Table 3* with formulas (32) through (37).

STAGE 2

Choose C and RIL, calculate ω_H, and use *Table 4* with Formulas (32) through (38), but replace ω_L with ω_H.

TABLE 2

THIRD-ORDER NORMALIZED ELEMENT VALUES: (STAGE 2) (NETWORK 1)

S	cm	r1L	r2L	r3L	r1H	r2H	r3H	k
3	.033	1.5421	1.4781	13.2945	.7212	.3190	4.3462	1.0136
4	.033	1.5027	1.5510	13.0015	.7190	.2972	4.6794	1.0021
5	.033	1.4897	1.5781	12.8894	.7185	.2897	4.8048	1.0005
6	.033	1.4842	1.5902	12.8387	.7183	.2864	4.8612	1.0002
7	.033	1.4815	1.5964	12.8128	.7182	.2847	4.8900	1.0001
8	.100	1.0292	7.4062	1.3119	.7182	.2838	4.9063	1.0000
9	.100	1.0294	7.3893	1.3146	.7181	.2833	4.9162	1.0000
10	.100	1.0296	7.3784	1.3164	.7181	.2829	4.9225	1.0000
11	.100	1.0297	7.3712	1.3175	.7181	.2827	4.9267	1.0000
12	.100	1.0297	7.3662	1.3184	.7181	.2825	4.9296	1.0000
13	.100	1.0298	7.3626	1.3189	.7181	.2824	4.9317	1.0000
14	.100	1.0298	7.3600	1.3194	.7181	.2823	4.9333	1.0000
15	.100	1.0298	7.3580	1.3197	.7181	.2822	4.9344	1.0000
16	.100	1.0299	7.3565	1.3199	.7181	.2822	4.9353	1.0000
17	.100	1.0299	7.3553	1.3201	.7181	.2821	4.9360	1.0000
18	.100	1.0299	7.3544	1.3203	.7181	.2821	4.9365	1.0000
19	.100	1.0299	7.3536	1.3204	.7181	.2821	4.9370	1.0000
20	.100	1.0299	7.3530	1.3205	.7181	.2821	4.9373	1.0000

TABLE 3

THIRD-ORDER NORMALIZED ELEMENT VALUES: (STAGE 1) (NETWORK 2)

S	cm	r1L	r2L	r3L	r1H	r2H	r3H
3	.100	.7667	12.5562	1.0388	.6475	.4540	3.4022
4	.100	.8387	11.2081	1.0638	.6685	.4063	3.6811
5	.100	.8778	10.4632	1.0888	.6793	.3811	3.8628
6	.100	.9031	9.9726	1.1104	.6860	.3648	3.9958
7	.100	.9209	9.6206	1.1287	.6907	.3532	4.0985
8	.100	.9343	9.3543	1.1442	.6942	.3446	4.1806
9	.100	.9447	9.1451	1.1575	.6969	.3378	4.2479
10	.100	.9530	8.9762	1.1690	.6990	.3324	4.3041
11	.100	.9598	8.8368	1.1790	.7008	.3279	4.3518
12	.100	.9655	8.7196	1.1878	.7022	.3242	4.3928
13	.100	.9703	8.6198	1.1956	.7034	.3210	4.4285
14	.100	.9745	8.5337	1.2025	.7045	.3183	4.4597
15	.100	.9781	8.4586	1.2087	.7054	.3159	4.4874
16	.100	.9813	8.3925	1.2143	.7062	.3138	4.5120
17	.100	.9841	8.3339	1.2194	.7069	.3120	4.5341
18	.100	.9865	8.2816	1.2240	.7075	.3104	4.5540
19	.100	.9888	8.2346	1.2282	.7081	.3089	4.5721
20	.100	.9908	8.1921	1.2320	.7086	.3076	4.5885

TABLE 4

THIRD-ORDER NORMALIZED ELEMENT VALUES: (STAGE 2) (NETWORK 2)

S	cm	r1L	r2L	r3L	r1H	r2H	r3H	k
3	.100	1.0837	7.1293	1.2944	.7371	.2801	4.8441	1.0920
4	.100	1.0503	7.2537	1.3126	.7252	.2807	4.9115	1.0340
5	.100	1.0400	7.2998	1.3172	.7216	.2813	4.9271	1.0167
6	.100	1.0357	7.3205	1.3189	.7201	.2815	4.9328	1.0095
7	.100	1.0335	7.3312	1.3198	.7193	.2817	4.9354	1.0059
8	.100	1.0324	7.3372	1.3202	.7189	.2818	4.9368	1.0039
9	.100	1.0316	7.3408	1.3205	.7186	.2818	4.9376	1.0028
10	.100	1.0312	7.3431	1.3206	.7185	.2819	4.9381	1.0020
11	.100	1.0309	7.3447	1.3208	.7184	.2819	4.9385	1.0015
12	.100	1.0307	7.3457	1.3208	.7183	.2819	4.9387	1.0012
13	.100	1.0305	7.3465	1.3209	.7183	.2819	4.9389	1.0009
14	.100	1.0304	7.3471	1.3209	.7182	.2819	4.9390	1.0007
15	.100	1.0303	7.3475	1.3210	.7182	.2819	4.9391	1.0006
16	.100	1.0303	7.3478	1.3210	.7182	.2819	4.9391	1.0005
17	.100	1.0302	7.3481	1.3210	.7181	.2819	4.9392	1.0004
18	.100	1.0302	7.3483	1.3210	.7181	.2819	4.9392	1.0003
19	.100	1.0301	7.3485	1.3210	.7181	.2819	4.9393	1.0003
20	.100	1.0301	7.3486	1.3210	.7181	.2819	4.9393	1.0003

TABLE 5

THIRD-ORDER NORMALIZED ELEMENT VALUES: (STAGE 1) (NETWORK 3)

S	cm	r1L	r2L	r3L	r1H	r2H	r3H	k
3	.033	1.3482	3.3570	6.6954	.7767	.0855	15.0515	1.0673
4	.033	1.3891	2.5380	8.5951	.7637	.1374	9.5269	1.0299
5	.033	1.4100	2.2431	9.5811	.7552	.1682	7.8735	1.0156
6	.033	1.4229	2.0897	10.1914	.7492	.1883	7.0874	1.0091
7	.033	1.4316	1.9958	10.6062	.7449	.2025	6.6308	1.0058
8	.100	1.1701	4.1685	2.0501	.7416	.2129	6.3332	1.0039
9	.100	1.1392	4.8623	1.8053	.7390	.2210	6.1242	1.0027
10	.100	1.1238	5.2183	1.7053	.7369	.2273	5.9694	1.0020
11	.100	1.1129	5.4695	1.6428	.7352	.2325	5.8504	1.0015
12	.100	1.1046	5.6620	1.5990	.7338	.2368	5.7559	1.0012
13	.100	1.0979	5.8162	1.5660	.7326	.2404	5.6792	1.0009
14	.100	1.0924	5.9431	1.5403	.7316	.2434	5.6156	1.0007
15	.100	1.0877	6.0498	1.5196	.7307	.2461	5.5621	1.0006
16	.100	1.0838	6.1410	1.5025	.7299	.2484	5.5164	1.0005
17	.100	1.0803	6.2199	1.4882	.7292	.2504	5.4769	1.0004
18	.100	1.0773	6.2889	1.4760	.7286	.2522	5.4425	1.0003
19	.100	1.0746	6.3498	1.4655	.7280	.2538	5.4123	1.0003
20	.100	1.0722	6.4040	1.4563	.7275	.2552	5.3854	1.0002

Connect stages as in *Fig. 1* and reverse polarity of the midrange and tweeter.

Network 3:

STAGE 1

Choose C and RIL, calculate ω_L, and use *Table 5* with formulas (32) through (38).

STAGE 2

Choose C and RIH, calculate ω_H, use *Table 6* and formulas (32) through (37) with ω_H replacing ω_L, and formula (39).

Connect stages as in *Fig. 1* and reverse tweeter polarity.

Network 4:

STAGE 1

Choose C, calculate ω_L and use *Table 7* with formulas (32) through (37).

STAGE 2

Choose C and RIL, calculate ω_H, use *Table 8* with formulas (32) through (38), but replace ω_L with ω_H.

Connect stages as in *Fig. 1* and reverse polarity of midrange.

FOURTH-ORDER APC. Since the rumble filter kit is limited at most to third-order filters, two kits are needed for each stage. So, the formulas for each stage are divided into two groups, one for stuffing board 1, and the other for board 2. Once the four boards are completed, they are connected as in *Fig. 4*. Luckily, there is only one possible network.

STAGE 1, BOARD 1

Choose C, calculate ω_L and use *Table 9* with the following formulas:

C1L = omit	C1H = jumper	(40)
C2L = (cm)C	C2H = C	(41)
C3L = C	C3H = C	(42)
R1L = jumper	R1H = omit	(43)
R2L = (r2L)/(Cω_L)	R2H = (r2H)/(Cω_L)	(44)
R3L = (r3L)/(Cω_L)	R3H = (r3H)/(Cω_L)	(45)

STAGE 1, BOARD 2

Choose C and RIL, let ω_L be as in board 1, and use *Table 10* with formulas (40) through (45) and

$$RFL = (k-1)RIL \qquad (46)$$

STAGE 2, BOARD 1

Choose C, and calculate ω_H, and use *Table 11* with formulas (40) through (45), replacing ω_L with ω_H.

STAGE 2, BOARD 2

Choose C and RIH, let ω_H be as in board 1, use *Table 12* with formulas (40) through (45), replace ω_L with ω_H, and let

$$RFH = (k-1)RIH \qquad (47)$$

Observe all loudspeaker polarities.

MORE ON GAIN. If your crossover network is to have a large spread between crossover frequencies, more often than not the gain setting resistors can be eliminated. As a matter of fact, whenever $k=1$ in the tables, this specifies $RF=0$. The gain is then unity, regardless of the value for RI, so neither resistor is needed. For example, in the third-order network 1, as long as $S \geq 8$, the computed RF is zero. Thus, the gain circuit is unnecessary. But even more, I would dispense with the gain circuit whenever the value of RF turns out to be less than p percent of RI, when p percent tolerance resistors are used. For example, I would use the gain circuits in the fourth-order network only for $S=2$, with 1 percent tolerance resistors, because the value of RF is less than 1 percent of RI for all other spreads.

VARIABLE GAIN CONTROLS. It is relatively easy to add variable gain to each channel of the network, as I briefly described in Part III. All that is needed is a potentiometer for an RF component in each channel. If a fixed RF is already required to set relative gains, just put the pot in series with it so the relative level is correct when the pot is set to minimum. Remember, the maximum possible gain is given by

$$G_{MAX} = 20 \log \left(1 + \frac{RF_{MAX}}{RI}\right)$$

where RF_{MAX} is the rated resistance of the pot, plus any fixed RF used. This formula is a good way to match channel sensitivities, but be careful how much gain you actually apply because it is possible to overdrive subsequent stages.

RESISTOR SPREADS. It is a good practice to avoid large spreads between element values. For example, suppose you want a third-order network with $f_L = 500$ and $f_H = 1.5k$. This yields $S=3$. Now, if you insist on observing all loudspeaker polarities, then Network 1 must be chosen. But

Continued on page 26

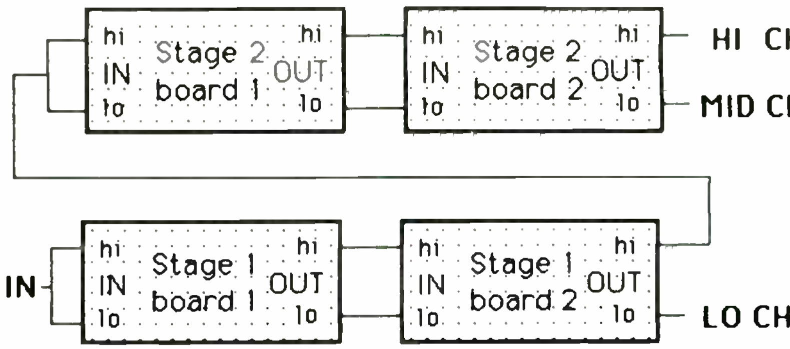

FIGURE 4: How to connect rumble filter boards for a fourth-order three-way crossover network.

TABLE 6

THIRD-ORDER NORMALIZED ELEMENT VALUES: (STAGE 2) (NETWORK 3)

S	cm	r1L	r2L	r3L	r1H	r2H	r3H	k
3	.033	1.4472	1.5893	13.1752	.7044	.2866	4.9524	1.0673
4	.033	1.4627	1.6005	12.9443	.7118	.2836	4.9531	1.0299
5	.033	1.4693	1.6039	12.8593	.7148	.2827	4.9481	1.0156
6	.033	1.4724	1.6052	12.8212	.7161	.2824	4.9449	1.0091
7	.033	1.4740	1.6059	12.8017	.7168	.2822	4.9430	1.0058
8	.100	1.0276	7.3617	1.3219	.7172	.2821	4.9419	1.0039
9	.100	1.0283	7.3580	1.3217	.7175	.2821	4.9412	1.0027
10	.100	1.0287	7.3557	1.3215	.7176	.2820	4.9407	1.0020
11	.100	1.0290	7.3541	1.3214	.7177	.2820	4.9404	1.0015
12	.100	1.0293	7.3530	1.3213	.7178	.2820	4.9402	1.0012
13	.100	1.0294	7.3522	1.3213	.7179	.2820	4.9401	1.0009
14	.100	1.0295	7.3517	1.3212	.7179	.2820	4.9399	1.0007
15	.100	1.0296	7.3512	1.3212	.7179	.2820	4.9398	1.0006
16	.100	1.0297	7.3509	1.3212	.7180	.2820	4.9398	1.0005
17	.100	1.0297	7.3507	1.3212	.7180	.2820	4.9397	1.0004
18	.100	1.0298	7.3505	1.3212	.7180	.2820	4.9397	1.0003
19	.100	1.0298	7.3503	1.3211	.7180	.2820	4.9397	1.0003
20	.100	1.0298	7.3502	1.3211	.7180	.2820	4.9396	1.0002

TABLE 7

THIRD-ORDER NORMALIZED ELEMENT VALUES: (STAGE 1) (NETWORK 4)

S	cm	r1L	r2L	r3L	r1H	r2H	r3H
3	.100	.7990	11.9538	1.0470	.6572	.4323	3.5198
4	.100	.8485	11.0232	1.0692	.6713	.4000	3.7241
5	.100	.8817	10.3870	1.0919	.6803	.3786	3.8827
6	.100	.9050	9.9353	1.1122	.6865	.3636	4.0063
7	.100	.9219	9.6002	1.1298	.6910	.3526	4.1046
8	.100	.9349	9.3421	1.1450	.6944	.3442	4.1844
9	.100	.9450	9.1374	1.1581	.6970	.3376	4.2504
10	.100	.9532	8.9711	1.1694	.6991	.3322	4.3058
11	.100	.9600	8.8333	1.1793	.7008	.3278	4.3530
12	.100	.9656	8.7172	1.1880	.7023	.3241	4.3937
13	.100	.9704	8.6180	1.1957	.7035	.3209	4.4291
14	.100	.9746	8.5323	1.2026	.7045	.3182	4.4602
15	.100	.9781	8.4575	1.2088	.7054	.3159	4.4878
16	.100	.9813	8.3917	1.2144	.7062	.3138	4.5123
17	.100	.9841	8.3333	1.2194	.7069	.3120	4.5343
18	.100	.9866	8.2811	1.2240	.7075	.3103	4.5542
19	.100	.9888	8.2342	1.2282	.7081	.3089	4.5722
20	.100	.9908	8.1918	1.2321	.7086	.3076	4.5886

TABLE 8

THIRD-ORDER NORMALIZED ELEMENT VALUES: (STAGE 2) (NETWORK 4)

S	cm	r1L	r2L	r3L	r1H	r2H	r3H	k
3	.100	1.0494	6.2729	1.5191	.7176	.2492	5.5924	1.0087
4	.100	1.0370	6.8957	1.3985	.7176	.2675	5.2100	1.0018
5	.100	1.0333	7.1169	1.3598	.7177	.2744	5.0769	1.0005
6	.100	1.0319	7.2148	1.3432	.7179	.2776	5.0187	1.0002
7	.100	1.0311	7.2646	1.3350	.7179	.2792	4.9892	1.0001
8	.100	1.0307	7.2926	1.3304	.7180	.2801	4.9728	1.0000
9	.100	1.0305	7.3095	1.3276	.7180	.2806	4.9628	1.0000
10	.100	1.0304	7.3203	1.3258	.7180	.2810	4.9565	1.0000
11	.100	1.0303	7.3275	1.3246	.7180	.2812	4.9523	1.0000
12	.100	1.0302	7.3326	1.3238	.7180	.2814	4.9493	1.0000
13	.100	1.0301	7.3361	1.3232	.7180	.2815	4.9472	1.0000
14	.100	1.0301	7.3388	1.3228	.7180	.2816	4.9457	1.0000
15	.100	1.0301	7.3408	1.3225	.7180	.2817	4.9445	1.0000
16	.100	1.0301	7.3423	1.3222	.7180	.2817	4.9436	1.0000
17	.100	1.0300	7.3435	1.3220	.7180	.2817	4.9429	1.0000
18	.100	1.0300	7.3444	1.3219	.7180	.2818	4.9424	1.0000
19	.100	1.0300	7.3451	1.3218	.7181	.2818	4.9419	1.0000
20	.100	1.0300	7.3457	1.3217	.7181	.2818	4.9416	1.0000

TABLE 9

FOURTH-ORDER NORMALIZED ELEMENT VALUES: (STAGE 1) (BOARD 1)

S	cm	r2L	r3L	r2H	r3H
2	10	.0792	1.9518	1.0155	1.5227
3	10	.0779	1.7730	.9254	1.4919
4	10	.0765	1.6703	.8734	1.4628
5	10	.0758	1.6074	.8416	1.4472
6	10	.0754	1.5648	.8201	1.4381
7	10	.0751	1.5339	.8045	1.4323
8	10	.0750	1.5105	.7927	1.4283
9	10	.0749	1.4921	.7835	1.4256
10	10	.0748	1.4773	.7760	1.4235
11	10	.0747	1.4651	.7699	1.4220
12	10	.0747	1.4549	.7648	1.4208
13	10	.0747	1.4462	.7605	1.4199
14	10	.0746	1.4388	.7567	1.4191
15	10	.0746	1.4323	.7535	1.4185
16	10	.0746	1.4266	.7506	1.4180
17	10	.0746	1.4216	.7481	1.4176
18	10	.0746	1.4171	.7458	1.4173
19	10	.0746	1.4131	.7438	1.4170
20	10	.0746	1.4095	.7420	1.4167

TABLE 10

FOURTH-ORDER NORMALIZED ELEMENT VALUES: (STAGE 1) (BOARD 2)

S	cm	r2L	r3L	r2H	r3H	k
2	10	.2319	.2789	.2554	2.5322	1.1471
3	10	.0912	.7944	.4428	1.6357	1.0053
4	10	.0822	.9519	.5171	1.5137	1.0005
5	10	.0792	1.0365	.5579	1.4717	1.0001
6	10	.0778	1.0903	.5840	1.4519	1.0000
7	10	.0770	1.1277	.6023	1.4408	1.0000
8	10	.0764	1.1553	.6159	1.4340	1.0000
9	10	.0761	1.1765	.6263	1.4295	1.0000
10	10	.0759	1.1933	.6346	1.4264	1.0000
11	10	.0757	1.2070	.6413	1.4242	1.0000
12	10	.0755	1.2183	.6469	1.4225	1.0000
13	10	.0754	1.2279	.6517	1.4212	1.0000
14	10	.0753	1.2360	.6557	1.4202	1.0000
15	10	.0753	1.2431	.6592	1.4194	1.0000
16	10	.0752	1.2492	.6622	1.4187	1.0000
17	10	.0752	1.2546	.6649	1.4182	1.0000
18	10	.0751	1.2594	.6673	1.4177	1.0000
19	10	.0751	1.2637	.6694	1.4174	1.0000
20	10	.0750	1.2676	.6713	1.4171	1.0000

from *Table 1* the first stage requires R2H = 190Ω and R3H = 7kΩ if C = 0.001 μF. I would prefer to avoid such a wide spread by using Network 2, where R2H = 1.5kΩ and R3H = 10kΩ for the same C.

OTHER NETWORKS. With these four articles you should have enough information to passively or actively implement what I think of as the "customary" networks for loudspeaker systems. You should be able to use them to obtain performance that rivals good commercial systems, if your complete design is well concieved and executed.

Many other "non-customary" crossover network designs can be found in references 4, 5, 6 and 7, but most of them that offer obvious performance advantages also require active filtering, which is difficult for the home builder to implement. If you are interested in these networks, I refer you to the references.

TABLE 11

FOURTH-ORDER NORMALIZED ELEMENT VALUES: (STAGE 2) (BOARD 1)

S	cm	r2L	r3L	r2H	r3H
2	10	.0773	1.3863	.7318	1.4636
3	10	.0747	1.3413	.7080	1.4161
4	10	.0747	1.3397	.7072	1.4144
5	10	.0747	1.3395	.7071	1.4143
6	10	.0747	1.3393	.7070	1.4145
7	10	.0746	1.3400	.7073	1.4138
8	10	.0746	1.3401	.7074	1.4137
9	10	.0746	1.3398	.7072	1.4140
10	10	.0746	1.3397	.7072	1.4141
11	10	.0746	1.3396	.7071	1.4141
12	10	.0746	1.3395	.7071	1.4140
13	10	.0746	1.3395	.7071	1.4140
14	10	.0746	1.3395	.7071	1.4140
15	10	.0746	1.3396	.7071	1.4142
16	10	.0747	1.3396	.7071	1.4143
17	10	.0747	1.3396	.7071	1.4143
18	10	.0747	1.3396	.7071	1.4142
19	10	.0746	1.3395	.7071	1.4141
20	10	.0746	1.3395	.7071	1.4141

TABLE 12

FOURTH-ORDER NORMALIZED ELEMENT VALUES: (STAGE 2) (BOARD 2)

S	cm	r2L	r3L	r2H	r3H	k
2	10	.0641	1.4562	.7601	1.2283	1.1471
3	10	.0725	1.3753	.7239	1.3778	1.0053
4	10	.0740	1.3512	.7126	1.4030	1.0005
5	10	.0744	1.3444	.7094	1.4096	1.0001
6	10	.0745	1.3421	.7083	1.4118	1.0000
7	10	.0746	1.3404	.7075	1.4134	1.0000
8	10	.0746	1.3398	.7072	1.4140	1.0000
9	10	.0746	1.3398	.7072	1.4140	1.0000
10	10	.0746	1.3397	.7072	1.4140	1.0000
11	10	.0746	1.3397	.7072	1.4142	1.0000
12	10	.0747	1.3397	.7072	1.4143	1.0000
13	10	.0747	1.3397	.7072	1.4143	1.0000
14	10	.0747	1.3397	.7072	1.4143	1.0000
15	10	.0747	1.3396	.7071	1.4142	1.0000
16	10	.0746	1.3395	.7071	1.4141	1.0000
17	10	.0746	1.3395	.7071	1.4141	1.0000
18	10	.0746	1.3395	.7071	1.4142	1.0000
19	10	.0747	1.3397	.7072	1.4143	1.0000
20	10	.0747	1.3396	.7072	1.4143	1.0000
21	10	.0747	1.3397	.7072	1.4143	1.0000

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BUILDING PENTAGONAL ENCLOSURES

BY DANIEL PATRICK COYLE

Pentagonal enclosures have several properties which should endear them to speaker builders. Their appearance is attractive and their acoustical qualities are more neutral than rectilinear boxes. Five sided enclosures' internal reflections present few surfaces to support internal standing waves, as only the top and bottom panels are parallel. The external shape has gentle corners which lessen diffraction effects, and while a box has 90 degree corners, a pentagon has 72 degree corners: 40 percent of the preferred 45 degree corner.

Another useful property of the pentagonal prism shape is that for a given internal volume it has narrower faces than a box, so there is less of an unsupported panel to flex between the edges. This causes the resonance of the panel to be higher in frequency (and therefore more easily damped), and lowers the amplitude of flexure.

WOOFER PLACEMENT. Five sided enclosures are amenable to a configuration which enables drivers to be mounted more compactly in the vertical dimension. In this deployment, the high and mid frequency drivers are positioned on the front face as usual, but the bass drivers are located on the adjacent sides of the enclosure.

For bass frequencies it is usually better to use two smaller drivers of an equivalent, or slightly larger, total area instead of a single driver. Two individual cones are each lighter, and as a result more linear in their response. The only problem with this advice is when the woofers are arrayed vertically. Then the resulting enlargement of the sound source strays from the ideal of a point source. If the two bass drivers are positioned on each of the side panels (*Fig. 1*) they will radiate as one driver at low frequencies, yet the total vertical height of the speaker array need be no more than that dictated by the dimensions of the mid and treble drivers.

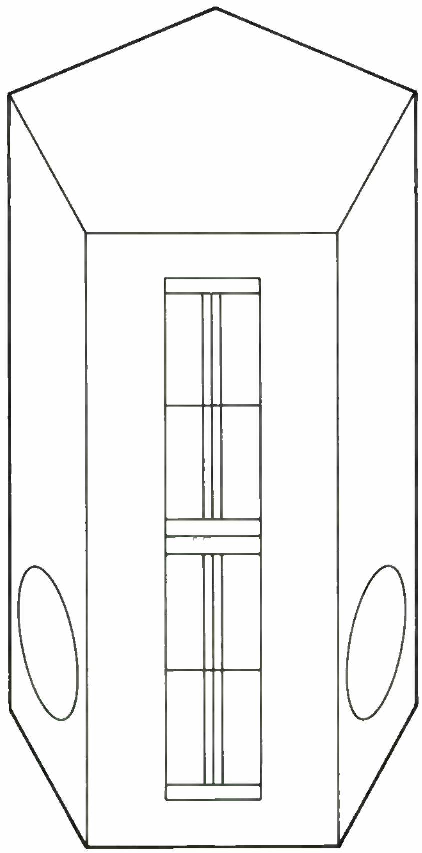

FIGURE 1: Bass drivers mounted on side panels will radiate as one at low frequencies.

Several manufacturers make use of this design, including KEF and AR. This configuration is also useful for a D'Appolito-type configuration (*SB* 4/84, p. 7)

In a previous article (*SB* 3/84, p. 20) I pointed out that drivers will give uniform sound distribution, ka < 2, below a frequency corresponding to the constant 8552, divided by the actual diameter of the driver in inches. For example, two drivers whose outside edges are 21 inches apart (a large cabinet) act as one driver below the frequency corresponding to 8552 ÷ 21, or 407Hz.

CUTTING THE PANELS. Five sided boxes are easily cut on a table saw. The angle of the blade is adjusted to 72 degrees instead of 90 degrees. In practice, the two edges of the panels need not have the same angle, but only angles which together total 144 degrees. [*See* SB *2/84, pp. 12-13 for Bruce Edgar's tips on cutting angles with a table saw.—Ed.]*

If you find after cutting, and a trial fit, your corner butt joints are not touching across their full length, remove the amount needed to ensure a flush joint from only one edge on the panel. The broken lines in *Fig. 2* show possible adjustments. Recutting just one edge is more accurate than trying to trim both edges.

DETERMINING VOLUME. The formula for finding the internal volume of a pentagonal enclosure is as follows:

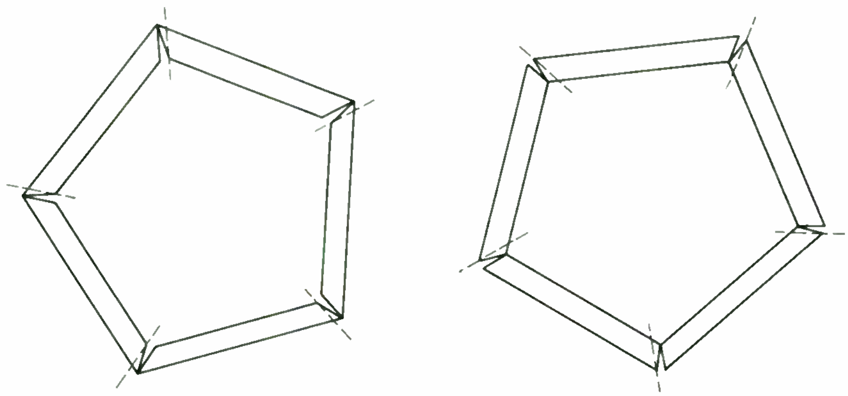

FIGURE 2: Exaggerated examples of miter cuts needed for flush butt joints. Broken lines indicate proper adjustments.

$1.72(F^2)(H) \div 1728$ = Volume in cubic feet

where F equals the internal face in inches, and H equals the height in inches. For example, consider an enclosure with a 12-inch internal face, which is 48 inches high. Its volume is 6.88 cubic feet. ($12 \times 1.72 \times 12 \times 48 \div 1728 = 6.88$.)

To estimate the overall width of this enclosure, multiply the internal face by 1.54 and add 1.7 inches, which is the width of two pieces of ¾-inch particle board in the pentagonal configuration (*Fig. 3*). The above example of a 12-inch internal face gives an overall width of (12×1.54) + 1.7 inches = 20.2 inches. Actually, this width is reduced if you round the enclosure edges after construction to further diminish diffraction effects.

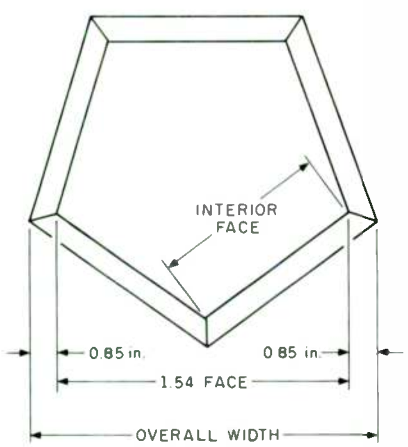

FIGURE 3: The proportional relationships of pentagonal enclosures used to measure volume.

CORNERS. I have made enclosures for 18 years, with and without wood screws in the corners. I have found if the joints are tight, if the glue blocks are properly fitted and their size is at least the thickness of the panels being joined, then the corners do not fail or vibrate.

I am a luthier and can assure you there are no wood screws in any stringed instruments, only glue blocks, yet plate junctions are rigid and behave as nodes.

ASSEMBLY. I have found the easiest way to test-fit and glue the panels is to hold them together with elastic Bungey cords placed around the cabinet's circumference. By jostling the panels while they are held together by the elastic, they will naturally find their closest and most symmetrical position. Panel bottoms will also even out with the surface of the table you are working on.

A five sided enclosure is most easily assembled by gluing the vertical panels first. With the sides assembled, without glue, just clamped by the elastic, I separate two panels at a time by inserting a scrap between them. Now applying glue to the edges is easy. Work quickly because of the large amount of glue to be applied. When the scrap is removed, the panels spring together. After the panels are glued, clean off the excess glue squeezed from the inner part of the joint.

When the joints have dried for 24-hours, I turn the enclosure on its side and add glue blocks to the panel corners (*Fig. 4*). Allow these glue blocks to set up (one hour with a Titebond type glue), rotate the cabinet, apply two more glue blocks to the next set of junctions and so on.

Glue block angles must be cut to closely match the angle of the surfaces they are to retain. I find it helpful to scrape ⅛-inch from the glue block's edge. This prevents any hardened glue left in the corner from displacing the glue block's bond to the joint.

DRIVER MOUNTING. I use a bead of silicone to fasten all the drivers to the enclosure. Then if I need to remove a driver, I can do so easily by inserting a thin blade knife between the driver and the cabinet and gently work it around the frame.

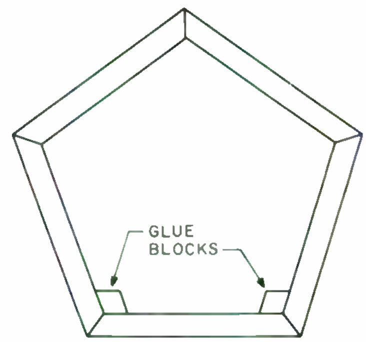

FIGURE 4: Position of the first two glue blocks. Be sure the cabinet lies on a flat, smooth surface.

In my experience silicone has held cast frame 15-inch bass musical instrument drivers in place indefinitely, even though they were transported nightly. Silicone provides an air-tight, resilient mounting and is inexpensive. The only caveat I have is to let it cure for a full 24-hours before severe jostling. Silicone seems to take more than its advertised time to set completely.

Many people find the appearance of the pentagonal enclosure "box" more visually interesting than a rectangular enclosure. I find them pleasing because they satisfy the Greek dictum that form should follow function.

ABOUT THE AUTHOR

After playing rock music and constructing electric guitars and basses (for Bo Diddley, among others), Daniel Patrick Coyle is now an electrical engineering undergraduate at the University of New Mexico at Albuquerque. He is most interested in loudspeakers, and is a student member of the Audio Engineering Society and the Acoustical Society of America.

Kit Report

Audio Concepts' Model G

Whenever friends ask me to recommend a speaker system, I first ask them two questions: "How much are you willing to spend?" and "What kind of music do you listen to most?" If the answers are low cost and rock 'n roll, then I would suggest they build Audio Concepts' Model G loudspeakers (*Fig. 1*). The kit is easy to assemble for a first time kit builder.

Kit Contents

As with most of Audio Concepts' kits, the Model G comes in two versions: the basic kit ($199), which includes drivers, crossover assembly, midrange subchamber and acoustic foam stuffing for a pair of loudspeakers; and the complete kit ($349), which includes the above plus a pair of enclosures. *Figure 2* shows one speaker of the complete kit.

The Model G is a three-way system using Seas drivers. The ten-inch woofer is a Seas P25REX, with a cast metal frame and polypropylene cone. The one-inch tweeter is a soft dome, ferrofluid Seas H253. Until recently Audio Concepts provided a Vifa M10 MD-5 for the midrange. They now use an almost identical four-inch Seas 10F-M, with a cast metal chassis and specially coated paper cone. The midrange design uses a parabolic shaped plastic enclosure.

The crossover is well designed while keeping the cost down and maintaining quality. As with most of Audio Concepts' systems, the Model G's pre-assembled crossover eases kit construction and uses first-order filters with phase compensation (*Fig. 3*). Five-way binding posts are mounted in the center of a piece of fiberboard, and the metallized polyester capacitors and air-core inductors are glued on the back with silicone adhesive. The component leads are soldered directly together, and to the binding posts. The wires connecting the crossover to the drivers are soldered to the circuit, and their free ends are terminated with push-on quick-disconnect lugs.

FIGURE 1: The Model G loudspeaker was introduced by Audio Concepts as the bigger younger brother to the Model C.

FIGURE 2: The kit includes these materials for each speaker. This is the complete kit, which is also sold without the enclosure.

Many of Audio Concepts' kits use an "aperiodic" enclosure. Not being familiar with this type of enclosure, I asked the company for some background information. Aperiodic is Dynaudio's term for a pressure release system that introduces acoustic resistance into the enclosure in order to lower the amplitude of the resonant-frequency impedance peak of the woofer. The oscillation of the woofer cone is aperiodically damped when pulsed, which results in a more precise bass response. The aperiodic concept is not new: Dynaco loudspeakers used it many years ago.

Dynaudio also produce "Variovent," a plastic damping material to be used in aperiodic enclosures. Variovent is not used in the Model G; rather, pressure release is accomplished by providing several small holes in the rear panel of an otherwise sealed enclosure.

The preassembled aperiodic enclosures are unfinished, real wood veneer, and come complete with snap-on grilles. The cabinets have all the necessary cutouts ready for driver installation. Their internal volume is two cubic feet, with external dimensions of 28½ by 14½ by 13 inches.

Assembly

Since I built the complete kit with assembled enclosures, I will not discuss enclosure construction. If you choose to buy the basic kit, and build your own enclosures, Audio Concepts provides a detailed drawing to guide you. Assembling the complete kit is a simple operation, requiring only hand tools and some basic material.

First, run a bead of silicone around the cut-out in the rear of the enclosure and press the crossover assembly in place. Then attach the acoustic foam to the interior walls with a small amount of adhesive. Make sure the foam side pieces are snug against the back panel, and covering the pressure release holes to provide the required acoustic resistance. Allow the adhesive to set overnight before sealing the enclosure.

Next, install the midrange assembly and connect it to the crossover assembly. Insulate the quick disconnects with electrical tape, and attach the wires to the drivers. Now is a good time to connect and check the system by playing it softly. Next, run a bead of rope caulk around each driver hole, set the drivers in place and secure them with screws. Glue the foam diffraction ring into place around the tweeter as shown in *Fig. 1*.

The final step is finishing the wood veneer. I have used Watco brand Danish Finishing Oil on many projects over the years and highly recommend it for finishing the Model G.

Lessons Learned

When I connected the speakers to my system, my jaw dropped. Something was obviously wrong—they sounded terrible. I checked the phase connections and reconnected my own speakers to make sure the problem was not with my sound system. I knew from experience that Audio Concepts could design a pretty good speaker, so I decided to return them to Audio Concepts and let them correct the problem.

Audio Concepts did find the problem: the midrange driver is mated with a special enclosure designed to correctly load the driver. The misassembly of this part of the kit resulted in a leak, which allowed the woofer to modulate the midrange, causing the terrible sound.

The plastic midrange enclosure has a small hole in the apex for the wires. I had to enlarge this hole to accept the quick disconnect lugs. Following the directions supplied with the kit, I glued the midrange wires through the hole and sealed the hole around the wires with a large glob of silicone adhesive. Then, I stuffed the midrange enclosure with the supplied damping material and allowed the adhesive to set overnight. The next night I installed the midrange driver, pushing the extra wire into the enclosure.

It seems several factors combined to cause a new sound problem. The hole had to be enlarged, which then required a glob of silicone to seal it. Applying the silicone is a tricky, blind operation—you have to stick the nose of the gun down into the chamber, and hope you hit the hole. I also suspect it takes longer than one day for the silicone to cure. When I pushed the stiff wires into the enclosure, they stripped the silicone from the hole. The lesson to be learned here is the ef-

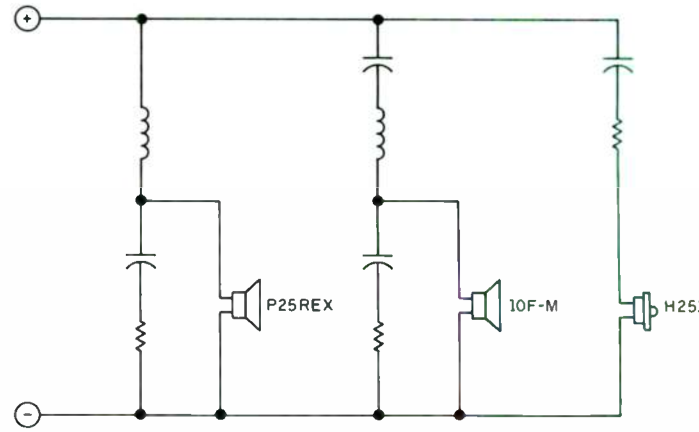

FIGURE 3: The first-order crossover network produces phase-coherent output. Audio Concepts holds the parts values as proprietary information.

Note

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Old Colony Sound Lab's Catalog No. 1

It may be removed for easier use and reference.

The Publisher

fect a small leak can have on the performance of an otherwise fine loudspeaker system.

I discussed the problem with Audio Concepts and they decided to make a subassembly of the subchamber with the speaker and wire installed. The subassembly is a nice improvement to the kit. It removes any possibility of error, and simplifies the assembly. All the kit builder need do is install the subassembly and connect it to the crossover with the quick-disconnect connectors. Note that the kit in *Fig. 2* shows what I received, not the kit including the midrange sub-assembly as it is now offered.

The second lesson learned had nothing to do with the sound of the speakers, but rather with their construction. Surprisingly, both woofers broke loose from their enclosures in shipment back to Audio Concepts. I had glued them in place with silicone adhesive, as I have done with many speakers in the past.

I did not use the long wood screws supplied because I believe with silicone they're unnecessary. I do not like using wood screws to mount speaker drivers. They never seem to go in right, and the screwdriver always manages to slip off the screw and pierce the cone. Where screws are required, I prefer to use

Continued on page 33

Dick Marsh Reviews and Revises Model G

I was curious about what level of loudspeaker performance could be obtained today in kit form for around $340 a pair. The Audio Concepts Model G is a the three-way system using the Seas H253 Ferrofluid hard dome tweeter and the P25REX woofer which has a polypropylene cone and a rubber surround. The Vifa M10 MD-5 cone midrange has its own subenclosure. The woofer and mid have cast frames.

The 6dB/octave crossover network with Zobel compensation appears to be good quality with film (polyester) capacitors, a ferrite core inductor and an air-core inductor. The wiring, of reasonable gauge and good quality, is terminated via 5-way binding posts.

I chose the oak finished enclosure with black grille cloth (off-white is optional). The enclosure came fully assembled, and well constructed with a good finish. It has a sloping front panel where the woofer, mid and tweeter mount vertically from bottom to top, respectively.

The Model G is also available with the parts only: no enclosure. It is available fully assembled too: drivers and crossovers mounted and ready to play. It has three half-inch holes in the lower rear corner to affect response and linearity (an aperodic design). All I need to provide is the labor to mount the drivers, crossover and internal damping material and to oil the wood.

Pretty good stuff for only $170 each. So I'm going to throw it together, give a listen and maybe a measurement. Right? Nah. Would I do that ?

We're going to see what cheap little things we can do to fine tune it. Tap on all sides is the customary first step. Yep, It will need some extra stiffness. I'll put a two-by-four across the back, extending from side-to-side, at about midway up. Then, I'll add 1 by 1-inch cleats along the joints. The resonances will be harder to excite now, have a higher Q and move above the woofer crossover frequency.

Next? Well, that crossover gets a polypropylene film substitution (Rel-Cap) for the tweeter, and PS 0.47μF bypasses on the others, and I'll touch up the soldering of the wires on the binding posts. The air core inductor can be remounted at a right angle to the ferrite inductor to minimize mutual inductance.

The push-on clips for the speakers are already soldered to the crossover wires, and they fit the mid and high speaker lugs very tightly. This is good. The woofer lugs are a bit loose fitting. However, I do have a soldering iron and, for the rest of this systems' natural life, it will probably never get opened up again. So they all get soldered to their speaker terminals. If you have a router (I do), use it to sink the speaker frames flush with the cabinet surface to reduce diffraction.

Lastly, what can be done with the damping material? Audio Concepts markets foam sheets for room walls and the inside of enclosures. After I line the inside with foam, I fill the rest with polyester fiberfill. This combination is an effective absorber over a broad range. On the outside, only the tweeter has foam around it. So I'll extend foam around the mid unit also, and a little between the mid and tweeter.

Speaking of foam and diffraction effects...Quite a lot of sonic improvement (sometimes attributed solely to enclosure diffraction effects) can be achieved by reducing the energy projected at nearby walls. Instead of absorbing these reflections on the wall nearest the speakers, it would be better to confine this troublesome energy at the source. Such as, using a foam/felt donut ring around the driver. A lot more of this foam on the enclosure front surface could help by effective-

Continued on page 33

Continued from page 32
machine screws and "T" nuts. I recommended this method to Audio Concepts, and they plan to modify the kit.

The third lesson I learned is that Audio Concepts is a company willing to stand behind their product, and improve its quality if possible.

Listening Evaluation

The repaired speakers sounded as I had expected them to when I first auditioned them. My general impression of the Model G is very good. Their most notable characteristic is superb detail and definition. However, as good as they are, their characteristics are such that they may not be the speaker for everyone's musical taste. Let me first describe the sound of the speakers, then my reaction to their performance with different types of music.

Audio Concepts stresses the importance of room placement to achieve the best possible sound. I found this to be good advice, particularly in the quality and quantity of bass. Audio Concepts sells optional 16-inch high speaker stands ($50 a pair) for the Model G.

As can be expected from two-cubic-foot enclosures, the lower bass rolls off. However, the mid and upper bass are tight and well defined. The quantity and quality of the bass seems just about right for most recordings without a lot of low bass. Occasionally, on some material, I noticed a slightly "muddy" quality in the bass. It's not noticeable enough to be a problem, and I only mention it because I have found muddy bass to be caused by polyester capacitors in other speakers. It is possible that with some experimentation, bypassing or replacing the capacitors, their sound could be improved.

The warmth of a musical instrument is determined by the region between the bass and midrange. I noticed a slight deficiency in this area. It is particularly evident in stringed instruments such as solo guitars or violins. The strings are predominant, overshadowing the bodies of the instruments. Presence is good—not too close, and not too far—indicating a stable and well balanced midrange.

The front of the enclosure is sloped at the correct angle to provide excellent imaging. Tilting the enclosure forward or backward did not improve the image, which is surprisingly sharp and well defined for such a large enclosure. The upper midrange and lower treble are responsible for the clear bright characteristic of the Model G. I suspect the speaker has a slightly exaggerated response in the 3-10kHz range. The high frequencies of the upper treble seem to also roll off, as evident by a lack of "sheen" in the cymbals.

I really don't know, but if I had to guess, I would say that Audio Concepts performed most of their listening tests for the Model G with rock music. The speakers really shine with this kind of music: their strong points are highlighted and their weak points minimized. The result is a lively presentation with as solid a foundation as I have heard in speakers costing twice as much. Individual instruments stand out clearly and are well balanced.

Unfortunately, as strongly as I recommend the Model G to rock enthusiasts, I cannot recommend them at all to classical music lovers. The emphasis of the upper midrange and lower treble, at times, creates an unusual effect. For example, in one piece, the sound of a triangle was as loud as the whole violin section. Massed violins are poorly presented, lacking sheen and air because of the deficiency in the warmth region.

The story is not as clear cut for other kinds of music such as popular, folk or jazz. Whether or not the Model G is right for you depends on your preference for their characteristics. If you like a speaker with the clear detailed sound that Audio Concepts has achieved, then by all means consider the Model G.

Dave Davenport
Raleigh, NC 27609

Continued from page 32
ly reducing the speaker dispersion, thereby increasing its directivity.

Way back when, controlled directivity was common: usually using horns and/or lenses for home use. But manufacturers thought sales could be increased if music lovers could hear all the frequencies from just about anywhere in the room. Thus, we were offered the wide-dispersion type system. I believe Acoustic Research (AR) was one of the first to make it a success using dome drivers. AR believed a flat power response, as well as a flat on-axis response, sounded better for their bookshelf systems. Their solution at that time was to provide nearly flat energy response over a full 2π-space radiation.

From this, other makers evolved omni-directional systems. Then multichannel systems came along, and their legitimacy got mixed up with the established omni "sound-is-everywhere" wide dispersion market. Now AR has come out with a system that uses substantial absorption around the dome drivers to narrow the dispersion and reduce room interaction problems. Twenty years of chaos.

I believe a response limited to 60-90 degrees would do wonders for audiophiles. Outside this angle, a rapid reduction in output is desirable to reduce the reflections from adjacent walls.

Perhaps one of the better methods of obtaining the desired angle is to use drivers with much wider dispersion, and to absorb the energy outside this area. This has benefits in crossover design, diffraction reduction and room reflections.

What does this have to do with the Model G? Nothing, I suppose. But it is a way to talk about a new found absorption material. In tests I found it to be an excellent absorber and panel vibration damper. It works well on walls as well as on loudspeaker enclosures. It is a ¾-inch three layer pad available in four sizes, ranging from 1 to 1½ square feet. It has a special rubber base, a felt middle, and a thin polyester foam top layer, and costs about $2 per pad. Ask for the deluxe felt business machine pad at your local business furniture or business supply stores.

Well, how does the Model G perform? I believe it is worth the price. The imaging is solid and doesn't shift with increased power levels. The frequency balance is fairly smooth with no really obnoxious peaks. It has a −2dB subjective midrange recession and a blurring around the bass-mid crossover region. The mid clarity is somewhat less than the better 2-way systems I've heard at slightly higher prices. It's not as fast and detailed, but this is offset by a very good bottom end. I later made a substantial improvement in clarity and resolution by rolling off the woofer faster, just beyond the crossover region. Try a 9µF cap across the woofer. This may not be the optimum value, so try others.

Over-all my reactions are favorable, especially in light of the cost and simple details described above. The cost could be further reduced by $140 a pair if you build the enclosures from the supplied plans.

This enclosure (and others) may be purchased separately, which might be useful for custom designs.

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Manufacturer's comments on page 59

Tools, Tips & Techniques

Sound Level Meter

My sound level meter *(Fig. 1)* was brought into being after I first used the Panasonic omnidirectional electret condenser microphones. You may use this meter for peak sound-pressure monitoring after calibrating it with an already-calibrated meter. I make frequency sweep tests using sine waves and pink noise in conjunction with octave band equalizers. You may use the meter to phase and position drivers by using sine waves and adjusting attenuator networks. You need not calibrate the meter when you use it only to measure relative levels.

You may leave R13 *(Fig. 2)* at the approximately one-half position, depending on meter sensitivity. I used an old military-type meter after disassembling it and removing the internal selenium bridge rectifier. This is a nonrectifier, DC-type meter with an estimated sensitivity of approximately 1mA. When looking for a suitable meter in my catalogs, I did not find much in the way of inexpensive meters, but by changing R14 to 2.7k and inserting another variable pot (R15) in series with R14, you may use virtually any meter (1mA to 1μA). R15 equals zero for 1mA and 100k for 1μA. One source of a 1mA VU meter is All Electronics Corp. (PO Box 20406, Los Angeles, CA 90006, Part MET-1, $4.50).

You may also use a step-type LED indicator in place of the mechanical meter. Possible replacements are the NSM3916 (red) and the NSM39168 (green, yellow, red), which cost $5.95 and $6.95, respectively. They are available from Jameco Electronics (1355 Shoreway Rd., Belmont, CA 94002). If you use an LED dot graph, connect it to the wiper on R13. Set the LED's sensitivity to approximately 3 to 4V for full scale. This setup might be your best bet. Originally, I avoided using an LED because of increased battery drain, but by using the "dot" mode, the drain should be less than 10mA. The LED modules are approximately 2 inches long and use the LM3916 programmable VU circuit.

In any particular gain setting, the output should be good for more than a 20dB range. In my version, I used standard 5 percent resistors and 10dB steps in the switch, providing a 40dB total range of

FIGURE 1: The author used a military-type meter in constructing his unit.

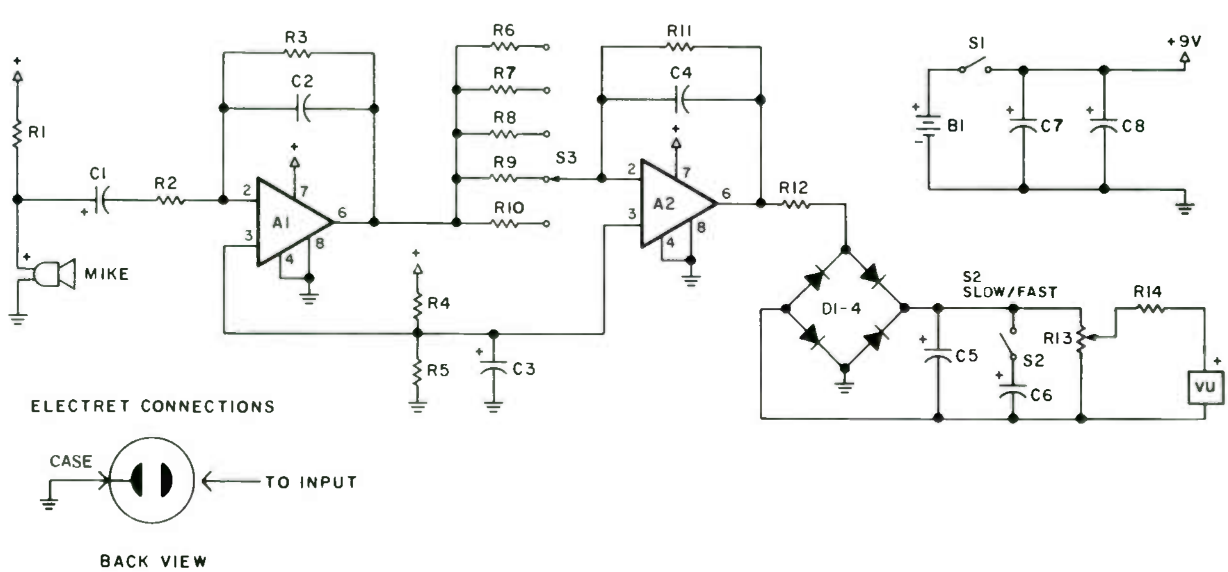

FIGURE 2: Schematic diagram of Mr. Szekeres's sound level meter.

control settings. You could devise 1 percent precision resistors in the gain stages.

I mounted all the components inside a plastic box with a metal faceplate. Using a prepunched and solderable eyelet perfboard, cut the board to size and mount the components. Keep the grounds together at one point, and keep R3-C2 and R11-C4 close to their respective amplifiers. Ground the front faceplate, then ground pin 8 on the low-power op amps for maximum bandwidth. Mount C7 and C8 near each op amp. If necessary, drill a small hole for R13 to facilitate calibration.

I mounted the electret capsule on the forward-facing side of the box. Use silicone and/or a small donut-shaped piece of foam. Whenever you use silicone rubber on electronics and speaker crossover wiring parts, make sure you use the electronic type if possible. It is alcohol based and does not corrode parts, as the standard stuff does. To reduce pickup refraction and reflection on the microphone side of the box, surround the capsule hole with a square of soft foam (⅛ to ¼ inch thick).

That's all there is to this sound level meter. I hope your project is as successful as mine was.

Greg Szekeres
Pittsburgh, PA 15236

PARTS LIST

Resistors*	
R1, R6	2.2k
R2	10k
R3, R11	330k
R4, R5, R10	220k
R7	6.8k
R8	22k
R9	68k
R12	1k
R13	100k variable trimpot
R14	4.7k
*All resistors 5%	
Capacitors	
C1, C3, C5,	
C7, C8	6.8μF, 15V tantalum
C2, C4	5pF
C6	47μF, 10 to 15V
Misc.	
D1-D4	1N914
VU meter	any calibrated DC meter (see text)
B1	9V
S1, S2	SPST
S3	Radio Shack 275-1386 or 275-1385
A1, A2	TLC271 (Radio Shack 276-1748)
Mike	Panasonic (Digi-Key Part #P9932**)
box	suitable size

**Digi-Key Corp., Highway 32 South, PO Box 677, Thief River Falls, MN 56701; $1.95 each plus shipping and handling.

Mini-Tips

Here are a few small but useful tips I have picked up. First, cover vents or transmission-line exit holes with a nylon screen to keep out insects and even mice. Second, check your capacitor values when assembling your own crossover. I often find that the true capacitor value is at the limit of its plus/minus tolerance. Third, banana plugs make the most useful amplifier-to-speaker connector. Once you use them, their strengths will become apparent.

Ian Dibbs
Ottawa, Ontario K1R 6H8

Hatching A Better Woofer

While modifying a pair of 8-inch woofers according to the guidelines laid out in Walter D'Ascenzo's article in *SB* 2/82 (p. 7), I reached the point where I needed a new pair of airtight caps. His statement that a "little imagination while shopping in a hardware or gift store can go a long way" strikes at the heart of what building one's own audio equipment, including speakers, is all about, so I considered what I could press into service. I found the answer in a predictably unlikely place.

L'eggs pantyhose, which are sold in just about any store, are packaged in a nifty little plastic egg, the contours of which are ideal for this application. After obtaining a pair of these eggs from a friend's wife, I cut off the narrow end of each with a Dremel Mototool. This left me with two plastic caps about 1½ inches in diameter. I used the narrow end because it looked as though it would clear the woofer's brass voice-coil form, allowing the cap to fit firmly on the cone. I also had to allow clearance for the foam piece glued to the woofer magnet, as per Mr. D'Ascenzo's article. After another 15 minutes with the Dremel shaping and fitting the caps to their respective woofers, they were ready to silicone-seal in place.

Be sure to clean off any dust every time you try the cap on the woofer for fit, and keep the driver's core covered whenever you are not working directly on it. You do not want plastic dust—or anything else, for that matter—floating around in the magnet gap. And need I mention safety glasses when cutting *anything?*

Since my initial, slightly self-conscious inquiries of female friends, I have built up quite a backlog of these eggs (in assorted colors, yet) and now have a more-than-adequate supply for future modifications.

By the way, Mr. D'Ascenzo's woofer modifications, taken as a whole, are a worthwhile undertaking. I performed them on a pair of drivers originally used in Radio Shack's Realistic Nova Six—a two-way sealed box of unexceptional ability—now being used in a 6-foot transmission-line design. The improvement in midrange clarity is remarkable and has made a good driver out of a barely passable one.

If you have some mediocre drivers that you are hesitant to use because everything seems to be polypropylene now, give these modifications a try. You will be pleased with the results. Executing them will also go a long way toward demystifying the innards of a driver. There is nothing like slapping a little polyurethane on a woofer cone to reassure you, on a gut level, that speakers are just machines and can gain from a little fine-tuning.

Robert Stacy
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CROSSOVERS & SPIKED BASES

Crossovers can be easily assembled by using tempered hardboard, silicone glue and eyelets (or grommets) as connection points. Parts can be more securely glued to the textured side of the hardboard with either a silicone glue or mortite glue. How you lay out your components is subject to your whim, but I use a "loop" layout for my second order design as shown in *Fig. 1*.

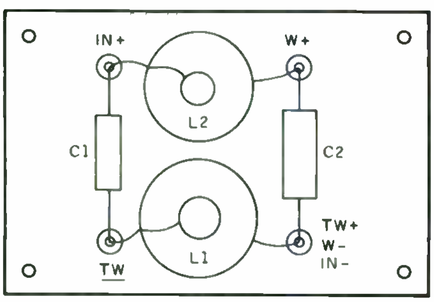

FIGURE 1: I use a "loop" layout for my second order design.

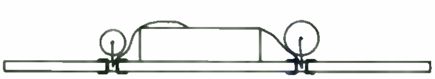

FIGURE 2: A cross-sectional view of my loop layout with grommets.

Eyelets, originally designed for use with leather or canvas, make inexpensive solder points. Most sewing shops have 3/16-inch diameter eyelets: the perfect size for most designs. Another source is any leather outlet. (*Be sure they are free of lacquer or can be cleaned for soldering—Ed.*)

To use the eyelets, simply drill a hole in your board, insert the eyelet and set with an ordinary eyelet setting tool or anvil setter. This method, as shown in *Figs. 2* and *3*, provides a stable connection without expensive terminal strips or binding posts. It is used by my company, as well as by Thiele loudspeakers.

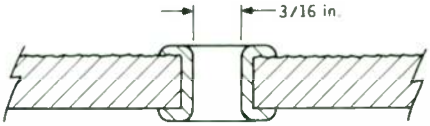

FIGURE 3: Close-up view of a grommet inserted into hardboard.

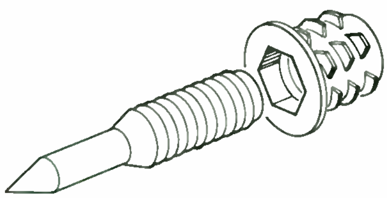

FIGURE 4: The spike with its threaded insert.

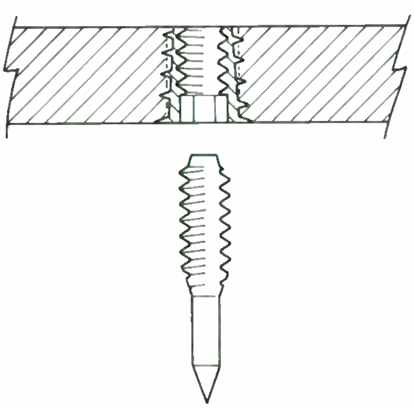

FIGURE 5: The insert is mounted into speaker bases, and the spike is easily screwed in or out for use on different surfaces.

The current rage for loudspeaker positioning is to use "spiked" bases, to provide optimal coupling to the floor. My company recently began stocking and supplying spikes with threaded inserts as shown in *Fig. 4*, which are easily mounted in holes drilled in speaker bases and screwed in with an Allen wrench as shown in *Fig. 5*. Most high-end dealers can do spike modifications to your cabinets, but charge up to $20 a pair. We sell home builders eight inserts and spikes for $6. Our spikes are removable if the cabinets are to be used on wood floors or easily marred surfaces.

Robert Conner
Shadow Audio
P.O. Box 55081 [Dept. SB]
Omaha, NE 68155

CABINET COMPUTATIONS

Here is a list of basic information and formulas for general cabinet computations, where V = volume, l = length, w = width, h = height, $\pi = 3.14$, and r = radius.

Volume Units

Cubic or rectangular shapes: $V = l \times w \times h$
Cylindrical shapes: $V = \pi r_2 h$
Spherical shapes: $V = 4/3 \pi r_3$ or $V = \pi d_3/6$
Cones: $V = 1/3 \pi r_2 h$

For cones, the radius measurement is taken at the base.

Liters

1 liter = 1 cubic decimeter = 10cc = 1/10 cubic meter

To convert liters to cubic feet, multiply the number of liters by 0.0353. To convert cubic feet to liters, multiply one cubic foot by 0.0283.

British projects usually give the measurements in millimeters. To get a direct reading in liter units, it is better to convert the millimeters to centimeters before the computation. Simply move the decimal point two places to the left (i.e., multiply each length measurement by 0.01).

Length Units

To convert centimeters to inches, multiply the number of centimeters by 0.3937. To convert millimeters to inches, multiply the number of millimeters by 0.03937. To convert inches to centimeters, multiply the number of inches by 2.54. To convert inches to millimeters, multiply the number of inches by 25.4. A ruler that divides the inch into tenths will simplify many measurements.

Carlos E. Bauza
San Juan, PR 00936

NEW FASTENERS

A new type of fastener is on the market which I have successfully used to mount drivers to enclosures. It is called a Well-Nut®, and works as shown in *Figs. 1-3*.

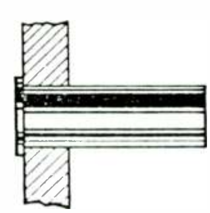

FIGURE 1: The Well-nut is a blind fastener, and can be installed from outside an enclosure.

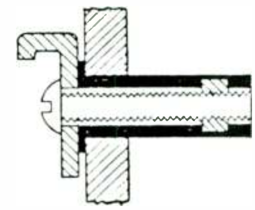

FIGURE 2: The part to be attached is placed against the flange and secured by the machine screw engaging the internal brass nut.

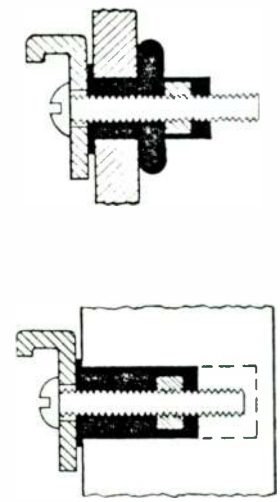

FIGURE 3: As you tighten the screw the neoprene body is compressed, forcing it to tighten behind thin walls (a) or expanding tightly against the walls of a hole (b).

The manufacturer is POP Fasteners Division, Emhart Fastener Group, 510 River Road, Shelton, CT 06484.

The Well-Nut is a blind fastener, that is essentially a neoprene bushing with a captive brass nut. It comes in 31 sizes for use in holes with diameters from ¼ to ¾-inch. I have used the #10-XL, which uses a 10-32 machine screw and works well in ¾-inch thick plywood. Some added bonuses of using the Well-Nut are that it can also fasten to thin walls or blind holes, seals air leakages and dampens vibration.

Larry Cartwright
Pittsburgh, PA 15213

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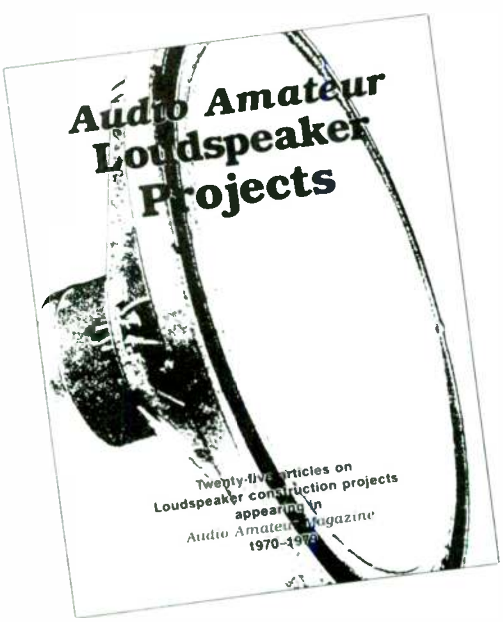
Twenty-five articles on Loudspeaker construction projects appearing in *Audio Amateur Magazine* **1970–1979**



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Craftsman's Corner

FULL-RANGE SATELLITE SPEAKERS

This description is of a speaker system I recently completed. The design is a full-range version of Mr. D'Appolito's "High Power Satellite Speaker" (*SB* 4/84, p. 7) using 6.5-inch instead of 5.25-inch bass drivers. My woofers are Audax Bextrene units, which are otherwise very similar to the Peerless TP-165F units that Mr. White used in one example in his "Realizing the Potential of Box Response" article (*SB*1/85, p. 28). Mr. White showed that four such drivers, in a fourth order alignment, can produce 115dB down to about 60Hz, or about 110dB down to about 40Hz, in a sixth order alignment.

My drivers' X_{max} is 21 percent greater than the subject driver's. Consequently, I estimate that my drivers can produce 115dB down to 54.5Hz in fourth order, or 110dB down to 36.3Hz in sixth order. *Photo 1* shows a completed unit. Each unit stands 35 inches high, is 16 inches wide, and 12 inches deep.

To minimize diffraction effects, produced by sharp discontinuities, I tapered each speaker panel back by 45 degrees, to the parallel rear side panels. The

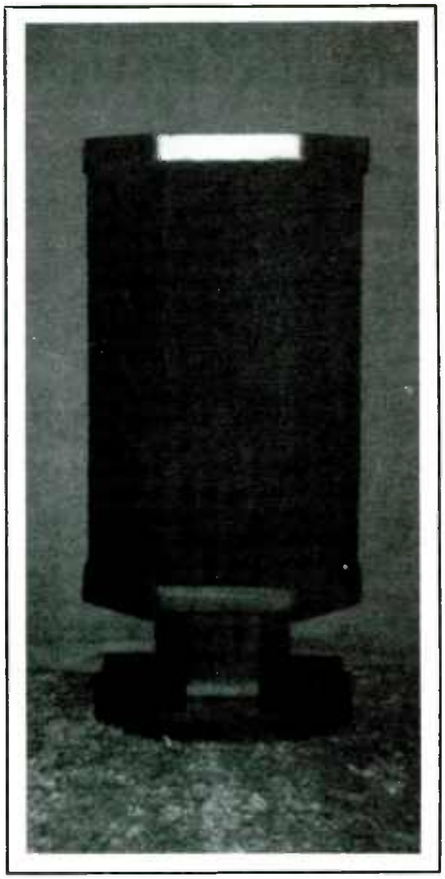

PHOTO 1: The finished product performs as well as it looks.

PHOTO 2: Cabinet construction front and back. Note bulkhead placement.

speaker enclosure is not very tall, and with the tweeter situated between the woofers, its axis is considered to be too far below listening height if it is floor standing. To correct this, and at the same time to give the speaker the illusion of having more mass, I raised the speaker with a column and base. I also extended the lower end of the cabinet with a skirt.

Photo 2 shows the construction details with the bulkhead situated near the middle of the enclosure. This does an excellent job of stiffening the ⅝-inch particle-board panels. I used dual tuning tubes, which project into the space between the column and the skirt. My crossover filter boards are mounted on the inside surfaces of the column for easy access. The filters are virtually the same as those used by Mr. D'Appolito. I have also applied several coats of auto body undercoating to the inside surfaces of the speaker enclosures.

The outside surfaces were covered with a hemp carpet padding, after making the speaker holes, suppressing diffraction effects further. *Photos 3* and *4* show a cabinet before and after applying the hemp padding.

I wanted to finish the cabinets in wood-grain plastic, but was intimidated by the somewhat complex shape of the cabinet. I used, instead, a wood-grain vinyl veneer I found at a local department store. This veneer is easily cut to size, and simply ironed on.

David J. Meraner
Schenectady, NY 12302

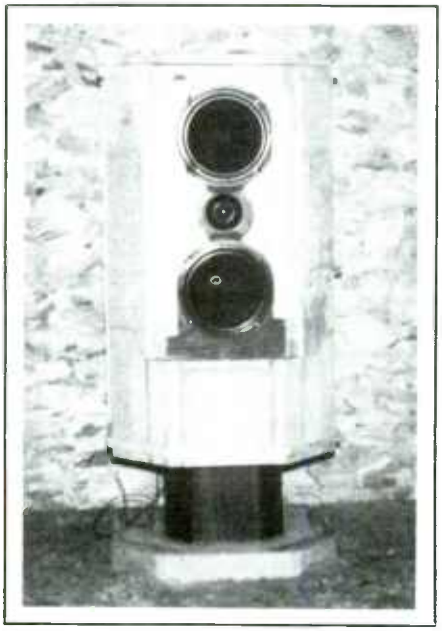

PHOTO 3: Speakers mounted before hemp padding has been applied.

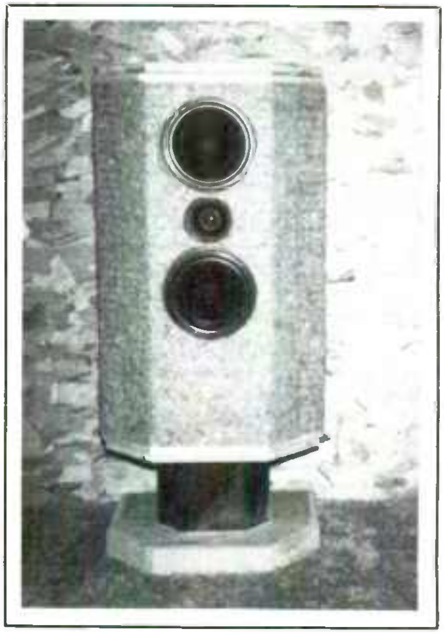

PHOTO 4: Speaker unit with hemp padding to help suppress diffraction.

SB *Mailbox*

CORRECTION: BOXRESPONSE

I am reporting an error in my BOXRESPONSE program as listed in *SB* 1/84, p. 13. I thank Joseph D'Appolito for finding the error and reporting it to me.

The problem with the program as published is that the voltages E and E1 in lines 440 and 450 are not comparable when an equalizer is used because E is then referenced to the equalizer output, while E1 is referenced to the equalizer input. The simplest solution is to refer E1 to the output and account for the equalizer after line 430. This can be done with the following line changes and additions. Replace lines 400, 401, 402, 410 and 432 with:

```
(*)400 1F 0=2 OR 0=4 THEN Y9=1: GO TO 420
   401 REM Y9 IS THE EQUALIZER
   402 REM OUTPUT TO INPUT
(*)410 Y9=X1^2/SQR((X1^2-1)^2+(D*X1)^2)
   423 REM VOICE COIL INPUT VOLTAGE
```

```
Insert the new lines 437/438:
```

```
   437 REM E2 IS AN EQUALIZER
   438 REM INPUT VOLTAGE
```

```
Replace lines 440, 450 with:
```

```
(*)440 if E1<E THEN E2=E1/Y9
(*)450 IF E<=E1 THEN E2=E/Y9
```

```
Insert the new lines 457, 458, 459:
```

```
   457 REM LINE 459 FACTORS IN
   458 REM THE EQUALIZER ACTION
(*)459 Y=Y*Y9.
```

Only the lines marked (*) are necessary. The others are explanatory REMs. *[The (*)'s do not go in the program, of course. —Ed.]*

When you run this corrected version for an equalized system, you will see very large maximum input powers at low frequencies. This is alright because the equalizer will cut them down to levels the driver can handle. This column now gives an indication of how powerful and amplifier you can use to driver the system.

Robert M. Bullock III
Contributing Editor

ELEVATED BOXRESPONSE

I have been using the Bullock-White BOXRESPONSE computer program since it appeared in *SB* 1/84, p. 13. It works fine and I have been very pleased with the results. My problem is that two months ago I moved to Albuquerque from Indianapolis. My house is located 6040 feet above sea level. The less dense air has a noticeable effect on my speaker systems. I wonder whether it is possible to change a constant or two in the program so it gives correct results at higher elevations. I am sure other readers at high altitudes would appreciate 'vice on this problem.

David Lu.
Albuquerque, NM 87122

Mr. Bullock replies:

Elevation can be taken into account in BOSRESPONSE by changing the air density from my assumed constant value of 1.18 to the variable value:

$$\rho_0 = 1.29 \ \frac{273}{T} \ \frac{P_0}{0.76} \quad \mathrm{kg/m^3}$$

where T is the temperature in degrees Kelvin and p_0 is the barometric pressure in meters of mercury. If you wish to supply barometric pressure and temperature for this purpose, you may as well change the speed of sound from the 345 I used to:

$$c = 331.4 \sqrt{\frac{T}{273}} \quad \mathrm{m/sec}$$

Both formulas are from Baranek's Acoustics, *p. 10.*

The following program lines added to BOXRESPONSE will calculate p_0 and c from Farenheit temperature and barometric pressure in inches. The lines must be added before present line 80. I will start with line 63.

```
63 "ENTER FARENHEIT TEMPERATURE"; T1
64 "ENTER BAROMETRIC PRESSURE IN INCHES"; B1
65  REM CONVERSION TO MKS AND KELVIN
66  T2 = 273+5*(T1-32)/9: B2 = .0254*B1
67  REM DENSITY OF AIR IS
68  PØ = 463.4*B2/T2

74  REM VELOCITY OF SOUND IS
75  C = 331.4*SQR(T2/273)
```

To incorporate these new values of p_0 and c, lines 80, 90, 690 and 700 must now be changed to read as follows:

```
80  K  = .4218*SQR(PØ)*SQR(V*FΛ3/Q1/R)/C
90  K1 = 132.5*SQR(V/F/Q1/R)/R2Λ2/SQR(PØ)/C
690 M9 = 1974*Y5/PØ/C/D9Λ2
700 L  = .01785*C*C*A*D9Λ2/V/(H*F)Λ2 - .7135*D9
```

You may want to change an inconsistency in line 690. Y4 as calculated in line 510 is not the true volume velocity, but the volume velocity multiplied by p_0. This is the only reason line 690 contains p_0. More correctly, lines 510 and 690 should be changed to:

```
510 Y4 = K*H2*E2*Y/F/XΛ3/PØ
690 M9 = 1974*Y5/C/D9Λ2
```

Either version gives the correct output, but the latter is also conceptually correct.

A NEW COOL BAND FLUKE

I thank John Cockroft for his interesting article on Isobarik loading. I believe the 4-inch Radio Shack woofer he mentions near the end of his piece is the 40-1022, the same woofer used in Radio Shack's notable Minumus-7 speaker (40-2030).

On a different, more intriguing note, the author bio note at the end of his article states he is a "Senior cryogenics technician at the Stanford Linear Accelerator Center." Could this mean a article is in the works on constructing a home brew driver with a super-conducting voice coil? What a sure-fire way to reduce DC resistance in the woofer circuit!

Now, if I can just figure out where to put the liquid helium tanks....

Robert Stacy
Danbury, CT 06910

Mr. Cockroft replies:

Mr. Stacy, you let my secret out prematurely. Someone will probably get the jump on me (and after all that darn work, I will probably be stopped cold). Since you asked, I must confess I plan to put the liquid helium directly behind the tweeter. This way, if any helium leaks (and it always does) the tweeter will function as a super tweeter.

The woofer surround has been giving me a bit of trouble, but I think I'm finally on the right track. I am now using Fibrefax® (a non-woven ceramic fabric) in conjunction with a ring burner around the outside edge of the woofer to keep the surround flexible. Why, you ask, is this necessary? Well, there goes another closely kept secret. For the woofer cone I use felt saturated with liquid helium to supply the ulitmate stiffness. When perfected it should make an excellent system for cool jazz.*

For the sake of economy the article will be published in a winter edition, to minimize the heat load (not to mention stiffness loss).

Continued on page 44

McGEE RADIO HOUSE of SPEAKERS

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Continued from page 42

Seriously, with the problems we have with the support equipment for the SC magnets, I think I'd prefer to design an ionic flame subwoofer system.
**In the ultimate system, the bass must be felt.*

MORE SATELLITES

Recently I built a pair of satellites using the D'Appolito article in *SB* 4/84, p. 7, and would like to thank him for the design. I also have a few questions about them. First is the "acoustic foam" D21 urethane the same as mentioned in Martin Colloms' book? I know Sonex and Audio Concepts stock a corrugated foam, but where can I buy the flat type? Secondly, in your impedance compensating network, the resistor is a 7.5Ω even though the woofer impedance is now approximately 4Ω.

Finally, will amplifiers with "soft" versus "stiff" power supplies put a different power ratio into the tweeter versus the bass-mids? Not having the time to use mastic, I used lead sheeting (for the first time) with good results and would appreciate your subjective opinion on damping materials.

Armand Fasano
Mendham, NJ 07945

Mr. D'Appolito replies:

Audio Concepts used to supply their acoustic foam in ½-inch flat sheets, but it no longer seems available. Foam rug padding, as I stated in my original article, is available from many carpet stores and may be used, but it is not as effective as acoustic foams like D21.

The 7.5Ω resistor was chosen not only to compensate for driver impedance, but also to set the overall Q of the low frequency crossover in my satellites for the flattest frequency response. When using Focal drivers, for example, change this resistor to 5Ω.

I suspect very little difference in speaker response between "soft" and "stiff" power supplies. In normal operation the output impedance of an amplifier, with reasonable amounts of negative feedback, is only slightly influenced by power supply regulation. The difference between soft and stiff supplies will generally only show up near or at clipping, where effects on the loudspeaker are difficult to predict.

The Focal modifications to my satellites appear in the Mailbox section of SB 4/85, p. 46, and my work on the 17W75's is presently on hold.

SOUND PRESSURE SENSITIVE

Mr. D'Appolito's reply to Ken Kern's letter on sound pressure levels (*SB* 2/85, p. 48.), while mathematically supportable, seems to have missed some important points. I believe they rest, as Mr. D'Appolito states but only indirectly elaborates on, in the way speaker sensitivities are rated and in the operating characteristics of solid-state power amplifiers.

If we take the example of paralleling two 8Ω 90dB SPL drivers, while holding input voltage constant at 2.83V, the current supplied by the amplifier must then double since load resistance is halved. Thus, while the sound pressure level doubles, amplifier output must also double. Because of its low internal resistance a typical amplifier with bipolar output devices willingly does this, within limits.

Since SPL is a function of current the increase is expressed as 20logI1/I2 or 20log2 = 6dB. The increase in amplifier output power required to realize this, on the other hand, is expressed as 10logP1/P2 or, in terms of changed current and load resistance, $10\log[(2)^2(0.5)] = 3dB$. Net system gain is the difference, or 3dB. It is also worth noting that this gain is achieved through greater directivity in the radiation pattern of the driver pair (ie., narrower beam width).

When we put the two drivers in series, again holding the input at 2.83V, load resistance compared to a single driver obviously doubles and amplifier output current is halved. The SPL then drops by $20\log 0.5$ or $-6dB$, while amplifier output power changes by $10\log I^2R$ or $10\log[(0.5)^2(2)] = -3dB$ resulting in a net difference of $-3dB$.

If we normalize the amplifier's role in this, we get an SPL of 93dB for parallel driver connection versus 87dB for series connection. The difference in levels does not conflict with the radiation gain realized by using stacked drivers, since normalized power input in the series case is one-fourth ($-6dB$) what it is in the parallel case. In other words, the apparent differences in sensitivity are, more than anything else, a reflection of how the amplifier responds to changes in load impedance for parallel versus series driver connections.

For simplicity I have neglected whatever effects on power division may be present due to mutual acoustic coupling of the drivers. The preceding analysis, by the way, is consistent with radiation theory in allied fields such as antennas, where doubling effective area produces a nominal gain of 3dB, which is also attended by increased directivity.

It seems misleading to suggest that "paralleling like drivers is a good way to improve loudspeaker system efficiency." It might have been better to say that this is a good way to increase loudspeaker output, because it increases the load on the amplifier. This is certainly a legitimate and useful way of making multidriver balance problems more tractable.

Taken at face value, however, the statement ignores total system efficiency. The fact that the amplifier must work much harder, not to mention increased current losses in speaker leads and networks, and other undesirable effects that can arise when resistive elements become an appreciable portion of the total load as loudspeaker system impedance declines.

Eager speaker manufacturers sometimes play the speaker impedance game to make rated sensitivities of their product look better on paper. Not only does this add to the confusion but, if trends to lower system impedances continue, amplifiers will soon need to be capable of deca-amperes of output, and we will all be using double-ought bus bar for speaker leads.

I am sure, however, that Mr. D'Appolito had nothing like this in mind. Nor does it detract from an excellent project and interesting article (*SB* 4/84, p. 7). But as he intimates, the whole approach to rating speaker sensitivities needs an overhaul.

Stuart E. Bonney
Richardson, TX 75080

Mr. D'Appolito replies:

Mr. Bonney believes my reply to Mr. Kern's letter is misleading and misses some important points. I must disagree. My letter was carefully written in direct response to Mr. Kern's questions. The letter is correct in every detail and, I believe, has the emphasis appropriate to the questions which prompted it. I agree proper sensitivity matching of single and multiple drivers is influenced by the driving amplifier, but I believe that point was clearly made in my original letter.

Beyond this issue Mr. Bonney raises additional issues, which if correctly analyzed could have further added to readers' understanding of the advantages of multiple drivers. Unfortunately his errors lead to incorrect or confusing conclusions which must be corrected. Specifically Mr. Bonney tries to make two points:

- *parallel drivers cause the amplifier to work harder, and*
- *the increased sensitivity of parallel drivers is obtained at the cost of increased directivity.*

Looking at the first point, Mr. Bonney misinterprets his own calculations. He correctly

SB Mailbox

asserts that an amplifier must supply twice the power to produce twice the SPL. It's not surprising that an amplifier works harder to produce higher SPLs, but that is not a fair comparison. We must compare the current and power required by single and paired drivers to produce the same SPL.

For the same SPL: the parallel driver pair will draw the same current and one-half the power of a single driver, while the series pair will draw one-half the current and one-half the power of a single driver. Thus, for any given SPL the amplifier always works harder with a single driver than with a pair of drivers.

Mr. Bonney's analysis of the series connected pair is also incorrect. As shown in my original letter to Mr. Kern, the series connected pair has the same voltage sensitivity as a single driver. It follows that there is also a +3dB gain in power sensitivity for the series pair and not a 3dB loss.

Turning to the directivity issue, Mr. Bonney's antenna analogy fails because it deals with the wrong relative frequency range and does not include electromechanical transduction effects that are critical to loudspeaker performance. Antennas are generally comparable in size or larger than the wavelengths they radiate. Under this condition doubling the antenna area doubles its on-axis gain, but gives little increase in an efficiency which is already very high.

Direct radiator loudspeakers are much smaller than the wavelengths they radiate over much of their operating range. For this condition, doubling of the area results in a factor of four increase in radiation resistance with no significant increase in directivity. However, this does not increase efficiency or radiate power since doubling the area at least doubles the cone mass causing a factor of four drop in efficiency for a net gain of zero. When we double the area by using two drivers, we also double the motor efficiency (two voice coils, two magnets).

For the parallel pair cone velocity is unchanged, input power is doubled and output power quadruples, for a true factor of two (+3dB) increase in efficiency. Of course we also double the maximum SPL capability of the system. The doubling of cone mass and motor efficiency and the quadrupling of radiation resistance are not represented by Mr. Bonney's antenna analogy. (A proper antenna analogy can be made.)

Incidentally, since placing two small identical drivers close together quadruples their radiation resistance rather than doubling it, is clear that the mutual acoustic coupling between the diaphragms at low frequencies is not negligible as Mr. Bonney asserts. It is in fact comparable to the self-radiation resistance of the drivers. See E.M. Long's "Design Parameters of a Dual Woofer Loudspeaker System," JAES (Volume 17, No. 5) 1969, p. 515, for more details. Repeating it is the doubling of both cone area and motor efficiency, plus the factor of four increase in radiation resistance, that produces a true doubling of acoustic efficiency.

As the size of the drivers and their separation becomes comparable to the radiated wavelength (about 800Hz for my satellites), the mutual radiation resistance begins to fall and the stacked driver pair does become more directional in the vertical plane. In fact the increasing directivity just offsets the falling mutual coupling to maintain constant on-axis response. Here Mr. Bonney's antenna analogy is more appropriate. I would point out, however, the increased directivity in this frequency range is not a penalty with my satellites, but a benefit since it reduces floor and ceiling reflections.

Finally, to bring home the benefits of multiple driver configuration most strongly, let us look at a concrete example using the Peerless TP165R 6½-inch midbass driver. From the published specifications for this driver, and Linkwitz's formulae in SB 4/84, we can construct Tables 1 and 2. Table 1 compares the performance of one driver with two in parallel or series and with four drivers in series/parallel at 100Hz. At its maximum linear excursion a single TP165 can put out 102dB SPL into a hemisphere at one meter and 100Hz. Columns 2 and 3 of Table 1 show the current and power required to produce the same 102dB with two and four drivers. Notice that power is halved in both of the two driver cases. With four drivers both current and power are halved. Under no circumstances does the amplifier work harder with multiple drivers.

TABLE 1

POWER AND CURRENT REQUIREMENTS FOR TP165s at 102dB and 100Hz

No. Connection	RMS Current (amps)	Power (watts)
single	2.12	36
two/parallel	2.12	18
two/series	1.06	18
four/series/parallel	1.06	9

Table 2 compares the maximum SPL capabilities of single and multiple TP165s. Notice that two drivers can produce a 6dB higher SPL, while four drivers can generate 12dB more. Column 2 gives the power required by each combination to generate its maximum SPL. Column 3 shows the power that would be required by a single driver to produce that same maximum SPL if it had the necessary additional throw. Again we see that an amplifier always works harder with one driver.

The same 114dB SPL could be produced by a single 12-inch speaker at much lower driver cost, but it would in general be less efficient. For example, the Peerless TA305F would require 5.8A and 270W to produce 114dB SPL. When the difference in cost between power amplifiers is considered, the multiple driver configuration may well be more cost effective.

The many advantages of using multiple drivers should be clear now. They provide an easy way to increase loudspeaker sensitivity, efficiency and maximum output capability with little penalty. In addition, the response of smaller drivers extends well beyond that of larger woofers making possible simple two-way high power systems.

TABLE II

MAXIMUM SPL AT 100Hz FOR TP165s

No. Connection	Max. SPL (dB)	Power (watts)	Power Single TP165
single	102	36	36
two/parallel	108	72	144
two/series	108	72	144
four/series/parallel	114	144	576

PASSIVE CROSSOVER QUESTION

In *SB* 1/85, p. 15, Contributing Editor Robert Bullock says of R_W and R_T: "For loudspeaker crossovers, you can take their values to be the DC resistance of the woofer and tweeter..."

Does this mean when an impedance equalizer is used, making the impedance of the driver equal to R_E, or in all cases? If so, why? And what if an L-pad or other control is used on the tweeter?

David B. Weems
Newtonia, MO 64853

Contributing Editor Bullock replies:

Your quote from my article should be taken in context with the last sentence of the previous paragraph, i.e., the loudspeaker loads must be equalized according to the directions in the impedance equalization section of the article. I apologize for not making this point clear.

The equalization procedure I described is intended to make a loudspeaker appear as a resistor of value R_E, at least for an octave or so on either side of a crossover frequency. In this sense, R_E is the appropriate design load.

For L-pads, it is my understanding that their purpose is to attenuate output without changing the load. If this is so, the design load is L-pad independent. Otherwise, the resistance of the L-pad-loudspeaker combination should be taken as the design load.

If you do not use equalization, or some scheme other than the one I described, I cannot recommend an appropriate design load. It is perhaps reasonable here to make an impedance magnitude plot of the loudspeaker/equalizer/attenuator system and choose the design load as some graphical average. Such a graph is also useful for checking results *when using the equalization method of the article.*

RIBBON MODE MATTERS MUDDLED?

In our response to Ralph Gonzalez's letter "Ribbon Speaker Mode Matters," *SB* 4/85, p. 54, *Figs. 4* and *7* were transposed.

Based on the information presented in the original article, and our reply to Mr. Gonzalez's letter, we believe that diffraction, whether caused by the edge or the face of the mounting plate, is not a cause of the response irregularities in free air response.

Diffraction produced by radiation from the edge or the face of the driver must occur within 0.2 milliseconds after excitement by the diaphragm. After this time period the pressure zone grows beyond the physical boundaries of the driver, and can no longer excite the edge or the surface of the driver or the enclosure.

Figure 3 of our reply to Mr. Gonzalez shows the response of all errors that occur during this critical time period. The 4kHz response dip that Mr. Gonzalez claims is caused by diffraction has not yet been defined. In fact the response is rising at 4kHz, not falling.

As correlative evidence for our non-diffraction position, please refer to *Fig. 15* of our article in *SB* 3/85, p. 22 (also see *Photos 1* and *2*). This test, with the face of the driver removed, should represent the response with minimum diffraction if the face of the driver is the source of diffraction. The response under these conditions does not appear smoother, flatter or exhibit more optimum transient decay which would seem to be the expected result of eliminating the supposed source of the error (i.e., diffraction).

As a final refutation of Mr. Gonzalez's position, we submit, without an attempt to eliminate diffraction effects, that we were able to minimize or eliminate the errors found in the free air Strathearn response—including the 4kHz dip.

Neither Mr. Spangler nor myself wish to give the impression that diffraction does not exist, or is not an important consideration when designing speakers. Under certain conditions, with certain drivers, diffraction is measurably and sonically important, but not in the case of the Strathearn driver.

Determining when diffraction is important or relevant is not as easy or as universally applicable as many publications, including *SB*, have made for the case of diffraction induced problems.

Mark McKenzie
Reynoldsburg, OH 43068

TO STRENGTHEN MY STRATHEARN

After reading the Spangler and McKenzie article on Strathearn modification, *SB* 3/85, p. 22, I am still not clear on some points and would appreciate answers to two questions.

First, what taps did they use on the toroidal matching transformers? Mine have three taps for approximately 4, 8, and 16Ω.

Secondly, did they use the $\frac{1}{8}$-inch open cell foam under the damping panels on the front and back, or just under the front?

Don E. Prock
Rubidoux, CA 92509

Mr. Spangler replies:

To answer to your first question, use the outside lugs. We made this choice because the transformer performed better. The Strathearn transformer is excellent. We used the foam damping strips under both the front and back "wall-boards."

SB *Mailbox*

SPEAKER SAVINGS FOR D-I-Y'S

For those of us "Do-It-Yourselfers" who must pay attention to cost, I offer some relief and design simplification. The Ellipse W0838RLE eight-inch woofer can be used as a suitable substitute for the Dynaudio 21W540 eight-inch woofer.

The Ellipse sells for less than $35 (as of 1/25/85) from Meniscus Audio Systems (3275 Gladiola S.W., Wyoming, MI 49509), and has power handling similar to that of the Dynaudio. At a 500Hz crossover, as described by L.A. White in *SB* 4/85, p. 18, a cast frame driver seems unnecessary.

The eight-inch Ellipse has a 38-ounce magnet (not structure), butyl surround, rear vented two-inch voice coil, inverted dust cap and a Q_{TS} of 0.24 with F_S under 32Hz (from the six I've measured).

If you use two drivers in the D'Appolito odd order configuration satellites the combined sensitivity is approximately 96dB and the impedance will be 4Ω in a parallel hook-up. The L-pad will not be needed, or any other tweeter attenuation (to my ears, and ten band analyzer). I simply doubled the box dimensions and braced the panels, and obtained excellent results.

I also request any information readers may have on shielding, or good quality shielded drivers for video speakers.

Connol Reid
1823½ E. Sheridan St.
Phoenix, AZ 85006

OF ISOBARIKS AND FERROFLUIDS

I have three technical questions.

1. In *SB* 4/80, p. 14, Mr. Linkwitz offers a useful formula for compensating the roll-off of drivers before and after their resonant point in his *Fig. 25*. However, this formula does not work with ferrofluid damped drivers such as Dynaudio's D28 which has a Q_T around 0.3. Does Mr. Linkwitz have any suggestions for dealing with such drivers? I have contemplated using a shelving equalizer from f_s to the crossover frequency to produce a 12dB/octave slope. Such a circuit has been described by Walt Jung in *TAA* 1/75, p. 22.

2. Both Mr. Linkwitz and Mr. D'Appolito go to great lengths to match the acoustical and electrical responses to produce a true third or fourth order roll-off of the tweeters in their systems. Why don't they try to do the same with the roll-off characteristics in their low-pass sections? I can see how this might be difficult to do with a passive crossover, but an active circuit could easily shape the woofer/mid-high end roll-off.

3. I have been curious about the Isobarik design and put off building a new system waiting for the promised article on the subject. Although I appreciate Mr. Cockroft's efforts in *SB* 3/85, p. 7, I was dissappointed to see no scientific references to this design. Are there any? I have access to a library full of old *Journals* of the Audio Engineering Society and *Wireless Worlds* and I am eager to learn more about this fascinating design. Did Mr. Cockroft measure the f_B and Q_{TC} of his designs? To what extent does power handling increase with this design? Do both woofers have to face the same direction? Can they face away from each other as well?

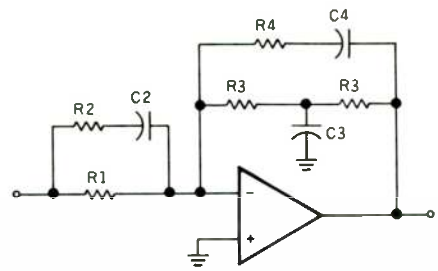
David Stanton
Seattle, WA 98122

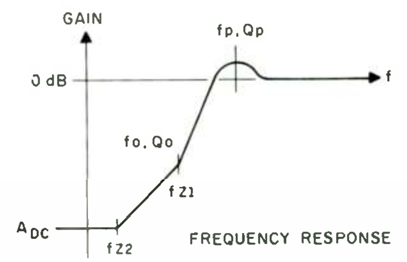
Mr. Linkwitz replies:

Mr. Stanton's question on how to compensate for the roll-off of a ferrofluid damped tweeter came at the right time. I was just in the process of modifying my loudspeaker system for the Dynaudio D28.

The compensating network in Fig. 25, SB *4/80, p. 16, must be changed slightly to handle Q_S of less than 0.5.* Figure 1 *is a new set of formulas for the design and the analysis of the new network; it is the easy part of the design. The difficulty comes in deciding which value of Q to compensate for.*

Continued on page 50





Design Formulas For $Q_o \leq 0.5$

1. Specify f_o, Q_o, f_p, Q_p
2. Calculate $f_{z1} = \frac{f_o}{2Q_o}(1 + \sqrt{1 - 4Q_o^2})$, $f_{z2} = \frac{f_o^2}{f_{z1}}$
3. Required $f_p > f_{z2}/Q_p$
4. Choose C_3
5. $R_3 = 1/\pi f_{z2} C_3$
6. $R_4 = 2R_3(Q_p \frac{f_p}{f_{z2}} - 1)$
7. $C_4 = \frac{C_3}{4}\left(\frac{f_{z2}}{f_p}\right)^2$
8. $R_1 = 1/f_{z1} C_4$
9. $R_z = \frac{R_1 R_4}{R_1 - R_4}$
10. $C_2 = C_4 \frac{R_4}{R_2}$
11. $A_{DC} = 20\log(f_{z1} f_{z2}/f_p^2)$

Circuit Analysis

1. $f_{z3} = ½\pi C_4 R_4$
   $f_{p3} = ½\pi C_2 R_2$
   $f_{p3} \approx f_{z3}$ required
2. $f_{z1} = \frac{R_2}{2\pi(R_1 + R_2)R_4 C_4}$
   $f_{z2} = \frac{1}{\pi R_3 C_3}$
   $f_o = \sqrt{f_{z1} f_{z2}}$
   $Q_o = \frac{f_o}{f_{z1} + f_{z2}}$
3. $f_p = \frac{1}{2\pi R_3 \sqrt{C_3 C_4}}$
   $Q_p = \frac{R_3}{2R_3 + R_4} \sqrt{\frac{C_3}{C_4}}$
4. $A_{DC} = 20\log(2R_3/R_1)$
   A_∞ 20log $[R_4(R_1 + R_2)/R_1 R_2]$

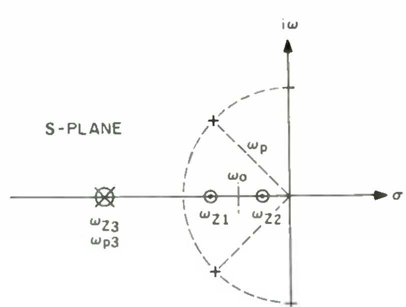


FIGURE 1: My new compensating network design and formulas.

SB *Mailbox*

Continued from page 48

When I bought the D28 only the 4Ω version was available. The published data sheet though, is for the D-2804-AF, which lists a Q_0 of 0.39 for the 8Ω version. I attempted to measure Q_0 using the method of Fig. 18 and 19 (SB 3/80, p. 13), using unit one I measured f_0 equal to 740Hz, Q_0 equal to 0.31, and with unit two f_0 equaled 88Hz, and Q_0 equaled 0.47, which is quite different. With the shallow impedance peak of R_{MAX} equal to $1.7R_{DC}$ the frequency f2 gets uncomfortably close to the impedance region where the voice coil inductance becomes dominant. I then use a more general and slightly more complicated formula, which allows you to determine Q_0 from a frequency f1 and impedance R1 below the resonance.

This formula is from A.N. Thiele, "Loudspeakers in Vented Boxes: Part II," JAES, June 1971, and I give it here in a form corresponding to my Fig. 18:

$$Q_0 = \frac{f_0 f_1}{f_0^2 f_1^2} \sqrt{\frac{1 - \left(\frac{R1}{R_{max}}\right)^2}{\left(\frac{R1}{R_{DC}}\right)^2 - 1}}$$

Using this I remeasured the units ten days later, and found disturbingly that f_0 had changed to 1050Hz for both units. Furthermore, Q_0 ranged from 0.29 to 0.42 for unit one depending on the frequency f1 from which Q_0 was determined. Unit two showed a variation of Q_0 from 0.26 to 0.41.

This means that the 6dB roll-off of the tweeter might start anywhere between 3.5kHz and 2kHz. My suspicion that the tweeter's behavior might be temperature sensitive was confirmed when I sprayed some Quick-Freeze® on the dome surround and observed an increase of the resonance frequency.

I then decided to remove the Ferrofluid. This takes some careful disassembly because the dome is glued to the circular front plate and adheres with double sided sticky tape to a rubber sealing ring glued to the magnet. With a little patience the plate and magnet can be pulled apart, but mark the alignment between the two. The fluid is easily removed with tissue paper. Re-assembly works best with a test tone applied before tightening the four screws to hear if the voice coil touches the magnet.

The modified tweeter now measured an f_0 equal to 700Hz and a Q_0 equal to 0.6. The Ferrofluid had not only increased the mechanical damping but also stiffened the suspension, and thereby increased the resonance frequency. Furthermore, the temperature sensitive viscosity of the fluid will change the resonance frequency and the damping with temperature. The variations which I observed were unacceptable to me.

The frequency response of the modified D28 had a broad peak around 2kHz which led to a further change. I removed the plastic rear enclosure that is glued to the magnet, and replaced it with a flat piece of cardboard. The hollow magnet pole piece I filled with a plug of long fiberwool to dampen resonances in the cavity behind the dome. This final modified version of the D-2804-AF now measured f_0 equal to 880Hz, Q_0 equal to 0.71 for unit one, f_0 equal to 910Hz, and Q_0 equal to 0.72 for unit two.

The maximum SPL, before audible distortion on shaped tone bursts, is 115dB at 1.6kHz and 1m, an improvement of 6dB over the HD 12.9 D 25. I think the improved definition and clarity from this tweeter made the replacement of the Son-Audax drivers worthwhile.

As for your question on the roll-off in the low-pass sections, the B110-1057 above 5kHz causes about 30 degrees of phase shift at a 1.6kHz crossover frequency to the tweeter. This should be compensated for in the tweeter channel by an all-pass network with the same phase response as that from the midrange roll-off. As first approximated, this mounting offset delay compensation in the tweeter channel is increased to match the increased delay of the midrange at crossover. Also see my crossover clarifications in SB 2/84, p. 36.

I will skip your third question about the Isobarik design principle since I have not been able to make sense of it. Possibly it produces good results for other than the claimed reasons, namely the substantial enclosure bracing required for its implementation.

Mr. Cockroft replies:

To the best of my knowledge no data has been published on rigid tests made with the Isobarik configuration. I'm sure that Linn has some of this information, but then I have not seen their patent disclosure (only their patent enclosure).

As I mentioned in my article, the only tools I had at my disposal were an old signal generator and my old ears. The speakers described in my article were based on Small's closed box theory and how I interpreted what my ears told me. Judging by some of the letters I have been receiving, it is possible that my ears have let me down (perhaps as a lifetime payback for having them lowered).

The prevailing opinion in those letters seems to indicate that resonant frequency is probably based on the sum of the combined masses, and is probably in the same box. This is possible, although I'm not completely convinced. I seem to recall that Dr. Bose, when he first came out with his 901 speaker, made a point of stating that he used various resonant frequencies, and they interacted with each other to give a smoother response. I think a similar response may be taking place here (please do not all hit me at once). Some writers also suggest that the two Isobarik units behave as a single unit, rather than two drivers in two environments. Here the ground does not seem so firm, and I think we must find out a lot more. My compelling reason for having written the article in the first place, was that I wanted to see more experimentation and information on this type of speaker.

I had measured (rather crudely) the internal woofer of project two (as mentioned in the article) when previously I had used it in an acoustic suspension system. I did not measure any of the other speakers either in or out of the system.

I cannot help on your other questions. You could add a series resistor to your Dynaudio tweeter to raise the Q, at the expense of power loss (do you really need the power?).

Mr. D'Appolito replies:

The numbered paragraphs respond to Mr. Stanton's numbered questions.

1. The low frequency roll-off of the D-28 transitions, from 6dB/octave to 12dB/octave, around 500Hz, is where it is down 20dB relative to 2kHz. Using a 6dB/octave crossover starting at 2kHz will put the tweeter down 32dB at 500Hz. At this level, correcting further roll-off to a true 12dB/octave with a shelving equalizer will have no sonic effect. In fact it may even degrade performance since the electrical roll-off is only 12dB down and substantial low frequency energy will pass into the tweeter, causing distortion and possible damage.

2. With regard to passive crossovers I think you have already answered the question. I believe that once individual drivers are down 20dB or more, correction of their roll-off curve to some desired or ideal response will probably have little sonic benefit. For example, the 5kHz roll-off of the B110 mid-bass drivers had little influence on the 2kHz crossover response of my satellites.

3. The low frequency theory of the Isobarik speaker is covered in my letters to SB 4/85, p. 49. The maximum acoustic output of this design is basically that of a single driver. Placing the drivers back-to-back greatly reduces the effect of suspension nonlinearities, and is recommended for woofer application where excursions are large.

MORE ON THE ISOBARIK

The Isobarik system (*SB* 3/85, p. 7) is certainly a worthwhile bass loading principle, although its merit may not be as great as Linn claims. Their explanation about how it works also seems a bit muddled.

A speaker in a sealed enclosure, which is what the Isobarik really is, is mass-controlled in the frequency range above the resonance frequency, resistance-controlled at resonance and stiffness-controlled below resonance. Thus, the cone excursion is controlled above resonance by the mass reactance of its cone and voice coil, while the stiffness of the suspension and the enclosed air is relatively unimportant.

Comparing the single driver sealed enclosure to the twin driver Isobarik, using similar drive units with equal voltage across each, they both have the same excursion in the midrange. As both Isobarik drivers exhibit the same excursion, no compression or rarefaction of the air trapped between the drivers occurs. The excursion is equal in the single driver case, both systems radiate equal sound power. The only difference is that the Isobarik system draws twice the power from the amplifier (i.e., efficiency is halved).

At low frequencies the cone movement is largely controlled by the stiffness of the trapped air in the enclosure. The volume of the tunnel between the two drivers is much smaller than the volume of the enclosure. This means the "spring" connecting the two drivers is much stiffer than the "spring" representing the enclosure. Thus, the two drivers are more strongly connected together than the rear driver's connection to the enclosure.

As the rear driver is pushing on the enclosure's stiffness, the front driver is helping by pushing through the tunnel stiffness. The force transmitted through the trapped air of the tunnel means the operation is no longer Isobarik. The two drivers, by virtue of their close coupling, work as one driver with twice the moving mass, but the same resonance frequency and the same Q. As the two suspensions are coupled together, the total stiffness is doubled and V_{AS} (the equivalent volume) is halved.

The conclusion is that with the Isobarik system you get the same bass response with half the enclosure volume as compared to the single driver system. This halving of the enclosure volume is, of course, paid for by doubling the power drawn from the amplifier.

A look at the impedance plot of the Linn Sara in *Fig. 1*, (from *HiFi Choice Loudspeakers*) shows the resonance frequency to be 48Hz, which seems probable for one KEF B200 in an enclosure with twice the volume of the Sara.

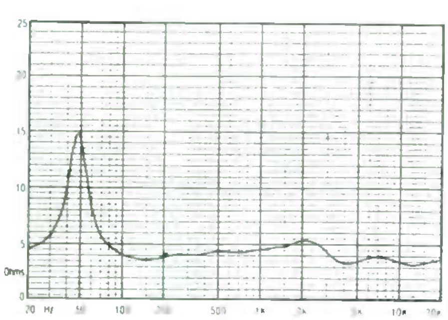


FIGURE 1: The impedance characteristics give an indication of amplifier loading for the Linn Sara. From *HiFi Choice Loudspeakers* (14 Rathbone Place, London W1P 1DE, England).

An improvement on the Isobarik theme is the configuration shown in *Fig. 2* which is probably what John Cockroft hints at in

SB *Mailbox*

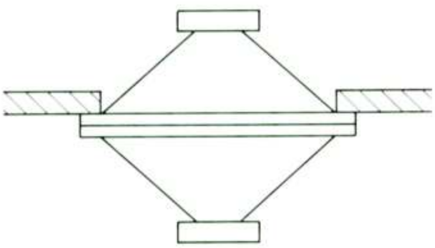


FIGURE 2: This improvement on the Isobarik is probably what Mr. Cockroft had in mind.

his article's last paragraph. The obvious advantage of bolting the two drivers directly together is the ease of construction. Some drivers can be bolted together directly, others must be mounted on different sides of the baffle. The less obvious advantage is that the push-pull action reduces second order harmonic distortion appreciably.

Measurements on a pair of bolted together KEF B139 units have shown a reduction of second harmonics of up to 20dB, as compared to a single driver. Mounting two drivers face-to-face means their asymmetric non-linearities are evened out. Thus second order distortion is reduced, but not third order. The face-to-back Isobarik mounting will probably give the same amount of distortion as a single driver system. As I see it there's a reason for facing your partner while dancing.

Here in Sweden the push-pull concept has been popular with bass reflex sub-woofers for some years, and several kits are available. I built a pair of bass speakers for use in a tri-amped system. The enclosure volume is 85L (three cubic feet), it is tuned to 20Hz and each enclosure contains a pair of push-pull mounted Philips AD 12250/WB units. The volume was calculated so that RH=0 (i.e., no overshoot), and the calculated F_3 is 33Hz. The intention is to take F_3 down to 20Hz with a bass boost circuit.

Even without bass boost the bass is deep, powerful and tight. I would rate it superior to the KEF B139 transmission line enclosures I once owned, especially for its tightness. The bass of the Linn Isobarik speaker, the Sara, can be described in the same way. A possible reason is that they all have a rather low resonance frequency with a low Q.

A slow roll-off, starting at a low frequency, gives a low group delay distortion in the audible band, and this seems to be important for good bass sound as explained by Laurie Fincham in the *Journal* of the Audio Engineering Society, June 1985.

The ability of the push-pull system to reduce distortion can also be used in systems with two woofers mounted side by side, by mounting one of them so its magnet points out of the enclosure. This is used by Audio Pro sub-woofers and the bass section of the new KEF 104.2 (at least the KEF works that way, even if both magnets are inside the box).

Summing up, the Isobarik system gives the possibility of deep bass from a reasonably sized enclosure at the expense of two extra bass drivers for a stereo pair. Alternatively, the money for the two extra drivers could be used for an active bass equalizer, which would give more flexibility and bass extension. As I pointed out earlier, with the single driver system the actual efficiency would be higher.

Leif Ryden
Uppsala, Sweden

MODIFYING THE AR-3

I recently came into possession of a pair of Acoustic Research AR-3 speakers. I may want to modify their crossover networks and the woofers as described in *SB* 2/82, p. 7 by Walter D'Ascenzo, but cannot find a non-destructive way of accessing the front panel. I would appreciate any advice on this matter. I have woodworking tools and I am not adverse to ripping the front grille lip, but I am presently hesitant for fear of damaging the woofer, whose frame extends to the full width of the speaker panel.

David J. Meraner
Schenectady, NY 12302

Mr. D'Ascenzo replies:

The biggest problem here is not the grille cloth, which must be destroyed in order to remove the grille frame. Simply find a spot over the approximate center area, above the woofer, cut the grille cloth with a razor knife, and pull away to reveal the grille frame. The grille frame is not glued onto standoffs, and can be pried off once you see where this can be done without damage to any of the drivers.

Your problems begin when you remove the woofer from the cabinet. The woofer is held in place with machine bolts and "T" nuts, and set in a movtite type sealer. The machine bolts may appear simple to remove, but can get very complex and frustrating if too much pressure is applied to your screwdriver. "T" nuts can easily dislodge from the particle board baffle, leaving you with a free spinning bolt that cannot be removed. In addition, the bolt and nut arrangement is also sealed with putty, which makes bolt removal even more difficult. Should you be faced with this difficulty, attach vice grip type pliers to the bolt head, while maintaining upward pressure (which reseats the "T" nut) and unscrew the bolt.

Your next problem is prying the woofer out of its sealed seat in the baffle. Use a large, broad bladed screwdriver, or a small pry bar, and carefully work your way around the frame of the speaker, prying an area at a time, until the speaker is free of the putty bond. Be careful. One slip, and you may end up with a hole in your woofer cone. The midrange and tweeter drivers are attached in the same problematical way as the woofer, with the further complication of having delicate external wiring which can easily be damaged.

Beside the improvements to the woofer as outlined in my article, you can also do the following:

- *replace the crossover capacitors with modern polypropylene types*
- *replace the level potentiometers with units of higher quality and wattage ratings*
- *replace the tweeters with high quality ribbon/planer types, like those offered by Foster and JVC.*

Do not bother changing crossover values, or fool around with driver phasing. AR did a fine job in these areas.

When you are finished, you will have a very up-to-date, high-performance speaker system, of reference quality, that will reveal many weaknesses in the rest of your audio system that went unnoticed before.

"BOXRESPONSE" IN PASCAL

After using Robert Bullock's "Boxresponse," *SB* 1/84, for a while, I decided to modify, combine, add to and totally rewrite it for the IBM PC. My version, written in Pascal, is now nearly 7000 lines of source code, and uses colors, graphics, pull-down menus and windows to perform different tasks. For example, once you have measured and calculated the standard Thiele/Small parameters for a driver the program can be used to try different types of closed, vented and passive-radiator designs.

The program provides a simple and smooth interface to the user, and can solve the majority of an amateur's speaker design problems without the need to know equations or use a calculator. Also you can edit driver parameters, change box parameters, wind an inductor, find the appropriate vent for a vented box, graph a frequency response or print your speaker design. Since the program can store up to ten designs, you can add the design you are currently working with into memory, and continue to try "what if?" changes to its parameters. . . . You can remove parts

Continued on page 54

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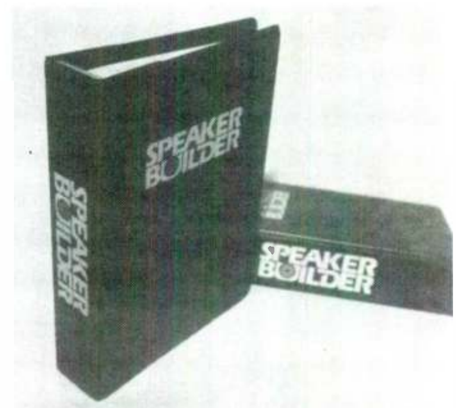

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SB *Mailbox*

Continued from page 52

of designs you no longer want, store them, and start over from the beginning.

The program operates quickly except when it has to calculate a new enclosure, then it takes several seconds. If you have an 8087 math co-processor in your PC, a separate version of the program is available that uses the co-processor, performing the calculations nearly instantly.

I am interested in having people use the program and return to me their criticisms and suggestions for improvements and additions.

Max Knittel
Bellingham, WA 98225

If readers are interested in a Pascal version of Boxresponse, let us know and we will ask Professor Knittel to write an article on the program. If you are interested in securing a copy of his program (IBM PC 5¼ inches) use Fast Reply no. 879, and if there is sufficient interest copies will be made available through Old Colony Sound Lab—Ed.

SOMETHING SEEMS AMISS

Could authors Linkwitz (*SB* 4/85, p. 24), and White (*SB* 4/85, p. 28), discuss and explain the following ambiguities in the results from their articles?

First, at 40Hz, White in his *Fig. 2* shows the 30W54 displacement limited at 98dB, whereas the Linkwitz nomogram, assuming 225mm diameter and 7mm PP shows an 88dB limit. That is a 10dB difference. Something seems amiss in their methods.

Second, the White curves show displacement limits for linear response. However, I believe that a speaker will fill in the voids shown in White's *Figs. 3, 5* and 6, with non-linear response. After all, the 30W54 is specified to have a maximum throw of one inch PP. It would be nice to have a method to determine what SPL can be achieved in the voids, at what power, what distortion, and which power levels cause bottoming. The reason is higher distortion levels may be acceptable at low frequencies, and bottoming can destroy drivers.

Third, can Mr. White explain how the four TP165F speakers with essentially the same piston area as the 30W54, and less X_{max}, can produce a better displacement limit than the 30W54 (*Fig. 2* curve *A* and *Fig. 7* curve *A* in White's article)?

Americo Borza
Sarasota, FL 33583

Mr. Linkwitz replies:

*According to my calculations, and corresponding nomograph (*SB 4/84, p. 25, Fig. 1*), I find that the sound pressure level is 88dB at 1mm distance from a 225mm diameter piston, which has a 7mm excursion PP at 40Hz rate, and is radiating into free space.*

To explain the 10dB difference between my calculations and those of Mr. White would require a close study of his article in SB 1/85, *p. 28 to see if a different definition for the response has been used. I'd rather leave it to Mr. White to clarify his results.*

Mr. White replies:

Thank you for your inquiry. It brings up some points about the relationship between drivers, their enclosures and the environment into which they radiate.

Question 1: *Using the article by Mr. Linkwitz (*SB 4/84*) I wrote a short BASIC program that does all the calculations; and it came up with 89.42dB at 40Hz for the Dynaudio 30W54. Now notice, there was a caution in the article that said the numbers were for free space. What this means is that the driver is suspended in space with no box or room. Usually there will be a floor under it (+3dB), and a wall behind it (+3dB), for a total gain of +6dB in a two environment. So, for a typical room, feel free to add +6dB to the number given by the nomograph. This now adds up to 95.42dB. We are not quite up to the 98.39dB at 40Hz that appears in* SB 1/85, *p. 29. The remaining +3dB comes from the damping effect the box has on the excursion. Generally, for sealed boxes, the smaller the box the less excursion allowed due to the restrictive nature of large alpha ratios.* Example: *A 30W54 in a 30 foot cubic box requires 22mm, and a 3.39 foot cubic requires 16¾mm.* NOTE: *the Dynaudio 30W54, with 210W of input will leave the gap (bottoming) at approximately 37Hz in each of the above examples.*

Question 2: *You are correct that the loudspeaker will "fill in the voids" to a certain extent, but keep in mind that as the loudspeaker is driven beyond its linear capabilities there are less and less turns in the gap. The fewer turns there are in the gap the less force is applied to the diaphragm, which results in less sound pressure level. Admittedly, this effect is most noticeable under full power using sine wave input, but manifests itself musically as sounding "wimpy" in the bass region. As for the second part of your second question, I have added some lines of code to* Boxresponse, *that calculate the required excursion for a given frequency and input power, and are tabulated in a new column as part of the output. (Code additions are available on request.)*

Your letter has caused me to consider the possibility of the woofer leaving the gap. As stated before, the Dynaudio 30W54 will leave the gap at 37Hz with 210W of input in almost all sealed boxes, but for instance will not leave the gap until 20Hz in a 7.08 foot cubic box tuned to 23.35Hz with the same 210W input. Notice how venting the box protected the driver to a much lower frequency. A proper choice of box volume and tuning will optimize the displacement-limited power handling capabilities of a driver. This statement becomes very apparent when you are designing and using Box Response *with the new added code.*

Question 3: *First of all the surface area of the two drivers differs slightly, the 30W54 has 62.0 inches, and the four TP-165Fs have 76.05 inches. So the Dynaudio 30W54 has 62.2 inches by 0.1378 inches = 8.54 inches cubed of displacement, and the four TP-165Fs have 76.05 inches by 0.13 inches = 9.89 inches cubed. Clearly the TP165Fs move more air. Second, as we saw earlier a vented box protects the driver better by reducing the required excursion. A third consideration is that the Dynaudio was receiving 210W in the illustration, and the TP-165Fs were only receiving 121W (-2.39dB). This just points out that the larger the diaphragm the less excursion required for a given SPL.*

Try this:

$$\mathrm{dB\ SPL} = 20\log\left(\frac{X_{max}\, f^2 a^2}{1180}\right)$$

Where X_{max} and the piston radius are both in millimeters:

$$X_{max} = \frac{(1.18 \times 10^3)\, 10^{SPL/20}}{f^2 a^2}$$

The above equations are limited to the driver alone in a half-space (room) environment at one meter.

AHEAD OF ITS TIME

The questions Fred Wagner raises in *SB* 4/84 (p. 38) were addressed as early as 1948 by Ben Drisko, then chief engineer of Massa Laboratories in Hingham, Massachusetts. Ben worked out a simple, practical method for grading horn drivers and routinely applied it to the selection of drivers for the horns he designed and built. He had no patience with scholarly publication, however, and never wrote up his methods. He gave me permission to do so, but with one thing and another, it took me an additional 12 years to produce the paper. It appeared in the *JAES* in 1962 (Volume 10, Number 4, p. 302).

Looking back, it is not a very good paper. Nevertheless, there was nothing else like it in print at the time, and I am somewhat chagrined that it has been overlooked by all anthologists and subsequent writers on horn design. All this took place decades before Thiele/Small, and I confess I have not established the correspondence between Ben's figure of merit and the Thiele/Small parameters. I would not be surprised, however, if the figure of merit turned out to be inversely proportional to Q_{TS}.

Continued on page 56

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Continued from page 54

Finally, I might add that the horns at issue were enlarged catenoidal variants of the Klipsch folded horn, still the best bass horns I have ever heard. I have the original blueprints and construction notes for these horns and could probably be persuaded to talk about them at the slightest encouragement.

Edward A. Fagen
Associate Professor
University of Delaware
Newark, DE 19716

Bruce Edgar replies:

The figure of merit to which Prof. Fagen refers is the quantity $F=(B\lambda)^2/M_M r_E$. It turns out that this figure of merit is not too far off the mark. If you divide F by pi, you end up with $2f_S/Q_{ES}$. This latter expression, as shown by Keele ("Low Frequency Horn Design Using the Thiele/Small Parameters," Preprint No. 1250, *presented at the May 1977 AES Convention), is the mass cutoff frequency for horn drivers. I ran across Prof. Fagen's paper a number of years ago and also wondered why it had been ignored by other writers.*

It is interesting that Prof. Fagen refers to Ben Drisko's corner horn design. The first bass horn I ever heard was a Drisko horn that a friend had in his Southern California home. I was completely bowled over by the impressive sound and started my horn quest, which SB readers have been following over the past five years. Incidentally, my friend learned his horn craft from Drisko while attending MIT in the late '40s. At one point, I started to build a Drisko horn, but had to give it up because I did not understand it. Since then, I have only built my own designs because I know where the design compromises are. Eventually, I hope to publish Drisko's bass horn design, with his permission.

BASS HORN BANTER

In *SB* 4/84, Fred Wagner asked about drivers for a low-frequency, folded bass horn ("Bass Speaker Selection," p. 38). Bruce Edgar replied that the Oaktron FEW12Y2 would be appropriate for that purpose. Who sells this driver, and roughly in what price range?

I, also, have the Speakerlab SK and am not very satisfied with the bass performance. Would the Oaktron driver make a suitable replacement with improved performance? I have already ordered the 1983 *SB* series and will look over the mini-horn plans in issue 2. The mini-horns would require a subwoofer, however, and I understand that they do not have excellent power-handling capacity. Therefore, I would prefer low-frequency bass horns.

I would appreciate any comments or suggestions.

Frederick J. Weber
Flushing, NY 11365

Bruce Edgar replies:

Unfortunately, the Oaktron driver did not work out for Mr. Wagner. He bought a pair and upon measuring their T/S parameters, discovered that his drivers had a Q_{TS} greater than 0.5. This high Q_{TS} destroyed the Oaktrons' usefulness as bass horn drivers. After several go-'rounds with Oaktron, the company's engineers found that a subcontractor was supplying voice coils with 25 percent more mass than was specified. Oaktron has supposedly corrected the problem, but Mr. Wagner has decided that he needs a driver with a larger optimum throat area. He is planning to try a JBL 2220H, which is a 15-inch driver with an optimum throat area of about 56 square inches. With the JBL driver, Mr. Wagner figures to shorten the bass horn length by several feet from the longer horn length associated with the Oaktron driver.

For your Speakerlab SK horn, I think the best replacement driver would be the ElectroVoice EVM 15L. It has an optimum throat area of 46 square inches, which is close to the SK horn size of 39 square inches. All the 15-inch, low-Q_{ES} drivers I looked at had optimum throat sizes that were either too small or too large. The EVM 15L has a 43Hz resonant frequency, which is just about right for the 35Hz flare rate of the SK horn. The mass cutoff frequency is about 350Hz.

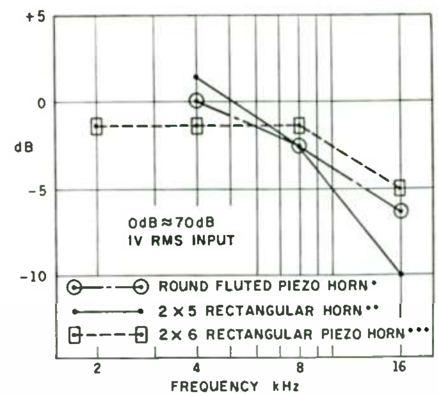
The EVM 15L is available by mail order from SRC Audio (3331 Towerwood Dr., No. 302, Dallas, TX 75234) for less than $150. Do not get the EVM 15S, as its higher Q_{ES} will produce a lower mass cutoff frequency. Occasionally, you might find an EVM 15L on the used market for about $100. If you do, test the Q parameters to see whether the driver meets your low-Q_{ES} specifications before you put down your money.

If any Speakerlab SK horn owner in Southern California wants to improve his or her bass horn, I will be happy to help with the mod. Please write to me care of SB. *I would also be interested to hear of other readers' experiences with SK horn modifications.*

PIEZO PERFORMANCE

I read with interest David J. Meraner's letter in *SB* 3/84 (p. 42). I also have tested and used piezo horns, but with somewhat different observations.

The specific tweeter horns I have tested are the round fluted horn, the 2-by-5 rectangular horn and the 2-by-6 rectangular horn. (See *Fig. 1* for model numbers.) For the testing, I used either a sound-level meter and octave bands of random noise or the ASA 10 octave band real-time analyzer (RTA) from Gold-Line Connecter Inc. and the PNG-1 pink-noise generator from Westside Electronics. In both cases, I conducted the tests in the reverberant field of a semi-reverberant room. I have made no attempt to measure the electrical characteristics of the tweeters or of any crossover networks.



* AVAILABLE AS PZ1 FROM UNIVERSAL SOUND (2243 RINGLING BLVD., SARASOTA, FL 33577); CATALOG NUMBER 40-1381 FROM RADIO SHACK; AND KSN1005 FROM MCGEE RADIO (1901 MCGEE ST., KANSAS CITY, MO 64108).

** AVAILABLE AS PZ2 FROM UNIVERSAL SOUND AND KSN1016 FROM MCGEE RADIO.

*** AVAILABLE AS PZ3 FROM UNIVERSAL SOUND; CATALOG NUMBER 40-1379 FROM RADIO SHACK; AND KSN1025 FROM MCGEE RADIO.

FIGURE 1: Frequency response of the three horns measured at 14′ 9″ in a 19′ 6″ L room at approximately 8° from the horizontal axis; 0dB to approximately 70dB, 1V RMS input.

The round tweeter and the 2-by-5 tweeter have approximately the same measured characteristics and to my ear the same sound. This sound is a harsh, one-note treble and a selective emphasis of program noise. The frequency response is characterized by a strong resonance (5 to 8dB) just before the low-frequency roll-off (approximately 5kHz). I disassembled one of the horns and found that the cone is driven by the piezo element through a pivoted stamped metal lever. The resonance, I believe, is caused by clearance (read slop) at the pivot points. The round and 2-by-5 units seemed to have the same driver.

The 2-by-6 tweeter sounds better and, not surprisingly, has a much smoother frequency response. The sound seems quite neutral at all frequencies and power levels. Of the two samples I measured, one was flat within the ±2.5dB resolution of my RTA from 2kHz to 16kHz and 10dB down at 1kHz. The other measured flat from

2kHz to 8kHz, 2.5dB down in the 16kHz band and 10dB down in the 1kHz band. I believe the cone on the 2-by-6 is driven directly by the piezo element, but I have not disassembled either of my two units to confirm this. To date, I have not measured any cone, dome or horn tweeter that has had as smooth an acoustic output.

I have done a little experimentation with C-R type crossover networks, as shown in David Weems's book *Designing, Testing and Building Your Own Speaker System* (Tab Books, 1981). In my experiments, I wanted to see whether the 5kHz resonance of the round and 2-by-5 tweeters could be tamed. The answer is a very limited yes. By the time the resonance is no longer apparent in the measurements, the output in the 8k and 16k bands is about 7.5dB less than it is without the crossover. I found no great difference in either the one-capacitor/one-resistor version or the two-capacitor/two-resistor version. I now use the 2-by-6 piezo tweeter in my speaker system without a crossover and am pleased with the sound. Whatever woofer or midrange you use with the 2-by-6 piezo should roll off or be rolled off so that it is 3dB down at 1.5kHz.

As a check, I reran my response tests *(Fig. 1)*. These tests agree closely with the original tests done in a different room, except in the 16kHz band. The current test room seems to be more absorbent in the 16kHz band due to a different type of wall construction.

Alan C. Smith
DeRidder, LA 70634

INDUCTANCE TEST

As a new subscriber to *SB* but a longtime speaker builder, I would greatly appreciate a solution to my current dilemma. Over the years, I have accumulated a number of open-air and ferrite-core coils for use in crossovers. I have no idea what the values of these chokes might be. Could someone explain in simple terms how I might determine the values using relatively inexpensive equipment? I have asked several service technicians, and none of them had any idea how to begin.

W. Fred Hart
Chicago Heights, IL 60411

Daniel P. Coyle replies:

In How to Design, Build and Test Complete Speaker Systems *(1st edition), David Weems suggests several ways of measuring inductance, but they all require a frequency generator and an oscilloscope or vacuum-tube voltmeter (VTVM). A slightly simpler way to measure inductance is to monitor the current flow through a coil while applying a known voltage.*

When an inductor is fed an AC voltage, it impedes current flow in proportion to its inductive reactance and the frequency of the applied signal. Where Z equals impedance or inductive reactance (measured in ohms), L equals inductance in henries, and f equals frequency,

$$Z = 2\pi f L$$
and
$$L = Z/2\pi f$$

Ohm's law states that the current flowing through a load is proportional to the applied voltage and inversely proportional to the impedance. Where I equals current in amps, and V equals voltage in volts,

$$I = V/Z$$
and
$$Z = V/I$$

Combining these formulas, we get

$$L = V/2\pi f I$$

You must apply a known voltage to the unknown inductor at a given frequency and measure the current flow. You can perform

SB Mailbox

all these functions with a volt-ohmmeter (VOM) and a frequency generator.

Allow me a digression on instruments. A VOM enables you to measure peak RMS AC voltage, DC voltage, resistance and milliamps. This is an essential piece of equipment for any kind of electronics hobby. Radio Shack stocks some cheap VOMs that are acceptable.

A frequency generator is a little more expensive. It is very useful for any audio hobby, from speaker testing to recording. For an entry-level instrument, I can recommend the $50 Heathkit Model IG-5282. It provides square and sine-wave output, although its amplitude varies over the frequency range by 0.5V in a 3V signal. You can mitigate this using a voltmeter across the output and adjusting the generator's amplitude control. Also, the frequency scale must have separate scales for each range. You can apply these numbers (in different colors for each range) by hand at the appropriate places on the dial. Locate the true frequencies with a frequency counter. After completing the kit, I suggest you have someone with an oscilliscope and frequency counter do the final adjustment. The IG-5282 produces symmetric sine waves and square-edged square waves when properly set up.

The test setup for Mr. Hart's coils is shown in Fig. 1. *Hook up the mystery inductor and the VOM (in the milliamp function) in series*

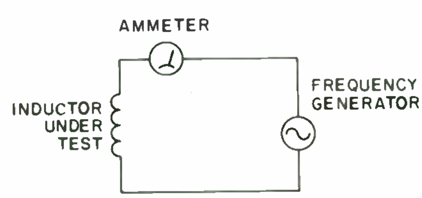


FIGURE 1: Test setup for determining the value of unspecified coils.

with the frequency generator. Starting at the highest frequency range so as not to damage the VOM, lower the frequency successively through the ranges until the milliammeter registers a mid-scale deflection. This is the value of I. The frequency at which this reading is taken is f. Next, removing the VOM from the circuit, connect the coil directly to the frequency generator and measure the voltage across the coil. This is the value of V. This method of measurement provides an accurate voltage factor regardless of the generator's regulation, while disconnecting the coil and reading V directly from the generator might not.

Consider a small inductor being fed 2V by the frequency generator at 6kHz. If the value of the current flow is 260mA (0.26A), then

$$L = 2/2(3.14)(6000)(0.26) = 0.0002\text{H, or } 0.2\text{mH.}$$

A larger inductor would not draw an equivalent current until a lower frequency. You may also use this method to determine a speaker's voice-coil inductance to calculate crossover values or impedance-compensating networks. If you have a coil of known inductance, you can use it to calibrate your test setup.

Unless inductors are made from 18-gauge wire (or thicker) and you have identical pairs, I question their suitability for quality speaker systems. Inductors have a Q factor (inductive reactance compared to DC resistance) and two coils that have the same inductance, but different DC resistances do not behave exactly the same throughout the audio spectrum. You will have to judge by ear the significance of this effect on imaging. It is probably less than the typical variations in a production run of speakers.

My thanks to Dave Hubbell of New Mexico State University at Grants for suggesting the measurement of current as a means to evaluate inductors.

A VOTE FOR VARIETY

Although I love to read some of the articles in *SB*, I think the magazine could use more variety. For instance, perhaps some readers would be interested in a project such as my set of "pipe organ" speakers for my car. I like the idea of suspending speakers in space, and the pipe is one way to do this. I have not yet experimented with this idea, but the pipes would suspend and isolate the speakers, while also providing their positioning.

On another note, I have been very curious about materials used in stuffing closed boxes. Has anyone run a complete test of different quantities and types of stuffing material, including wool, polyester and cotton? Of special interest are comparisons between the various types of polyester such as hollow versus solid and constant diameter versus mixed batches.

My applications often require the use of materials such as cotton to prevent midrange or high-bass sounds from reverberating inside. Organic materials work well for this purpose. The only problem with cotton is that it absorbs moisture, but you can use it sparingly on opposite walls. I am sure wool would be excellent, but I cannot find it in the stores where I shop. *(Editor's Note: Check the Trade section of our Classifieds for a source of long-hair wool.)* Heavier foams are appropriate for the lower frequencies. Fiberglass works well, but I try to stay away from it as much as possible for the obvious reasons. In most cases, I try to use a variety of materials instead of just one type. I would like to hear from anyone with information about the various materials.

Gregory J. Szekeres
Pittsburgh, PA 15236

Kit Report

Model G continued from page 30

Audio Concepts replies:

Our thanks to Dave Davenport, Dick Marsh and SB for the review of and feedback on our Model G. Their efforts enabled us to make several improvements to the kit. We have a few additional comments to share with readers.

The tweeter now supplied with the Model G is a Seas H297, which is almost identical to the older H253 but has a cast faceplate with a flat foam front. The H297 is slightly smoother, with relatively less output below 10kHz and more above 10kHz.

All Model Gs include the internal two-inch foam to which Dick Marsh refers. The builder need not purchase as an extra. Although we have not had previous assembly problems with midrange systems, all are now supplied pre-assembled. The enclosures are now also drilled for, and supplied with, T-nut mounts for the woofers. The cabinet volume is under 1.5 cubic feet.

The sonics of the Model G depend, of course, on the source equipment and material used. The two reviewers do not seem to agree on the sound quality of the system. Our best testimony is from the many warranty cards we have received. They have been overwhelmingly favorable from acoustic music lovers as well as amplified music lovers. The system sounds like many of the better mass marketed "east-coast sound" speakers that cost $700 to $1400 a pair, and find favor with listeners of classical music.

For those with a bit more money to spend, we recommend trying our Vanguard loudspeaker, which uses the same cabinet with two Dynaudio drivers per side. It has improved high-end midrange performance, at a slight increase in cost.

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An increasing number of readers are telephoning for information about their personal classified ads. Ads are filed as received, set in type late in the magazine production cycle, and put on page approximately one month before the magazine is mailed. Delivery of the mailed magazine can take from four days to three weeks.

Personal classified ad copy that is included in one issue is discarded when it goes on the page, and a new file for the next issue is begun. Ads arriving after the issue closes will be run in the next issue.

We strongly suggest that you keep a carbon or photo copy of your ad. We ask your cooperation in following these rules so that we can give you the best possible service.

1. All "For Sale" ads must be personal sales and not for profit. Ads for resale at a profit must be submitted as "Trade" ads at 55¢ per word, prepaid.
2. Ads are limited to one per issue, 50 words or less. A word is any collection of letters or numbers with a space on both sides of it. Words over 50 must be prepaid at 20¢ per word. Longer ads not prepaid will be discarded and not published.
3. Ads cannot be returned. We will make every effort to substitute a revised ad for an earlier submission if both are clearly labeled with the same name and address at the top of the sheets.
4. Copy for ads or corrections to ads will not be taken by telephone. We cannot answer telephone queries about whether or not an ad is in a particular issue.
5. Ad copy should be clearly printed in block capitals or typed. Illegible ads will be discarded.
6. If you include only your name and a telephone number in your ad, your full name, street address, city, state and zip must accompany the copy.
7. Personal ads are a free service to current subscribers. It will help us a lot if you include your subscriber number (upper left corner of your mailing label) with your ad.
8. If you want an acknowledgment of your ad, including in which issue it will appear, please include a stamped, self-addressed postcard with your ad copy.

Classified Advertising

CONNECTICUT AUDIO SOCIETY is an active and growing club with activities covering many facets of audio—including construction, subjective testing, and tours of local manufacturers. New members are always welcome. For a copy of our current newsletter and an invitation to our next meeting, write to PO Box 346, Manchester, CT 06040 or call Mike at (203) 647-8743.

WASHINGTON AREA AUDIO SOCIETY (N. VA, DC and MD) is looking for sincere audiophiles who are eager to devote their time and get involved with the direction of the society and the publication of a monthly newsletter. Please write: Horace J. Vignale, 5434 Taney Ave., Alexandria, VA 22304-2002.

SAN FRANCISCO BAY AREA AUDIOPHILES. Audio constructors society for the active, serious music lover. We are dedicated, inventive and competent. Join us in sharing energy, interest, expertise and resources. Send self-addressed, stamped envelope to S. Marovich, 300 E. O'Keefe St., East Palo Alto, CA 94303 for newsletter.

THE AUDIO SOCIETY OF HONOLULU cordially invites you to attend one of our monthly meetings and meet others like yourself who are interested in the hows and whys of audio. Each meeting consists of a lively discussion topic and equipment demonstrations. For information on meeting dates and location, contact Craig Tyau, 2293A Liliha St., Honolulu, HI 96817.

MINNESOTA AUDIO SOCIETY. Monthly programs, newsletter, special events include tours and annual equipment sales. Write Audio Society of Minnesota, PO Box 32293, Fridley, MN 55432.

THE INLAND AUDIO SOCIETY IN THE SAN BERNADINO-RIVERSIDE AREAS, recently formed, is now inviting audiophiles in the San Diego, Los Angeles and Orange Counties to join us. Our goal is to share common interests, ideas, construction points, modifications and system changes, and other members' equipment at every meeting. Plans for the future are to invite audio luminaries to lecture, and to incorporate and include "live" music occassionally. We are presently meeting every 5-6 weeks (subject to change). Audiophiles interested contact Frank Manrique, President, IAS, 1219 Fulbright Ave., Redlands, CA 92373, (714) 793-9209.

ORGAN MUSIC ENTHUSIASTS: If live recordings of fine theatre pipe organs or electronic organs are your thing, I have over two thousand of them. I lend you the music on reels or cassettes. All operation is via the mail. A refundable dollar will get you more information. E.A. Rawlings, 5411 Bocage St., Montreal, Canada H4J 1A2.

THE COLORADO AUDIO SOCIETY is a group of audio enthusiasts dedicated to the pursuit of music and audiophile arts in the Rocky Mountain region. We offer a comprehensive annual journal, five bimonthly newsletters, plus participation in meetings and lectures. For more information, send SASE to: CAS, 4506 Osceola St., Denver, CO 80212, or call Art Tedeschi, (303) 477-5223.

THE VANCOUVER AUDIO SOCIETY publishes a bimonthly newsletter with technical information, humor and items of interest to those who share our disease. We have 40 members and meet monthly. Six newsletters per year. Call (604) 299-4623 or write Dan Fraser, VAS, Box 4265, Vancouver, BC, Canada V6B 3Z7. We would like to be on your mailing list.

MEMPHIS AREA AUDIOPHILE SOCIETY being formed. Serious audiophiles contact J.J. McBride, 8182 Wind Valley Cove, Memphis, TN 38115, (901) 756-6831.

FOR SALE

ReVox B77 tape recorder, $895; Denon DP 2500 with 103D cartridge, $325; JBL 077 slot radiators, $95 pair. Wesley Kellie, 3805 Sherwood Dr., Coeur d'Alene, ID 83814, (208) 765-8521 after 5 p.m. PST.

Yamaha M-80 amp, $700; C-80 preamp, $500; tuner, $175; T-700 turntable, $100; SAE 2900 preamp, $175; Nakamichi Dragon cassette deck, $1400; Denon DP62L turntable, $450; Audio-Technica 160ML, $175; Grado GF + 1, $40. John W. List, American Embassy, MSG Det., APO San Francisco, CA 96503.

Sontek bass amp/crossover from kit, 75 and 110Hz modules, $75; pair Becker 15" woofers sold by Sontek, unused, $45; Shadow passive/active first order crossover from kit, 275, 300 and 400Hz modules, $65; pair Dynaudio 21W54s, $45; pair Strathearns without transformers, $100. Francis LeJeune, (504) 866-7381.

Pair Peerless TX255F 10" woofers; pair Audax HD13B25H4C12 5" Bextrene midranges; pair Peerless TP165R 6½" poly mid/woofesr; pair Panasonic EAS 400 leaf tweeters. All unused. $135 firm. Micro-Seiki MA505L transcription length tonearm, excellent, $125. Dean Carl Johnson, 370 E. Tratebas Rd., Valparaiso, IN 46383, (219) 462-3171.

Harman Kardon Citation II, distortion checked plus extra output tubes, $300; complete set, Citation II, III and IV, all with manuals and operating, $495. F. Blanding, 6 Cornell Rd., Cranford, NJ 07016, (201) 276-5132.

Precision Fidelity C7 cascode preamplifier, 100 hours of use, $200; Stax SR-X MK III electrostatic headphone with SRD-7 power supply, $200, Dennis A. Billo, 1145 Wilson Ave., Chicago, IL 60640, (312) 989-6052, 9-12, 1-3 CST, Monday-Friday.

Becker 910A100F 10" woofers, never used, $48 pair; Audax HD 12x9 D25A 1" dome tweeters, never used, $18 pair; Audax HD13D27 1½" dome mid, mounted, $32 pair; Falcon Acoustics Nightingale, two-way speakers, built from kit, excellent balanced sound, two pairs, one used, $175 (used), $240 (new). Dan, (602) 838-2789.

Radford harmonic distortion analyzer, like new condition, battery powered, serial no. DMS90113, $150 ($800 new); EICO Model 377 sine/square generator, no manual, $25; Heath FM test alignment generator, $25; Dynaco FM-3 tuner, no cover, as is $25; Viking tape deck with spare heads and tube-type record/play electronics, as is, $60; two new ElectroVoice D054 dynamic mikes, 150Ω with calibration, cases and cables, $75 each. Audio Amateur Publications, Dept. UE, PO Box 576, Peterborough, NH 03458.

Futterman H-3aa/OTL-3 amps, $1400 pair; Futterman CS-1 OTL headphone amp, $400; Audio Research SP-10 preamp with RAM tubes/spare tubes, $2600; Counterpoint SA-TT phase inverter, $150; Stax Lambda headphones with 10 meter extension cord, $200; Harman-Kardon HK-400, three-head cassette deck, $300; Fulton Gold six-foot speaker cables, $30 pair. Ted Kircher, 6859 Arlene Dr., PO Box 141, N. Kingsville, OH 44068, (216) 266-5916 days, (216) 224-2866 evenings.

ElectroVoice's finest ultimate woofer, new, EVM-18B Pro-line, serial no. 0200082265, 400W, 8Ω, 18" subwoofer, fs = 33Hz, Qt = .36, $200 includes data, 5 year warranty, insured shipping. D. Jensen, c/o A.V.M., PO Box 328, Nevada City, CA 95959, (916) 273-5328.

General Radio 1551-C sound level meter, excellent condition, with manual, $150; G/R 1900A wave analyzer, unused, $100 with manual; H/P 5232A DC-1.2MHz frequency counter, excellent condition, $30 with manual. Mark Matthews, 3318 Shield Lane, Garland, TX 75042, (214) 495-0039 evenings and weekends.

NYAL Moscode 300, $600; Hitachi HCA8500 MK II preamp, $200; Sherwood 6040CP, dual mono, MOSFET, $200; new AR tonearm, (Mission), $50; new Dyna 410, $300; Dyna 70, $100; Dyna SCA 35, $100; Pioneer SF850 electronic crossover, $175. Roger Artman, Box 2463, Grass Valley, CA, (916) 272-8118.

Gold Sound 10" dual voice-coil woofers, four pieces, $25 each; Gold Sound 8" dual voice-coil woofers, two pieces, $20 each; Audax HD 12x9 D25 tweeters, two pieces, $7 each. All are slightly used. Dan Greene, 216 Auburn Ave., Santa Cruz, CA 95060, (408) 425-8956.

Harman Kardon Citation 12 amp, $115; Tannoy 15" dual concentric speaker, $75; Dyna PAT-5 preamp, $85; Dyna ST-70 amp, $65; Sony ST5055 AM/FM tuner, $75; Teac A-6010 reel tape deck, $245; Eico HF-89 100W amp, $50; Quad adapter (Dyna type), $10; Stromberg Carlson AU 36B 200W tube PA amp, $55. David Cummings, Ridge Rd., RD4, Cazenovia, NY 13035, (315) 687-9454.

General Radio model 1350-A audio generator/chart recorder assembly and extra accessories, $1000. Johan VanLeer, 230 Bay St., Santa Monica, CA 90405, (213) 396-3005.

McIntosh 30W power amplifier, model A-116 and Audio compensator preamp, model C108, with instruction manuals, good condition, $100 delivered. R.B. Jensen, 8 Heather Lane, Reading, PA 19601.

Experimenter's delight: KLH 9 set with non-working power supplies. Looks like new, in original cartons, $50 plus freight. Bob Tucker, (215) 627-5326.

Technics RS-M253X cassette deck, three heads, B-C, DBX, bias adjust, under 200 hours, list $450, sell $220; Pioneer F-77BL AM/FM tuner, direct digital decoder, S/N 86/81, new in box, list $225, sell $160; ADC SS-315X ten-band equalizer, spectrum analyzer, pink noise generator, electret microphone, feature laden, as new, list $380, sell $150. John (904) 641-5830 evenings.

Nakamichi 700 microphone system, two omni, one cardoid capsules, attenuation add-on, windscreen mount, calibration curves, used occasionally for measurements only. Forward money order for $185 (US funds) plus $10 shipping and insurance to Steven Clark, 9 Armadale, Toronto, M6S 3W7 Canada, (416) 766-8186.

WANTED

Jensen KT-31 Imperial three-way speaker system kit. Matthew F. Callahan, 395 Roosevelt Pl., Grosse Pointe, MI 48230, (313) 882-9103.

Heathkit harmonic distortion analyzer IM-5258. R. Orford, 23-19th St., Hermosa Beach, CA 90254, (213) 376-6827.

Can anyone provide a copy of the "Daline" article by Robert Fris in the November 1974 issue of *Hi-Fi News & Record Review*? I am willing to pay for postage and copying. Dave Davenport, 626 VanThomas Dr., Raleigh, NC 27609.

Dynaco Quadaptor. Walter L. Marple, 4156 Belle Pk. Dr., Houston, TX 77072.

Advertising Index

McGee Radio

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- 1 12" Dual Voice Coil, 8 OHM, 40 oz. Magnet Sub Woofer.
- 2 6-1/2" Polypropylene Curve Linear Cones, Butyl Edge , 30 oz. Magnet, 1.33" VC Mid-Bass.
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- 2 1" 10 oz. Soft Dome Tweeters.
- 2 12 dB (Second Order) Phase Coherent Crossovers, 150 Watts
- 1 Dual Input Subwoofer Crossover.

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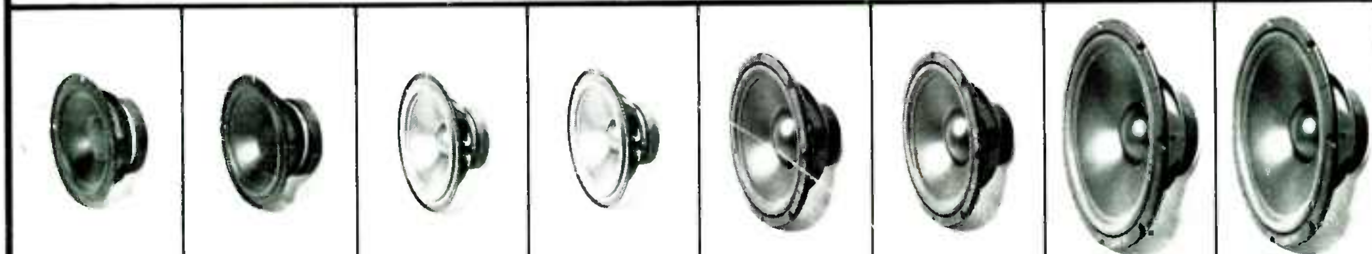
If for any reason, you don't like the system, send it back.



TP165R TP165F TX205F TD205R TX255F TD255F TA305FSW

Never say die! That was the feeling of several employees of the old Peerless factory in Leominster, Mass. Through determination and solid business planning, PLI (Precision Loudspeakers Inc.) was born. Headed by the Vice President of National Sales, Howard Jacks, and the Wizard of Production Control, Bob Clark, PLI purchased all of the original equipment used to make those quality American-made models listed here. **McGee Radio** is proud to offer these quality woofers to you at substantial savings. Call for quotes in case quantities.

Speaker:	*Size*	*VC*	*Imp*	*Mag*	*fs*	*O+S*	*VAS (Cu.Ft)*	*Power**	*Eff*	*Retail Price:*	*McGee Price:*
TP165F	6-1/2"	1.25	4&8	15oz.	53	.42	.55	80	89	27.95	15.95
TP165R	6-1/2"	1.25	8	15oz.	33	.321	1.41	80	80	38.95	21.95
TX205F	8"	1.25	4&8	20oz.	42	.40	1.73	80	90.5	34.95	19.95
TD205R	8"	1.5	8	27oz.	32	.326	2.12	100	90	47.95	26.95
TX255F	10"	1.25	4&8	20oz.	28	.527	4.31	100	88	40.95	22.95
TD255F	10"	1.5	8	27oz.	23	.252	5.47	150	90	48.95	27.95
TA305F	12"	2.0	8	38oz.	25	.306	6.64	200	89.5	54.95	30.95
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- 1 Crossover. 2nd Order, Phase Coherent, 150 Watt.

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