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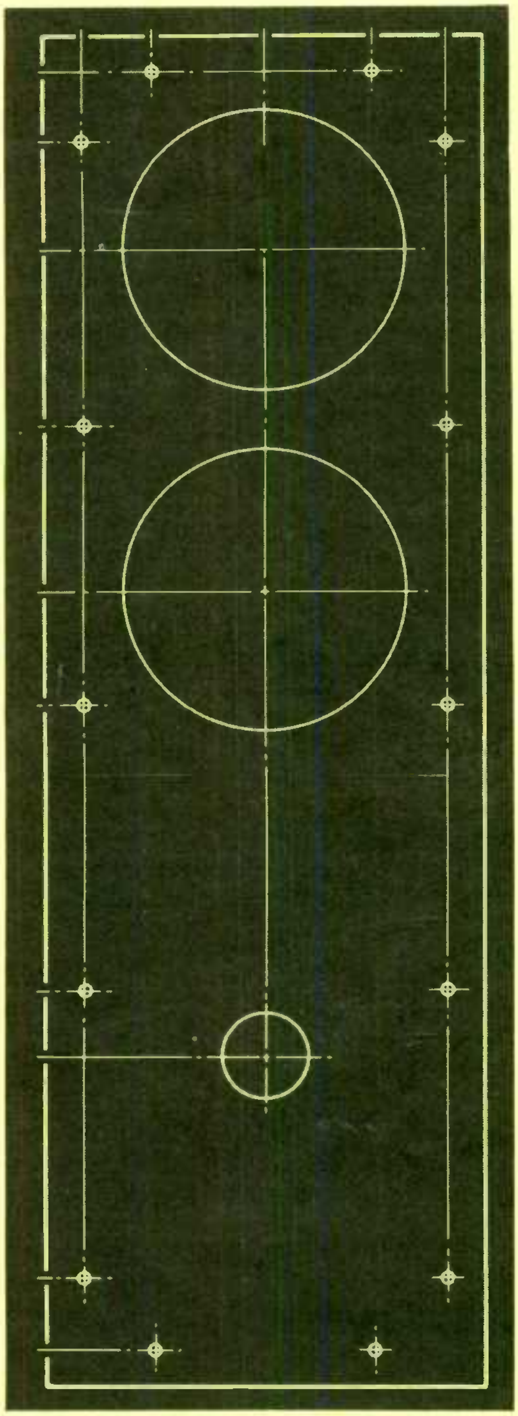
# SpeakerBuilder

THE LOUDSPEAKER JOURNAL



## LEVREAULT
*Sixth Order Vented Bass*

## WEEMS
*Tapered Pipes*

## EDGAR
*Inside Boston Acoustics*

## COCKROFT
*Tiny Double Woofer*

## GONZALEZ
*Loudspeaker Modelling*

## Reviews

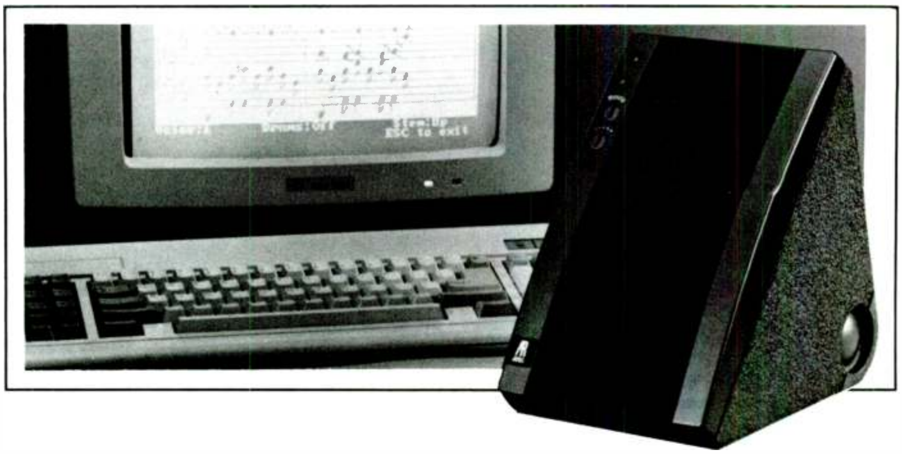
**GALO**
*Audio Control's Richter Scale*

**BULLOCK**
*Loudspeaker Design Cookbook*

# Good News

Users of **ACOUSTIC RESEARCH** Powered Partner speakers have found the units are compatible with personal computers and music software packages. The speakers have shielded drivers, so you can use them near your monitor and disks safely. The Powered Partner was originally designed for portable compact disk (CD) players, personal stereos, and MTS-encoded television. Their suggested retail price is $359.95/pair. Contact Teledyne Acoustic Research, 330 Turnpike St., Canton, MA 02021, (800) 225-9847.

***Fast Reply #FB909***



**OHM ACOUSTICS** has released two new speakers: the Ohm Sound Cylinder and the Ohm 3XO. Both use Ohm's patented inverted-cone driver. If you send two yards of fabric, Ohm will turn it into a speaker wrap for your Sound Cylinders. You can also order any of seven standard finishing kits, available in aluminum, black, white, and genuine wood veneers. The Sound Cylinders, which come with a standard "walnut-grain" vinyl finish, have a suggested retail price of $549 per pair, plus $76 per pair for optional finishes, and $46 per pair for custom finishes.

The Ohm 3XOs are said to offer the full dynamics of compact disks, even when used with amplifiers that supply only 10W. Their sensitivity rating is 91dB, and their frequency response extends to 22kHz ±4dB. The suggested retail price is $1,495/pair. Contact Ohm Acoustics, 241 Taaffe Place, Brooklyn, NY 11205, (705) 783-1111 for more information.

***Fast Reply #FB375***

**BOSTON ACOUSTICS** has introduced the T830, its new tower speaker system, which is a three-way floor-standing unit featuring an 8″ woofer, a 3½″ midrange, and a 1″ CFT dome tweeter. Frequency response is 48–24,000Hz, ±3dB. The speakers can handle 100W and have an efficiency rating of 90dB/1W/1m. The T830 is available in a "rosewood" vinyl veneer and has a molded grille with tapered inner edges that the manufacturer claims virtually eliminates sonic diffractions. The suggested retail price is $500/pair.

The company has improved two of its speakers, the A70 and the A150 II. The new versions, the A70 II and the A150 III, both feature new drivers designed and manufactured by Boston Acoustics in the US. The A70 II two-way speaker includes a new woofer, crossover, and bookshelf-size enclosure. The A150 III, a three-way floor-standing unit, has a newly designed 10″ woofer, 3½″ midrange, and CFT 1″ dome tweeter. The suggested retail prices are $300/pair A70 IIs and $550/pair A150 IIIs.

Contact Boston Acoustics, 347 Lynnfield St., Peabody, MA 01960, (617) 532-2111.

***Fast Reply #FB336***

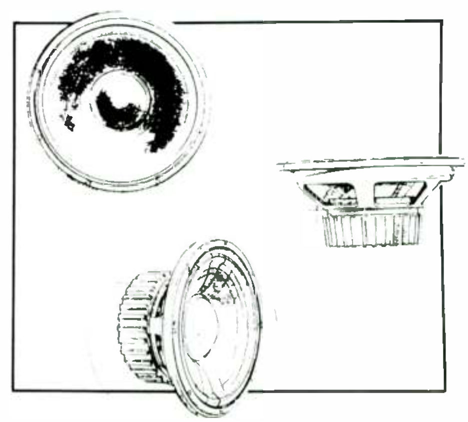
**INFINITY SYSTEMS, INC.,** has announced that the RS6000, the new top of its RS series, is a floor-standing model featuring 2″ polydome-k midrange and EMIT-k high-frequency drivers. The EMIT-k tweeter's diaphragm weighs half as much as its predecessor and is surrounded by neodymium magnets, offering high-frequency response to 44kHz. The midrange features an edgewound voice coil. The RS6000 has a round-edged enclosure and a minimum-diffraction grille, a system that the company claims provides natural spatial relationships among instruments and voices. The speaker's suggested retail price is $425 each.

Infinity systems has changed its recently introduced Reference Standard 7 Kappa speaker from bookshelf to floor-standing design. The company claims the enclosure's larger volume increases bass response and overall sound quality. The suggested retail price of the Reference Standard 7 Kappa is now $659.

For more information, contact Infinity Systems, Inc., 9409 Owensmouth Ave., Chatsworth, CA, 91311, (818) 709-9400.

*Fast Reply #FB354*

**ATC** (Gypsy Lane, Aston Down, Stroud, Gloucestershire, England) produces an extensive line of professional studio loudspeakers, power loudspeakers, and studio control monitors. They range in price from $180 for a 9″ studio loudspeaker to



$5,800 for the top-of-the-line pair of studio control monitors. The company also offers a three-way electronic crossover. In the US, contact Audio Ecstasy, 231 Elmwood Dr., Rochester, NY 14616, (716) 865-9938.

*Fast Reply #FB31*

According to the *Journal* of the Audio Engineering Society (Vol. 35, No. 1/2, January/February 1987) Leo L. Beranek's *Acoustics* is now available in paperback. *Acoustics* is a compilation of texts of courses the author has taught on subjects including electromechano-acoustical circuits, sound radiation, microphones, noise control, loudspeakers, hearing, speech intelligibility, and psycho-acoustic criteria. The book is intended as a textbook, and costs $15 per single copy and $12.50 each for five or more copies. For more information, contact the **ACOUSTICAL SOCIETY OF AMERICA**, 335 E. 45th St., New York, NY 10017.

*Fast Reply #FB43*


# SPEAKER BUILDER

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# LISTEN WITH US...

## PERFECTION

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

*NOMEX and KEVLAR are registered trademarks of DUPONT*
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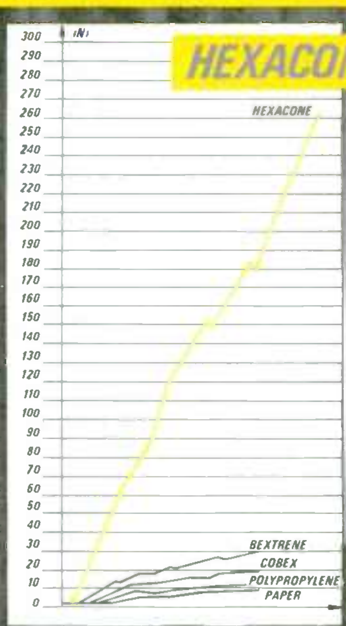
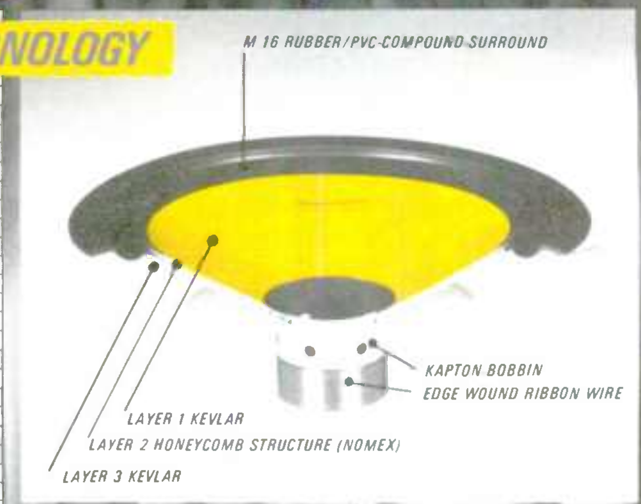
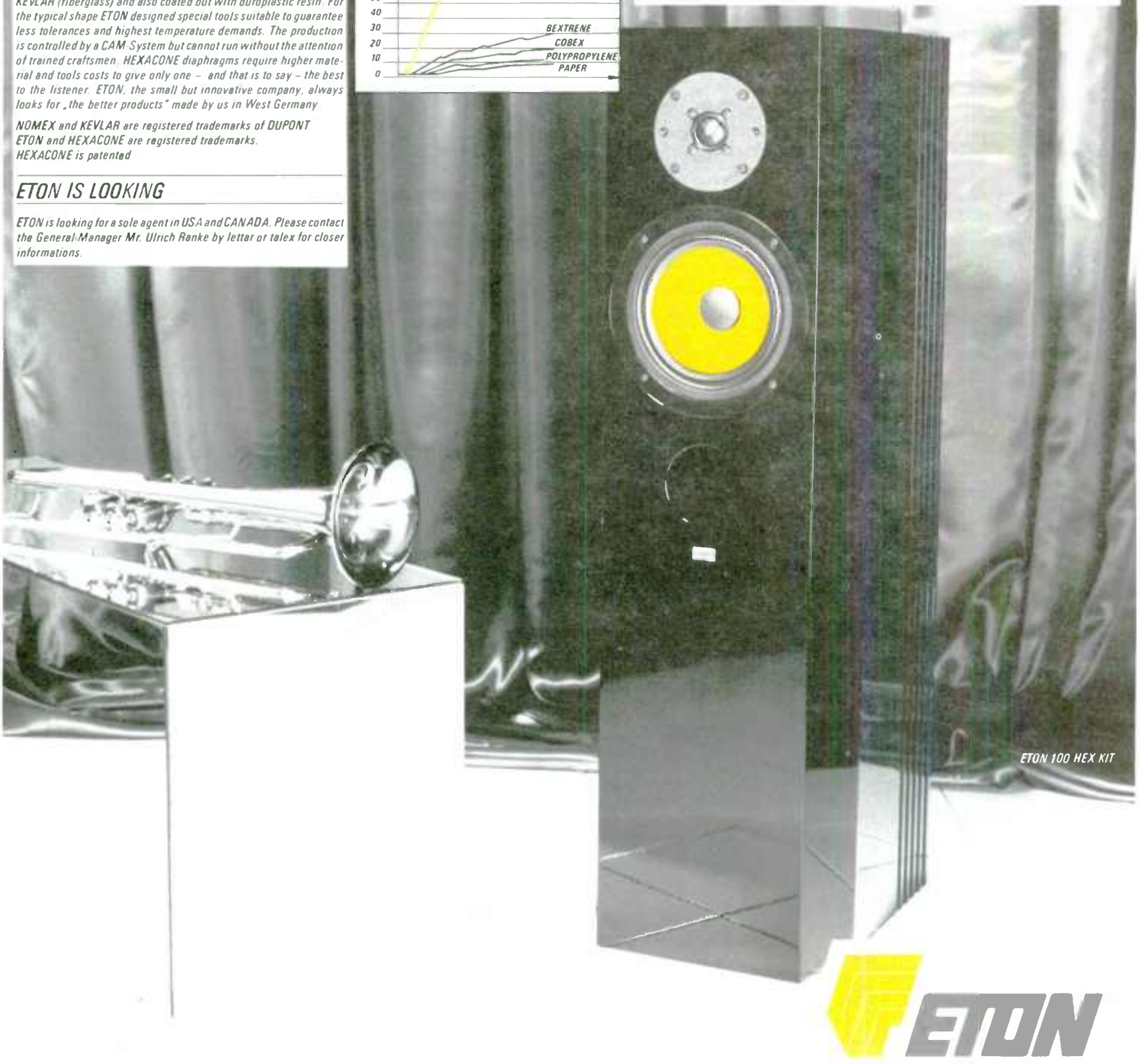
*HEXACONE is patented*

## ETON IS LOOKING

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



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# About This Issue

**John Levreault** has a passion for loudspeaker design, a comprehensive professional knowledge of electronic components and a strong aversion to carpentry. These attributes will be evident in his sixth-order vented design beginning on page nine, the second of three subwoofer designs we will publish in 1987. **David Weems** ventures gingerly into what is, for him, new territory on page 18. He offers three transmission line designs with commentary on their characteristics.

When Contributing Editor **Bruce Edgar** came to Boston last year he visited Boston Acoustics' plant in Peabody, Massachusetts. Andy Petite, who designed loudspeakers at KLH and Advent, now does so for his own company. He talks freely about the tradeoffs and traumas of bringing a good loudspeaker to market (p. 33).

**John Cockroft** is with us again, offering what he calls his Demonstrator on page 29. **Ralph Gonzalez** continues the exploration of his loudspeaker modelling program starting on page 42. If you are wondering whether your subwoofer is properly set up for your room, Audio Control's *Richter Scale* can answer your question and then optimize the system's low-end response. Contributing Editor **Gary Galo** examines the unit starting on page 50. Don't miss Contributing Editor **Bob Bullock's** review of the third edition of **Vance Dickason's** *Loudspeaker Design Cookbook* (p. 41).

With the greatest regret I report the deaths of two persons who have contributed in outstanding ways to our knowledge of audio. **Richard C. Heyser** died in March after a brief illness at Tujunga, California. Although an aerospace industry professional, Mr. Heyser regulary contributed papers on loudspeaker technology to the *Journal* of the Audio Engineering Society, of which he was a Fellow. He also wrote illuminating reviews and articles for *Audio*. **Hugh D. Ford**, also a Fellow of the AES, reviewed professional equipment for the British publication *Studio Sound* for many years. He was a staunch proponent of thorough and critical audio equipment reviews, and wrote many outstanding examples. Both men will be sorely missed.

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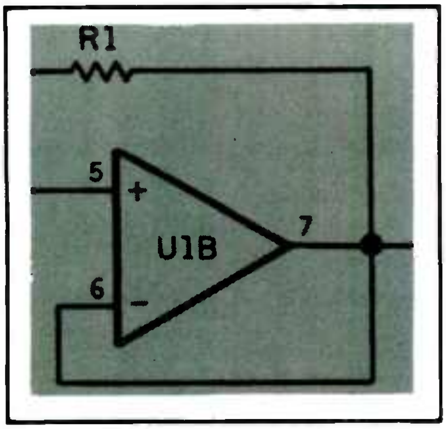


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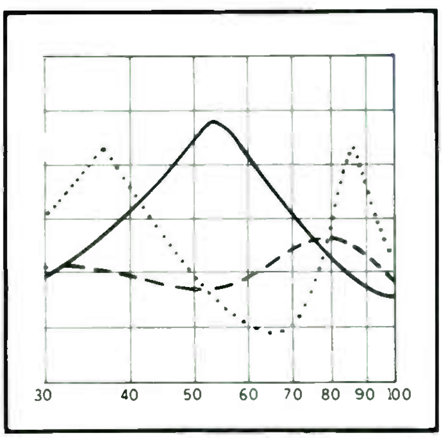


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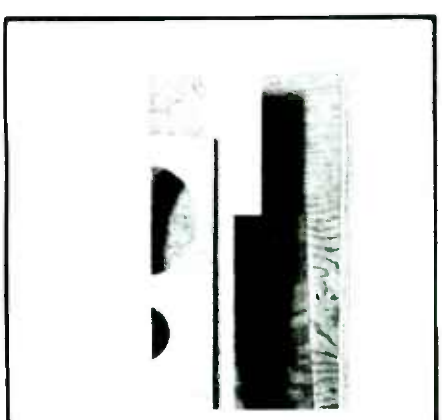

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## FEATURES



## DEPARTMENTS

# Editorial

## THE TTT TEST

Back in the good old days when radio was in flower Bob Elliot and Ray Goulding of Bob & Ray fame used to do a routine panning cigarette ads. Setting the scene in Boston's old Scollay Square among all the vaudeville theatres, they did their "Sturdley Short Cigarette Test." In those days the king-size version of America's favorite carcinogens were new and the ad agencies were constantly doing man-in-the-street interviews presumably asking smokers to try their product on the spot. Bob and Ray's victim always coughed uncontrollably before the scene ended and asked in a whisper, several times during the interchange, "When do I get the money?"

I have in these pages bemoaned the craven fear that seems to strike most American hearts at the sight of anything more technologically complex than an egg beater or a bottle opener. I have decided it is time to bring this malaise out in the open, once and for all. To do this I propose a simple test anyone can use to determine whether their loved ones or friends are subject to this deadly and crippling syndrome I call technological fright.

Every civilized human in the western world must, sooner or later, face the harrowing experience of changing the toilet paper roll. For some people it is a task better left to others and they will, in craven fashion to be sure, leave the task to the next occupant. Generally the next roll is set out in a convenient spot for the unlucky person who gets half-way through this lonely business of intensely personal hygiene. I will not here detail the difficulties involved when the fresh roll has not been provided conveniently. Some things are too painful to discuss with even so sophisticated an audience as this.

When the awful moment arrives, however, that we are faced with the dire, not to say, unavoidable chore of removing the empty gray tube, unwrapping the fresh supply and then facing the extraordinarily demanding dual hazard of finding the end, we then face the genuinely daunting challenge of successfully detaching it from the roll.

If these hurdles are passed successfully, (even if there are 30-50 sheets or partial sheets stuffed into the bowl to be flushed into deep-sixed safety so as not to embarass the technician) then comes the final test of technological *savoir faire*. We here put aside the genuinely daunting variety of tissue roll holders, with their devilish variety of spring-loaded mounting spools that can, without warning, fly apart and across the floor of the bathroom. Retrieving them successfully with various garments hobbling your feet can be aggravating in the extreme.

Our Toilet Tissue Technology Test is simple. Only one question of technological rectitude is involved here. "Which way do you mount the roll?" An ancillary question is, "Do you notice which way you mount the roll?" One further question might be, "Have you ever noticed that there are two ways to mount a new roll of tissue?"

There is, in fact, only one correct, which is to say convenient, way to mount the tissue. One makes it very easy to unroll the paper, the other makes it excessively difficult at least 50% of the time. Mounted one way the end of the paper is always out in a position where it may be grasped. Mounted the other way, there is a 50/50 chance the end will be at the back of the roll and therefore very difficult to grasp.

Check the toilet tissue rolls in your home. How are they mounted? Do they roll outward or inward? Are the members of your family aware that mismounted rolls indicate a technological shortcoming that is the result of inattention, ignorance or capricious neglect of the technological abc's not to mention the convenience of others?

I suggest a family conference of a most serious sort to discuss this matter thoroughly. To some this may seem trivial. But it has profound import for us as the world's leading technological nation. Early sensitization of all of us to technological issues such as this very simple one could lead, in time, to a better relationship for each of us with the world of mechanical gadgetry in which we all live. The increased sensitivity will affect the mounting of other rolls such as the paper towels, for instance. People may even start to put clothes on coathangers facing the same direction, and avoid the *faux pax* of putting the hanger on the rod with the open side facing outward.

I have high hopes for this new aid to increased technological sesitivity. The Toilet Tissue Technological Test will not be patented or copyrighted in order that it may have the most free and wide usage possible. It is our contribution to the enhanced relationship all Americans can enjoy with their environment.

Spreading the good news in this manner will be a boon to many. Some, of course, will continue in their insensitive ways. Those we will have with us always. But I doubt not that most people of good character will embrace this simple yardstick with good humor and grace. Let us hope for the sake of the nation, not to mention generations yet unborn, it is so.—E.T.D.

# A SIXTH ORDER VENTED WOOFER SYSTEM

BY JOHN E. LEVREAULT, JR.

The recent demise of the woofer portion of my bi-amplified speaker system has given me the opportunity to experiment with some new system design techniques. I had long been attracted to the art of vented box systems design, as espoused by Robert Bullock in *Speaker Builder*, and chose to pursue some of his methodologies in developing a new low frequency unit for my homebrew loudspeaker system. The following design is a conglomeration of techniques learned through several years of speaker building experience in addition to my first attempt at vented system design based on Bullock's articles.

**PHOTO 1: A completed vented woofer system.**

**REQUIREMENTS DEFINED.** As mentioned above, I use a biamplified speaker system. For the midrange, I use a pair of Strathearn ribbon transducers on each side of my stereo pair. Finding a woofer that blends acceptably with the Strathearns has proven to be quite difficult due to the transparency and naturalness of the ribbons. I achieved good sonic results with a pair of Audax 8″ Bextrene woofers in a closed box, a system described in Craftsman's Corner, *SB* 3/81. I was able to improve on the sound of the closed box system by using aperiodic damping. I cut a hole in the back of the box and stuffed the hole with foam, a technique similar in principle to the Dynaudio "Variovent."

Unfortunately, my preference for loud rock music and its attendant power demands eventually caused the drivers to fail. My latest experiments involved the Seas P25REX 10″

polypropylene-cone drivers, which sounded very nice but couldn't handle much power or transcend into the deep bass regions. Clearly, a new approach was needed to fill the gap between the lowest sonic reaches and the 200Hz crossover point to the Strathearns. Rather than continue with closed box designs, I decided it was time to apply some of the newer technology presented in *SB*.

**SPECIFICATIONS.** The first step in any new design, be it speakers or computers, is the specification. This is where you make a list of all the things you would like this new product to do, in other words, a "wish list." Here's my list:

Low frequency cutoff = 30Hz
Maximum frequency = 500Hz
Maximum response ripple = 0.1dB
Maximum cabinet size = 4 ft³
Minimum power handling = 150W
Minimum SPL capability = 110dB

To explain, I primarily listen to records, which means I wish some kind of equalized system. I don't listen to organ music or cannons, so I am satisfied with 30Hz response. My cabinet size is limited by my desire to avoid woodworking wherever possible. In order to match the Strathearns' dynamics, which are used in a fairly large (8x14x24') sound room, SPL capability is very important to me. Response ripple is undesirable in any modern system.

From the above, I concluded a sixth-order vented alignment was most likely to meet my objectives. I studied Mr. Bullock's *SB* 1/82 article on sixth-order design, and I suggest you study it too, since I will refer to it periodically.

The first thing you notice with the alignment charts is where the transition from QB5 to C6 alignments occurs. This transition point is the B6 alignment, which is unique and will likely prove difficult if not impossible

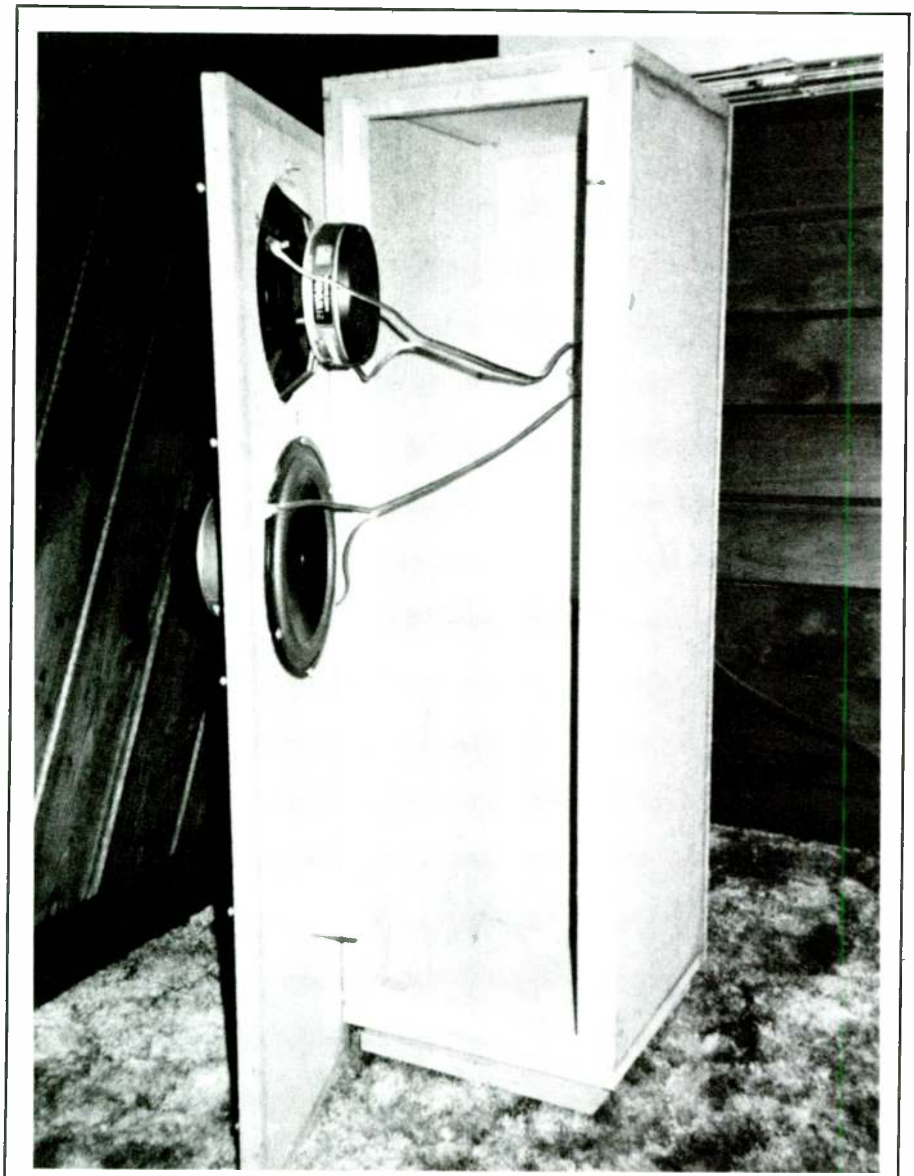


**PHOTO 2: Inside the box, note where author used pine rinners, foam strips, and patio bricks for volume adjustment.**

**TABLE 1**
*Target Sixth Order Response*

| FREQ (Hz) | RESP (dB) | POWER (W) | SPL (dB) |
|---|---|---|---|
| 20 | -24.37 | 154 | 86.7 |
| 25 | -13.54 | 211 | 98.9 |
| 30 | -5.82 | 220 | 106.8 |
| 35 | -1.81 | 189 | 110.1 |
| 40 | -0.50 | 94 | 108.4 |
| 50 | -0.05 | 97 | 109.0 |
| 60 | -0.01 | 145 | 110.8 |
| 80 | -0.01 | 220 | 112.6 |
| 100 | -0.01 | 220 | 112.6 |
| 120 | 0 | 220 | 112.6 |
| 150 | 0 | 220 | 112.6 |
| 200 | 0 | 220 | 112.6 |

**TABLE 2**
*Estimated Closed Box Response*
*(2 drivers in parallel)*

$V_{as}$ = 4.6
$R_e$ = 3.2
P = 440W
D = 11"
α = 1.76

| FREQ (Hz) | RESP (dB) | POWER (W) | SPL (dB) |
|---|---|---|---|
| 20 | -17.18 | 105 | 95.2 |
| 25 | -13.96 | 123 | 99.1 |
| 30 | -11.53 | 145 | 102.3 |
| 35 | -9.65 | 175 | 104.9 |
| 40 | -8.17 | 212 | 107.3 |
| 50 | -6.03 | 317 | 111.1 |
| 60 | -4.59 | 440 | 114.0 |
| 80 | -2.88 | 440 | 115.7 |
| 100 | -1.95 | 440 | 116.6 |
| 120 | -1.40 | 440 | 117.2 |
| 150 | -0.92 | 440 | 117.7 |
| 200 | -0.53 | 440 | 118.1 |

**TABLE 3**
*Measurement data*

Left channel:

| | | | |
|---|---|---|---|
| $f_s$ | = 29.6 | $f_c$ | = 50.3 |
| $Q_{ms}$ | = 1.944 | $Q_{mc}$ | = 3.83 |
| $Q_{es}$ | = .3453 | $Q_{ec}$ | = .6665 |
| $Q_{ts}$ | = .2932 | $Q_{tc}$ | = .568 |

α = 2.28
$V_{as}$ = 5.7 cubic ft

Right channel:

| | | | |
|---|---|---|---|
| $f_s$ | = 29.8 | $f_c$ | = 50.6 |
| $Q_{ms}$ | = 2.036 | $Q_{mc}$ | = 3.810 |
| $Q_{es}$ | = .3591 | $Q_{ec}$ | = .6367 |
| $Q_{ts}$ | = .3053 | $Q_{tc}$ | = .546 |

α = 2.01
$V_{as}$ = 5.0 cubic ft

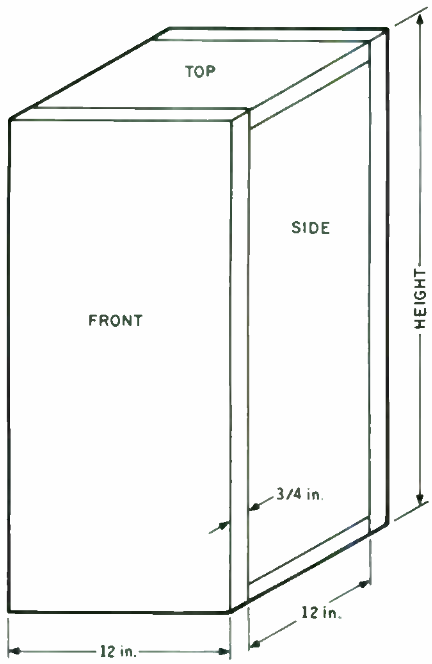



**FIGURE 1: Arrangement of 12″ wide by ¾″ thick panels used for the author's simplified construction technique.**

to obtain. Preferred low ripple alignments will, therefore, be obtained only for driver Qs below the B6 transition point, or perhaps slightly above. For a B6 Type I alignment with $Q_l$ = 7 (see *Table 3* in the referenced article), this transition occurs for driver $Q_{ts}$ = .31 and $\alpha$ = 2.32. This relatively large $\alpha$ (which = $V_{as}/V_b$, the ratio of the driver compliance-equivalent volume to box volume) should allow a modest box size.

**DRIVER SELECTION.** I focused my search for the ideal driver on 8″ and 10″ plastic cone types with a $Q_{ts}$ of about .3 and high power capacity. Experience has shown me that small drivers offer the best transient response and lowest coloration, which are prerequisites for a seamless blend with the critical Strathearns.

After combing through driver data sheets, I settled on the Dynaudio 21W54MPS, which has the following characteristics:

$$
\begin{aligned}
f_s &= 30\text{Hz}\\
Q_{ts} &= .30\\
V_{as} &= 2.3\text{ft}^3\\
\text{diameter} &= 8.75''\\
\text{power capacity} &= 220\text{W}
\end{aligned}
$$

I have used Dynaudio products in the past and have always been impressed by their quality. Optimistic about my chances for success, I proceeded.

**ALIGNMENT ANALYSIS.** To examine the suitability of my drivers, I turned to the computer and the most powerful tool available to the loudspeaker designer: BOXRESPONSE. This wonderful program, published in *SB* 1/84 and originally written in BASIC by Mr. Bullock for the Apple computer, lends great insight into the suitability of any driver for a low frequency application. Although I don't own an Apple, I was able to modify the program listing for my Commodore VIC-20. This allowed me to proceed with the necessary analysis.

I invoked BOXRESPONSE with the above Dynaudio 21W54 parameters, plus the following data from Dynaudio and the alignment data from Bullock's *Table 3* to determine the resultant response:

$$
\begin{aligned}
\text{driver D} &= 7.75''\\
R_e &= 6.4\Omega\\
x_{max} &= .25''\\
Q_l &= 7\\
h &= 1.014\\
\alpha &= 2.5305\\
f3 &= 33\text{Hz}\\
f_c/f_s &= 1.0945\\
A &= .5642
\end{aligned}
$$

The resultant data is summarized in *Table 1*. This alignment features a small box, good low frequency extension, and good SPL capability. Given my positive experience with such designs, I chose to use two drivers in a push-pull arrangement to increase the SPL capability. The box size would still be less than two cubic feet.

I procured four of the necessary drivers which maintain the outstanding quality of other Dynaudio products. These drivers feature cast frames and a light stiff cone. I performed some quick resonance checks and found resonant frequencies and Qs about 10% higher than specified, but very consistent. Whereas drivers require several hours of break-in before their characteristics settle, I didn't attach too much importance to their parameters "right out of the box," but I did want to verify they were functional. So far, so good.

**BUILDING THE BOX.** I calculated the target volume for the box from the formula:

$$V_b = \frac{V_{as}}{\alpha} = \frac{(2.3 \times 2)}{2.53} = 1.82\text{ft}^3$$

I took Mr. Bullock's advice and oversized the box by about 40%, to about 2.5 cubic feet. Understand I do not design and build loudspeakers because I enjoy woodworking. Rather, I consider woodworking a necessary evil in the enjoyment of quality sound. Consequently, I have developed some shortcuts to box fabrication which don't seem to have compromised the design's integrity.

I have found that a narrow enclosure is preferable to minimize diffraction effects, a necessity in maintaining a good blend between woofer and Strathearns. Obviously, the enclosure can be no wider than the diameter of the driver plus the thickness of the two enclosure side walls. Using ¾″ particle board construction, the minimum enclosure width is:

$$9 + (2 \times .75) = 10.5''.$$

You'll find it surprisingly convenient to use standard 12″ wide particle board shelving stock for all panels. If you construct your box as in *Fig. 1*, the total internal volume will be equal to 10.5 × 12 × (height − 1½) cubic inches. Furthermore, using standard 4′ lengths of the shelving means minimal cutting. This is exactly what I did in the following sequence, starting with eight pieces of 4′ long by 12″ by ¾″ particle board shelving.

1. Take a 48″ long piece of 12″ wide by ¾″ thick particle board and orient it in front of you on your work surface

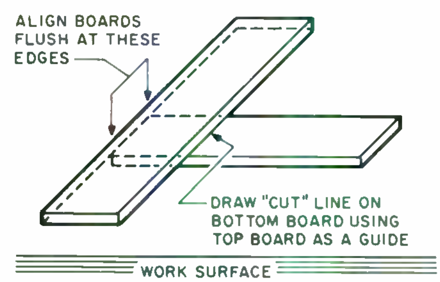



**FIGURE 2: Use the width of one panel to measure the length of another.**

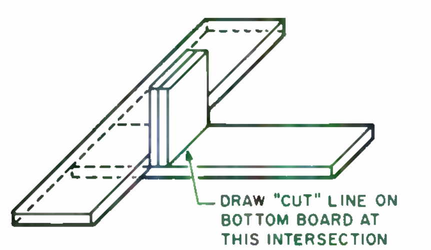



**FIGURE 3: Method for determining the length of front/back pieces.**

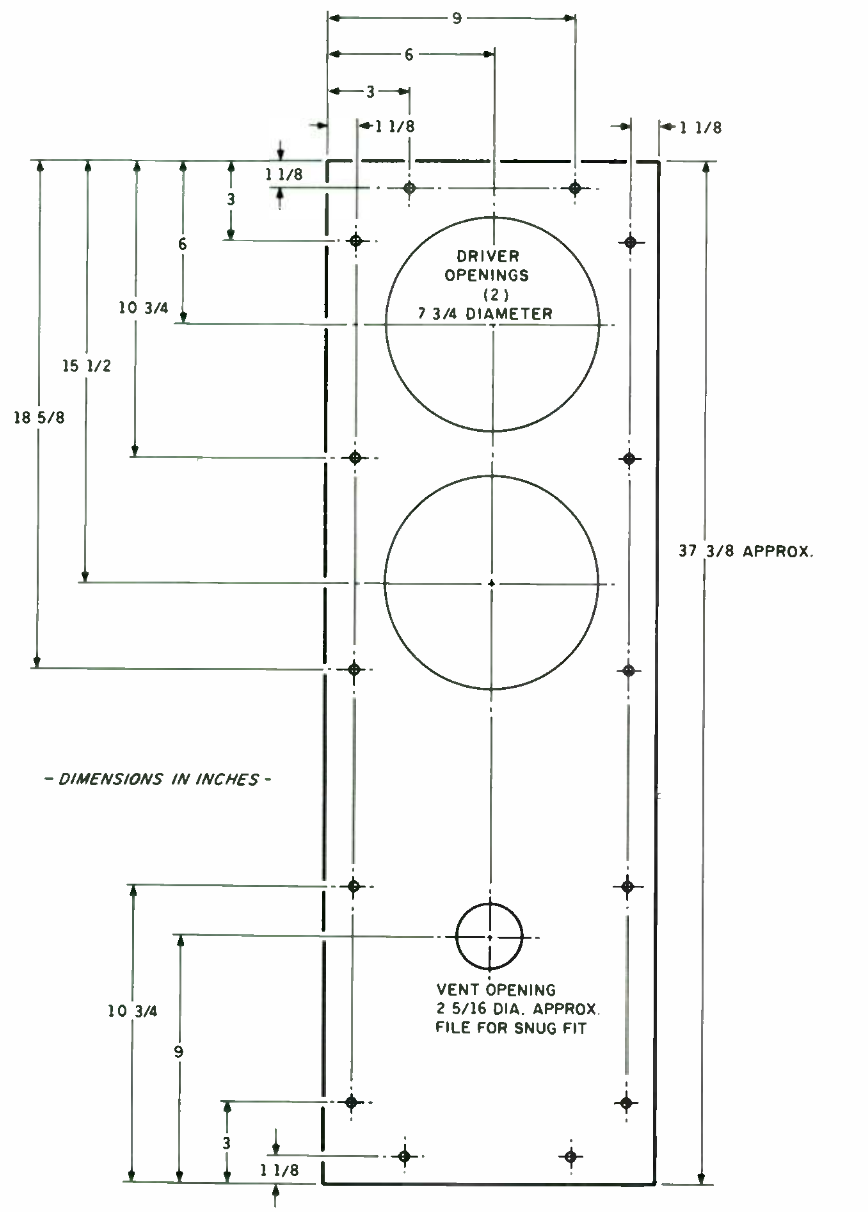



**FIGURE 4: Front panel layout.**

with a 48″ side closest to you. Take a second piece of the same material and place it on top of the first piece at a 90° angle (the side of the top piece flush with an end of the bottom piece). Draw a line on the bottom piece using the width of the top board as a guide, thus marking a 12″ length along the bottom board. See *Fig. 2.* Using a table, radial or circular saw, cut off this 12″ length, being careful to stay on the correct side of the drawn line. Remember, you want a piece of wood whose length is the same as the width of the other. This 12″ piece is a top (or bottom). Place the leftover piece, which should be about 35⅞″ long, aside. This is a side panel.

2. Repeat the above procedure on three more 48″ boards. You will now have four 12″ pieces (two tops and two bottoms) and four sides about 35⅞″ long.

3. Now, cut your front and back pieces, which should be as long as a side plus the thicknesses of the top and bottom pieces. Lay another 4′ long piece in front of you (the long side closest to you) and place a side piece directly on top of the future front/back. Place two top/bottom pieces on top of each other and butt the top of this stack against the end of the side piece on top of the front piece. Line up the end of the front piece with the end of the side piece and draw a line after the second top/bottom piece. See *Fig. 3.* Cut on the appropriate side of the line, and you will have a front/back piece about 37⅜″ long.

4. Repeat step 3 on three more uncut boards, and you will have four boards about 37⅜″ long, two fronts and two backs.

The net result will be a cabinet with an internal volume of 10.5 × 12 × 35.875″, or 2.62 cubic feet. This design has good rigidity, since all panels are very narrow, and could be reinforced along the long dimension at your discretion. In addition, by placing the drivers at the top of the enclosure's front panel, midbass response will be enhanced.

Furthermore, you don't even need a ruler or tape measure to measure the panel dimensions. This compensates for slightly inaccurate board widths. Four boards and four cuts with your

**Table 4A**
*Actual closed box response*
(Left channel)

$f_s$ = 29.6 $V_{as}$ = 5.7
$Q_{es}$ = .3453 $R_e$ = 2.7
$Q_{ms}$ = 1.944 $\alpha$ = 2.28

| FREQ (Hz) | RESP (dB) | POWER (W) | SPL (dB) |
|---|---|---|---|
| 20 | -18.04 | 112 | 95.2 |
| 25 | -14.66 | 126 | 99.1 |
| 30 | -12.08 | 144 | 102.3 |
| 35 | -10.06 | 167 | 104.9 |
| 40 | -8.45 | 197 | 107.3 |
| 50 | -6.13 | 282 | 111.1 |
| 60 | -4.58 | 410 | 114.3 |
| 80 | -2.78 | 440 | 116.4 |
| 100 | -1.84 | 440 | 117.4 |
| 120 | -1.30 | 440 | 117.9 |
| 150 | -0.84 | 440 | 118.4 |
| 200 | -0.48 | 440 | 118.7 |

**TABLE 4B**
*Actual closed box response*
(Right channel)

$f_s$ = 29.8 $V_{as}$ = 5.0
$Q_{es}$ = .3591 $R_e$ = 2.7
$Q_{ms}$ = 2.036 $\alpha$ = 2.01

| FREQ (Hz) | RESP (dB) | POWER (W) | SPL (dB) |
|---|---|---|---|
| 20 | -17.49 | 115 | 95.2 |
| 25 | -14.15 | 130 | 99.1 |
| 30 | -11.61 | 150 | 102.3 |
| 35 | -9.63 | 176 | 104.9 |
| 40 | -8.07 | 210 | 107.3 |
| 50 | -5.82 | 305 | 111.1 |
| 60 | -4.34 | 440 | 114.2 |
| 80 | -2.62 | 440 | 115.9 |
| 100 | -1.73 | 440 | 116.8 |
| 120 | -1.22 | 440 | 117.3 |
| 150 | -0.79 | 440 | 117.8 |
| 200 | -0.45 | 440 | 118.1 |

*Continued on page 14*

*Continued from page 12*

**TABLE 5A**
*Equalized closed box response*
(Left channel)
A = .52, $f_e/f_s$ = 1.40

| FREQ (Hz) | RESP (dB) | POWER (W) | SPL (dB) |
|---|---|---|---|
| 20 | -28.84 | 440 | 90.4 |
| 25 | -20.46 | 440 | 98.7 |
| 30 | -13.35 | 193 | 102.3 |
| 35 | -7.38 | 90 | 104.9 |
| 40 | -3.16 | 58 | 107.3 |
| 50 | -0.66 | 80 | 111.1 |
| 60 | -0.63 | 165 | 114.3 |
| 80 | -0.62 | 440 | 118.6 |
| 100 | -0.49 | 440 | 118.7 |
| 120 | -0.37 | 440 | 118.8 |
| 150 | -0.26 | 440 | 118.9 |
| 200 | -0.15 | 440 | 119.0 |

**TABLE 5B**
*Equalized closed box response*
(Right channel)
A = .54, $f_e/f_s$ = 1.40

| FREQ (Hz) | RESP (dB) | POWER (W) | SPL (dB) |
|---|---|---|---|
| 20 | -28.46 | 440 | 90.1 |
| 25 | -20.17 | 440 | 98.4 |
| 30 | -13.18 | 215 | 102.3 |
| 35 | -7.35 | 104 | 104.9 |
| 40 | -3.18 | 68 | 107.3 |
| 50 | -0.52 | 90 | 111.1 |
| 60 | -0.44 | 183 | 114.3 |
| 80 | -0.47 | 440 | 118.1 |
| 100 | -0.38 | 440 | 118.2 |
| 120 | -0.30 | 440 | 118.2 |
| 150 | -0.21 | 440 | 118.3 |
| 200 | -0.12 | 440 | 118.4 |

saw is all you need to complete one channel. This is truly my kind of cabinetry.

Before gluing the pieces together, match the cut boards for best fit, since they may not be exactly the same width or have perfectly square ends. Now, paint all surfaces to be glued with a 50:50 mixture of water and glue. I use Elmer's, but any good wood glue will do. Make sure you paint both surfaces to be joined. This step allows for good spreading of the final gluing to eliminate voids and enhance rigidity. Let stand for about 30 minutes.

Next, lay a "back" piece flat on the floor or work surface and lay a generous bead of glue all around the edges. Position the top, bottom and sides on the back piece. Don't worry about the pieces sliding around, since it takes several minutes before the glue will start to set. Also, don't be afraid to use too much glue. It's cheap and water soluble, so cleanup can be done with a wet cloth. I used a full 12 ounce container of Elmer's on my two enclosures.

Now, clamp the structure together with straps, weights or clamps to maintain pressure on the glued surfaces. Let stand overnight. Note that the front panel is not yet attached. When the glue is dry, glue some 1 × 1″ pine runners around the inside front opening of the box. The screws that will secure the front panel will anchor into these pine runners. The completed box, without front panel, is shown in *Photo 2.*

Some builders will question my avoidance of screws and corner blocks, but remember, I'm not a woodworker. Anything a builder would like to add is his or her choice. I do believe the above construction technique has adequate rigidity, especially in view of the fact that this is an equalized system. The equalizer provides a high-pass function which avoids low frequency pressure build-ups that can flex the cabinet.

*Figure 4* shows the locations of the driver opening and the screw holes for securing the front panel to the box. I used a saber saw to cut the driver openings and a 3/16″ drill for the screw holes. Use a pencil to position the drivers in their cutouts and to locate their mounting holes. I drilled the driver mounting holes with a ¼″ drill so I could use tee-nuts, but ordinary #10 sheet metal or wood screws will do.

Next, drill pilot holes in the pine runners to accept the #10 x 1½″ screws that will hold the front panel in place. Lay the box on its back and then position the front panel in its intended location on the box. Push a screw through each hole in the front panel and tap it with a hammer to identify the location of the pilot hole. Be careful the front panel doesn't slip. Drill the pilot holes in the pine runners with a ⅛″ clear hole.

You can now install electrical connections from the back panel to the woofers. Good quality, low resistance cable and connections are a must, since the drivers will be wired in parallel. For connections, I used 2″ x ¼″ brass screws with brass washers and nuts, although copper is better. The screws are located in the center of the back panel, 1½″ apart and about 10″ down from the top. I used 24″ lengths of #12 AWG stranded wire to connect the drivers to the screw connections. See *Fig. 5* for details.

You can now assemble the kit. First, install the drivers on the front panel, using some kind of gasket to seal any leaks. I used Mortite, but adhesive-backed weather stripping also works well. I applied some ¾″ wide adhesive-backed foam stripping along the edge of the front of the box (visible in *Photo 2*). Solder the leads to the drivers, and use Mortite or caulking to seal the holes in the front panel through which the leads pass to the lower driver. Screw the front panel to the box, repeat the above on the second kit, and move them to your listening room to start breaking in the drivers.

**DRIVER BREAK-IN.** The characteristics of a driver will tend to change during the first 10 to 20 hours of use. It is very important, especially for a vented design, that you make all measurements on drivers that have been used for some period, thereby guaranteeing parameter stability. You can break in drivers by playing a low frequency sine wave for many hours, but

**TABLE 6A**
*Equalized vented box response*
(Left channel)
A = .48, $f_e/f_s$ = 1.1

| FREQ (Hz) | RESP (dB) | POWER (W) | SPL (dB) |
|---|---|---|---|
| 20 | -24.08 | 264 | 92.9 |
| 25 | -12.99 | 342 | 105.1 |
| 30 | -4.79 | 440 | 114.4 |
| 35 | -0.88 | 241 | 115.7 |
| 40 | -0.04 | 143 | 114.3 |
| 50 | 0 | 167 | 115.0 |
| 60 | -0.04 | 256 | 116.8 |
| 80 | -0.04 | 440 | 119.2 |
| 100 | -0.03 | 440 | 119.2 |
| 120 | -0.02 | 440 | 119.2 |
| 150 | -0.01 | 440 | 119.2 |
| 200 | -0.01 | 440 | 119.2 |

**TABLE 6B**
*Equalized vented box response*
(Right channel)
A = .50, $f_e/f_s$ = 1.05

| FREQ (Hz) | RESP (dB) | POWER (W) | SPL (dB) |
|---|---|---|---|
| 20 | -21.93 | 250 | 94.2 |
| 25 | -11.00 | 423 | 107.4 |
| 30 | -3.57 | 440 | 115.0 |
| 35 | -0.64 | 204 | 114.6 |
| 40 | -0.09 | 144 | 113.6 |
| 50 | -0.05 | 182 | 114.7 |
| 60 | -0.06 | 285 | 116.6 |
| 80 | -0.03 | 440 | 118.5 |
| 100 | -0.02 | 440 | 118.5 |
| 120 | -0.01 | 440 | 118.5 |
| 150 | -0.01 | 440 | 118.5 |
| 200 | 0 | 440 | 118.5 |

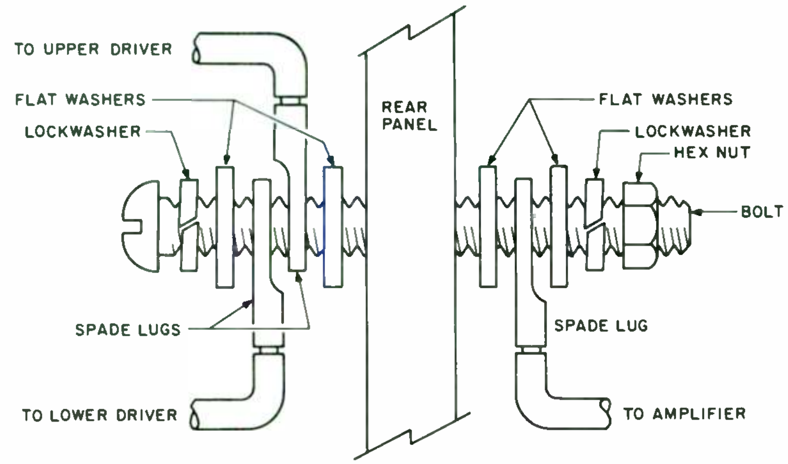


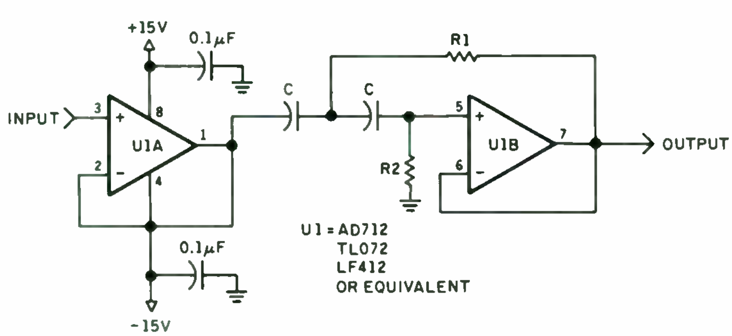

**FIGURE 5: Electrical connection detail.**

this seems like a waste of time when you can use real music and listen to it at the same time. Why is this important? Because the builder who doesn't wish to get involved with the alignment of a vented system may be finished with the project. Let's see why.

I first estimated the system's frequency response in its present form with BOXRESPONSE, using the manufacturer's published data on the drivers (*Table 2*). This gave me an idea of what to expect during the initial listening phase. After listening to the system for about 20 hours, I sensed the design, in its present form, was too "rolled-off" for my tastes. In my homebrew preamp, I do not use tone controls which might compensate for the drooping woofer response. By using a 200Hz crossover point, however, I was most impressed with the coherency of the blend between the woofer and the Strathearns. This was one of my main objectives. In addition, I was also pleased with the dynamic range. Having established the nominal break-in period, I moved on to the measurement phase.

I found the free-air resonant frequencies ($f_s$) of all four drivers were virtually identical. Consequently, I performed all subsequent measurements on the pair of units with their voice coils wired in parallel, thus eliminating the time-consuming and unnecessary task of matching devices. (This applies only to Dynaudio drivers, so if you use less expensive drivers, proceed with caution.)

**SYSTEM I—CLOSED.** *Table 3* shows my free-air and in-box measurement results, and *Table 4* summarizes the BOXRESPONSE data. Note the similarity of the response for this closed-box




| DESIGN EQUATIONS: | SYSTEM | CHANNEL | A | $f_c$(Hz) | C(μf)* | R1(Ω) | R2(Ω) |
|---|---|---|---|---|---|---|---|
| For a given C, | II | Left | .52 | 41.4 | .249 | 4014 | 59.38K |
| $R1 = A/4\pi f_c C$ and π | II | Right | .54 | 41.7 | .249 | 4139 | 56.77K |
| $R2 = 1/\pi A f_c C$ | III | Left | .48 | 32.6 | .249 | 4706 | 81.69K |
| | III | Right | .50 | 31.3 | .249 | 5105 | 81.68K |

*value on hand, actual value not critical.

**FIGURE 6: The equalizer design. Obtain resistor values by using series/parallel combinations of 1% metal-film resistors (available from Digi-Key and others).**

system with the one expected just from using the manufacturer's data. I have dubbed this closed box second-order design "System I." Depending on your needs, this low-Q system may be adequate. Since low frequency response of any woofer is highly room-dependent, an acceptable spectral balance is possible. In addition, this alignment approximates a Bessel or "linear phase" response, which some listeners may find pleasing. When combined with a Dynaudio D28af tweeter with a first-order impedance equalized crossover at 2.5kHz, you should have a very fine system. My objective, however, called for more bass, so I proceeded.

**SYSTEM II—EQ CLOSED.** Before getting down to the vented design, I thought a bit more break-in time was needed, so I invoked BOXRESPONSE once again to see what an equalizer could do to perk up the bottom end. (Note: this is an order 3.5 system.) My results are shown in *Table 5*, and the equalizer schematic is in *Fig. 6*. Although this system's power capacity is compromised, the critical mid-bass region is still capable of excellent SPL. I found this alignment quite good, although lacking in the very deepest bass, especially on rock recordings which seldom go below 60 or 80Hz. The sound was very "tight," with no evidence of distortion.

You may wish to stop here. Further box tuning is not required, unless you want to use BOXRESPONSE to juggle the $f_c/f_s$ and A coefficients for the equalizer. My requirement demanded no ripple, so I was able to get response into the 40s. If higher ripple is acceptable, however, you can obtain a lower cutoff by juggling the equalizer constants until you obtain the desired response.

The equalized closed-box system has many positive attributes. Virtually no complex design is required, just the measurement of the various Qs to establish the response parameters. The use of the equalizer allows for a nearly infinite variety of design tradeoffs.

A full-blown sixth-order system needs an equalizer anyway. Adding it now will allow you to take a breather from the rigors of construction so you can sit back and listen for a while. Who knows, maybe you'll be happy with the results. The suggested addition of the D28af would surely make this a system worthy of respect, but as pleased as I was with the performance of this system, it was time to move on and finish the job I set out to do.

**SYSTEM III—VENTED EQ.** As mentioned earlier, this was my first attempt at a vented alignment. Do not be intimidated. You can perform the necessary tuning in a couple of afternoons or evenings. First, you must have a target alignment. I used the sixth-order Type I QB5 alignment for a $Q_t$ of .293 for the left channel and .305 for the right. Using the data in *Table 3* of Bullock's article, I then calculated the necessary box volume based on the requisite $\alpha$ for the respective driver $V_{as}$. Target alignment data for the left channel is:

$$Q_t = .293$$
$$Q_l = 7$$
$$h = 1.025$$
$$\alpha = 2.689, \text{ and}$$
$$f3 = 33.9\text{Hz}$$

For the right channel,

$$Q_t = .305$$
$$Q_l = 7$$
$$h = 1.008$$
$$\alpha = 2.424, \text{ and}$$
$$f3 = 31.6\text{Hz}$$

Since the $V_{as}$ of my drivers was slightly higher than expected, the required box volume was a bit larger than I had originally calculated, making me glad I had oversized the boxes. I decreased the box volume by adding patio bricks, and calculated the vent length necessary to tune the volume to the necessary frequency, which was about 2⅞" of the 2" diameter PVC pipe I had on hand. Next, I performed the procedure outlined in Mr. Bullock's *SB* 2/81 article, but found that $\alpha$ was somewhat larger than expected. I then removed sufficient bricks to decrease $\alpha$, left the vent tuning alone, and I achieved the following results:

$$Q_l = 6.2$$
$$h = 1.021$$
$$\alpha = 2.677$$

Although I didn't exactly hit my target alignment, I turned to BOXRESPONSE to see whether I could obtain an acceptable response by juggling the equalizer constants. *Table 6A* shows the results, which virtually met my specification. Such is the beauty of BOXRESPONSE. I don't know exactly what kind of response I have achieved, but there's hardly any ripple, it goes low, and it handles the power.

I repeated the above procedure on the right channel, arriving at the alignment:

$$Q_l = 7.4$$
$$h = .9756$$
$$\alpha = 2.452$$

I invoked BOXRESPONSE and arrived at the response detailed in *Table 6B*. See *Fig. 6* for the modified equalizer resistor values.

**CONCLUSIONS.** This is, without a doubt, the finest woofer system I have ever used. I believe all my design goals have been met with a minimum of compromises. The sound is deep, powerful and tight, and blends nicely with the Strathearns. I am using an active third-order Butterworth low-pass filter between the woofer equalizer and the woofer power amplifier, in addition to a passive first-order filter in front of the Strathearn power amp.

I strongly favor using spikes or "tiptoes" for any speaker system, especially a woofer. Deep bass response will be severely compromised should this system rest on a carpeted floor. You must firmly anchor the cabinet on a solid surface, such as a concrete basement floor or a wooden floor. I have obtained satisfactory results by driving three nails in a triangle pattern through a 12"x12" piece of particle board, which serves to hold the nails in place such that the nail points contact the subflooring and the woofer cabinet rests on the nail heads. Sturdy nails are a must: I used 1½" concrete type. This platform is visible in *Photo 2*.

In this article, I have attempted to convey a number of techniques. First, you need only a minimum of cabinetry work to achieve first class sound with excellent drivers. Second, a number of different system alignments are possible with a given driver-box combination, depending only on how much time you want to invest. And third, the power of BOXRESPONSE is indispensable in turning a questionable alignment into a useable one, highlighting the use of an equalizer for either a closed-box or vented system.

Again, do not be afraid to pursue this ambitious design. Measuring a voice-coil impedance curve to within a few percent requires care and patience, but the results are worth it.

One small aside before I close. I tuned up the left channel vented box on a cool afternoon when the tempera-

*Continued on page 64*

# EXPERIMENTS WITH TAPERED PIPES

BY DAVID B. WEEMS

"If you run into trouble, use the line as a dog house and design either a closed box or reflex enclosure for your woofer![1]" For champions of transmission lines who are disturbed by those frivolous words,[2] here is a peace pipe. Or, to be more specific, a tapered pipe. Like the labyrinth, the tapered pipe makes use of quarter-wave loading. Unlike the labyrinth, the driver is placed some distance from the end of the pipe.

Tapered pipes were popular in Britain years ago, but except for a small amount of damping material behind the driver, they were bare-walled. In fact, the designer of one 1960s pipe enclosure said damping material in the pipe would spoil the performance.[3]

The stuffed tapered pipe is an alternative to those designs. Voigt was the first to use the "stopped" pipe, which is similar in action to an organ pipe. In 1949, Ralph West developed the Decca corner speaker. Both systems had relatively small drivers.

**THEORY.** The principle behind the pipe is shown in *Fig. 1*. The sketch in *Fig. 1a* shows relative pressure and velocity in a closed pipe at resonance. Pressure points occur at nodes, high velocity points at loops. If you mount a speaker at the closed end of the pipe (*Fig. 1b*), it will be loaded by the high pressure at that end, greatly increasing efficiency. Two problems hinder this arrangement. First, the pipe's fundamental frequency is so strongly favored, low frequency performance can sound like one-note bass. Another disadvantage is odd harmonics production. The first harmonic is the most serious, occurring at three times the fundamental's frequency (*Fig. 1c*). To correct it, place the driver at one-third the distance from the pipe's stopped end (*Fig. 1d*). At that point, the pressure will become somewhat lower than at the closed end, but still high enough to provide good loading at the fundamental. At the first harmonic frequency, the drive point occurs at a loop instead of a node, and output will be reduced.

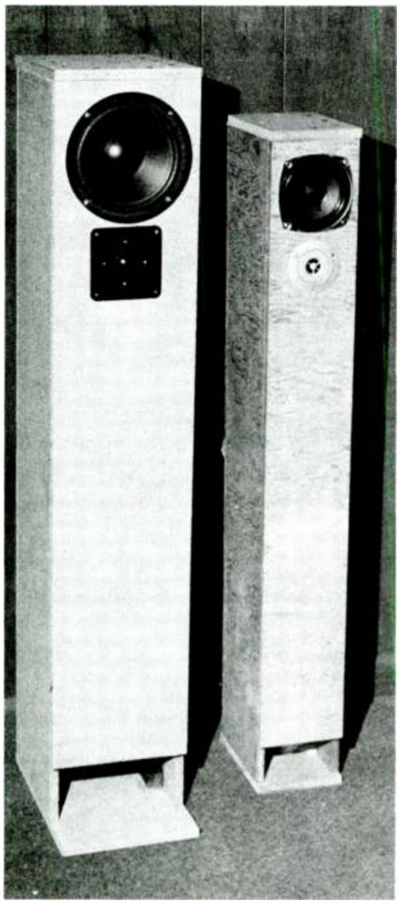


**PHOTO 1: The tapered and damped pipe systems.**

Voigt tapered the pipe to reduce the one-note effect and to spread the resonance over a band of frequencies (*Fig. 1e*). In later versions, the throat area was reduced to zero for smoother response and the driver installed at the midpoint of the line (*Fig. 1f*). More about that later.

The English builders of a generation ago designed their pipes for a single-cone 8″ driver. Total cost of a driver and enclosure for one popular model was about £5, or at the rate of exchange in those days, about $14. The speaker fired upward at the rear wall so the highs dispersed around the room at a subdued level. In an attempt to restore the lost overtones, listeners applied treble boost. Some thought the arrangement's high frequency reproduction was inadequate. One asked, "Who put the blanket over the cymbals?" The problem of highs lost in the reflection process was compounded by the voice coil inductance in the single-cone speaker.

British builders almost always used ⅜″ plywood to construct their pipes. Thick walls were unnecessary, they said, because their enclosures were stiffest at the point of greatest pressure. Some suggested using a slight degree of wall flexure because it would increase acoustic coupling near the fundamental frequency and damp air column resonances. Such statements, however, made pipe design seem more like magic or luck than science. Everyone seemed to agree on the necessity for tight joints. One author said a 1/16″ gap in the driver mounting would reduce the output at 35Hz by a factor of four.[4]

About five years ago, after having ignored pipes because of their resemblance to the infamous "air coupler" of early US hi-fi days, I built a pipe for a dual cone 6″ x 9″ car driver featuring a 10 oz. magnet and foam suspension. The $7 speaker had an amazing bass re-

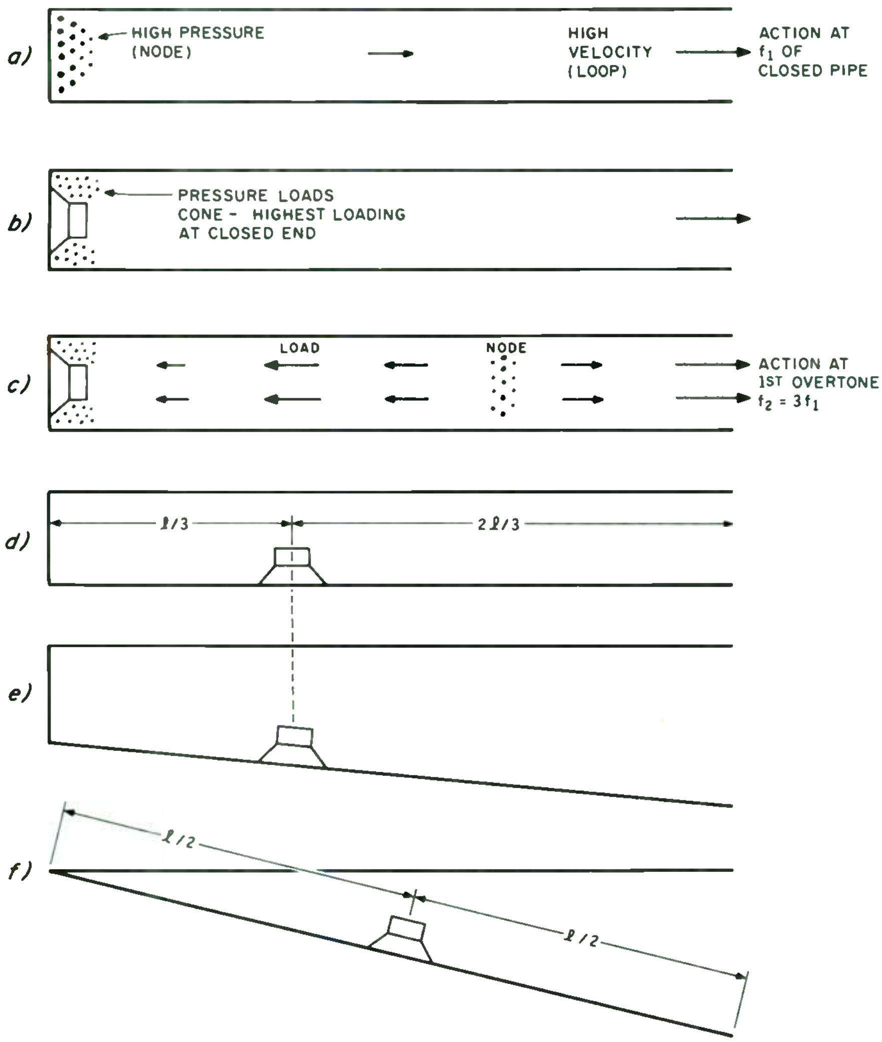



**FIGURE 1: Various pipe systems. See text.**

sponse in the pipe, but it also had some obvious peaks, the worst of which occurred at 95Hz and 220Hz. The 220Hz peak was probably caused by the second harmonic which arises at five times the pipe's fundamental frequency. A driver with a heavier magnet gave much smoother response in the 95Hz region, but showed no improvement in the higher resonance. I considered attacking the 220Hz problem with a Helmholtz resonator designed to absorb energy at that frequency, but other endeavors interrupted my plan.

Recently, while digging through abandoned projects in my shop, I discovered the old pipes and wondered how the principle would work with better drivers and with damping material in the line. To save time and material, I decided to make a miniature pipe for a 4″ woofer, the Radio Shack #40-1022. I knew I could later apply my experience to larger systems, but first, I had to design the enclosure.

**PIPE DESIGN.** To design a tapered pipe system, you must choose appropriate dimensions for pipe length, throat and mouth area, and drive point. The fundamental frequency is almost totally set by one parameter—the total line length. The formula given for a labyrinth or pipe length is:

$$l = C/4f - 1.7r$$

where l is the length of pipe, C is the speed of sound in air, f is the pipe's fundamental frequency, and r is the pipe's radius or equivalent radius.

This formula, however, produces a longer than necessary pipe length. Opposing factors are at work here because tapering the pipe tends to raise the fundamental frequency, but folding and choking the mouth at the port lowers it. A bare pipe will usually perform as though it were about 30% longer than its measured length, and when you add damping material, it can seem even longer. As a rough estimate, expect the required length to be about 65% of the calculated length.

The pipe's cross-sectional area will vary from near zero to zero, to a maximum of about 2.5 times that of the driver's effective piston area. The pipe area should be about equal to the cone's area at the drive point, but ±20% is acceptable. The port area is typically equal to the cone area, or about 0.4 that of the maximum area.

Taper rate is one aspect of the tapered pipe that varies with driver size. A 4″ woofer requires less than a third of the pipe area needed by a 9″ driver. If a pipe for a 4″ woofer was designed to have the same taper rate as that for the larger speaker, its total length would be 20″. Even if it "acted" like a 30″ pipe, the fundamental frequency would be about 100Hz. That is too high, even for a 4″ woofer. You can neglect taper rate as a design factor for small drivers, but a low taper rate may demand more stuffing. If

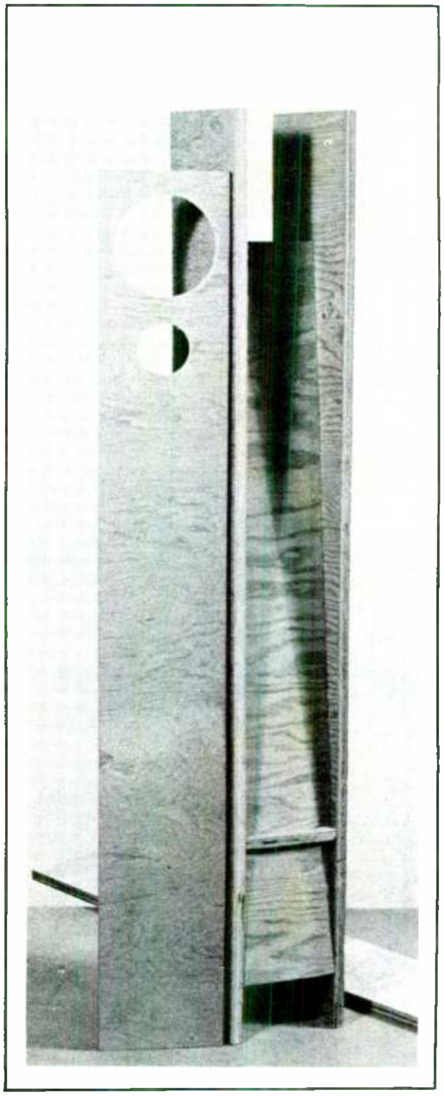


**PHOTO 2: Internal construction of a tapered pipe.**

you wish to experiment with bare pipes, use an 8″ driver.

I made a quick guess at the proper length by setting the enclosure height at 3′, which placed the drivers close to ear level. Unlike the British designs, I decided to mount the drivers in the conventional way, on the enclosure's front. To complement the Radio Shack woofer, I used a Radio Shack ¾″ tweeter (#40-1376). The crossover for this simple system is nothing more than the high-pass capacitor that comes with the tweeter. An L-pad adjusted the tweeter level to match the woofer's rather inefficient level. Unlike conventional systems, however, I placed the tweeter below the woofer, a move which made more of an effect that I expected.

After drawing a rough design sketch, I noticed the drive point would be somewhat short of the mid-point from the throat to the mouth. You can only estimate the pipe length; the true acoustical length may be different from that estimate. The formula for optimum drive point distance from the throat is:

$$d = 1/(2 + \sqrt{A_t/A_m})$$

where d is the drive point distance from the throat, 1 is the pipe length, $A_t$ is the throat's cross-sectional area, and $A_m$ is the mouth's cross-sectional area.

The mouth is defined as the port area. The maximum area is that of the section just before the port. As you can see from the above formula, d can vary from one-third where the area of the throat is equal to that of the mouth, to one-half where the pipe tapers to zero throat area. Again, unless you use a straight pipe, you can only estimate the drive point's optimum location.

After doing some arithmetic, I found that by starting the throat at a point 6″ above the mouth and using an initial area of 2″², the pipe would approximate the formula for the drive point. The maximum area would be 24″², which would be choked to 9.25″² at the mouth (*Fig. 2*).

My final decision concerned the building material's type and thickness. For such a small pipe, ¼″ plywood seemed adequate. But, because I had a supply of ½″ material and because using ¼″ plywood seemed like heresy, I opted for the thicker walls. If you wish to try a pipe system without damping material, try a thinner material.

Here is a quick summary of my construction methods, should you choose to follow them:

1. Cut out the parts and prepare to attach the partition to the sides with glue and nails.
2. To place the partition precisely, first set it in place on the inner surface of each side and mark an outline around it. Then, drill guide holes for nails down the center line of that outline.
3. After reversing the side, drive a nail into the partition through a hole at each end of the line of holes, just far enough to temporarily hold the partition.
4. Remove the side and spread glue on the matching surfaces. When you do the final nailing, the pre-nailed holes will ensure the parts don't slip out of place as you drive down the other nails.
5. After the glue sets, caulk the joints and install the front and back panels with silicone rubber sealer and nails. The sealer prevents air leaks on the joints you can't reach during internal caulking.
6. To make the top and bottom removable, install cleats inside the pipe to receive screws.
7. Gasket any removable parts with foam weather stripping or other sealer.

*Continued on page 22*

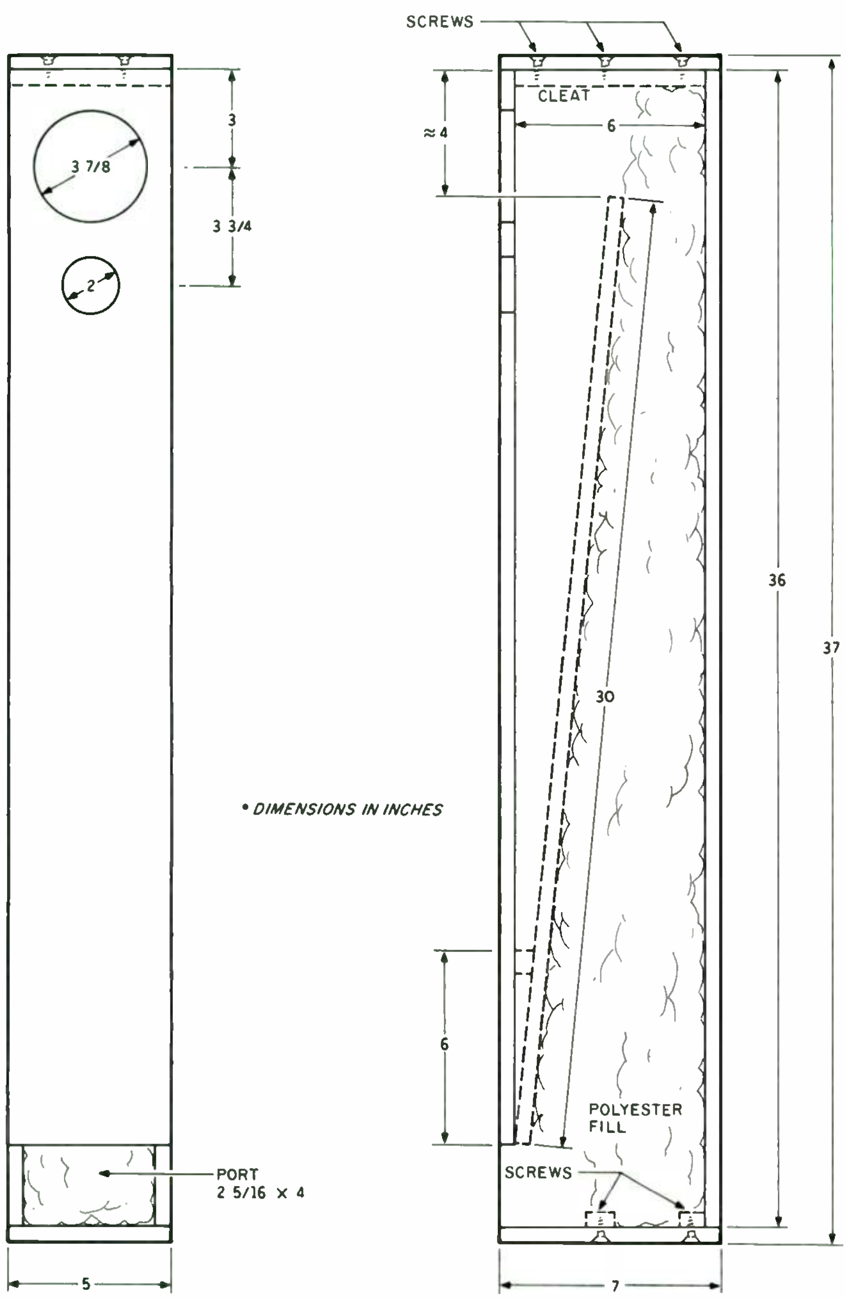



**FIGURE 2: Dimensions of tapered pipe system for 4″ woofer.**

# SLEDGEHAMMER

*The SLEDGEHAMMER 15258DVC & 15254DVC are 15" polypropylene woofers, black cone, 2.5" Kapton dual voice coils, 60oz. ceramic magnet.*

| Model | 15258DVC | 15254DVC |
|---|---|---|
| Fs | 20Hz | 19Hz |
| Mmd | 170 Grams | 160 Grams |
| Cms | .41 x 10-6cm/dy | .48cm/dy |
| Vas | 360lt. ±2% | 360lt. ±2% |
| Rscc | 5.7Ω | 3.5Ω |
| Z min | 6.3Ω | 3.8Ω |
| Z max | 58.0Ω | 43.0Ω |
| vcL | 1.2mH | .72mH |
| Qms | 4.69 | 4.79 |
| Qes | .53 | .41 |
| Qts | .47 | .38 |
| Xmax | 5.5mm Pk | 5.5mm |

Surround: Foam
Power Handling: 200W 100/100
Frequency Response: 30–1500Hz
Efficiency: 90.4dB/1W/1M
Uses: Home or Autosound subwoofer
Price: $58.00

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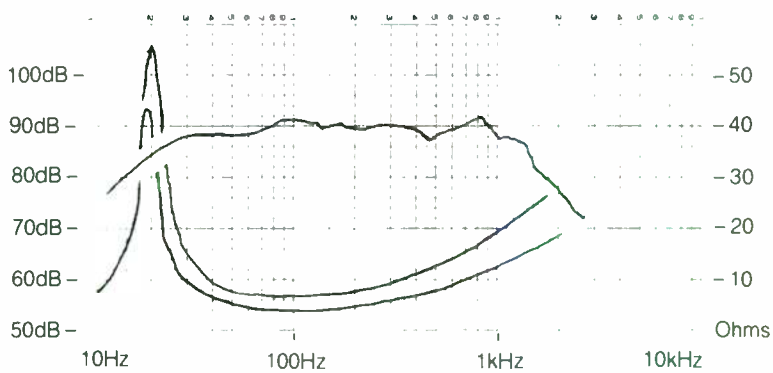


**SLEDGEHAMMER 15258DVC SUGGESTED ALIGNMENTS (ALL SEALED)**

| Bass Volume: VB | 35 liters | 56 liters | 70 liters | 70 liters | 100 liters |
|---|---|---|---|---|---|
| Bass ½ Power: F3 | 47Hz | 44Hz | 42Hz | 40Hz | 38Hz |
| Fill in box | Yes | Yes | Yes | No | No |
| QTC | .86 | .80 | .74 | .86 | .75 |
| Peak at Res: R dB | +1.0 | +.3 | +.1 | +.5 | 0 |

**SLEDGEHAMMER 15254DVC SUGGESTED ALIGNMENTS (ALL SEALED)**

| Bass Volume: | 56 liters | 70 liters | 100 liters | 100 liters | 150 liters |
|---|---|---|---|---|---|
| Bass ½ Power: F3 | 40Hz | 38Hz | 37Hz | 35Hz | 34Hz |
| Fill in box | Yes | Yes | Yes | No | Yes |
| QTC | .95 | .88 | .79 | .91 | .70 |
| Peak at Res: R dB | +1.0 | +.5 | +.1 | +.3 | 0 |

SledgeHammer 15DVCs can be connected in series or parallel to achieve net impedance from 2–16Ω.

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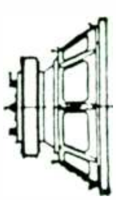

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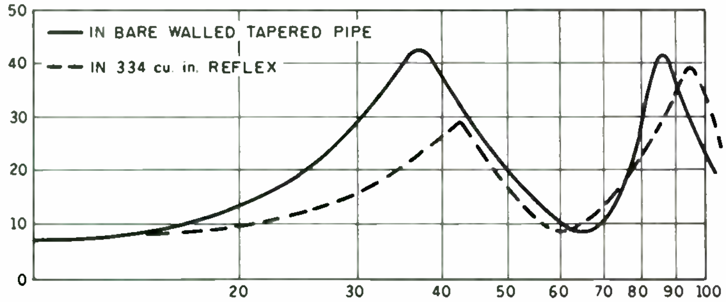


**FIGURE 3: Impedance curves of 4" woofer in bare tapered pipe and in reflex.**

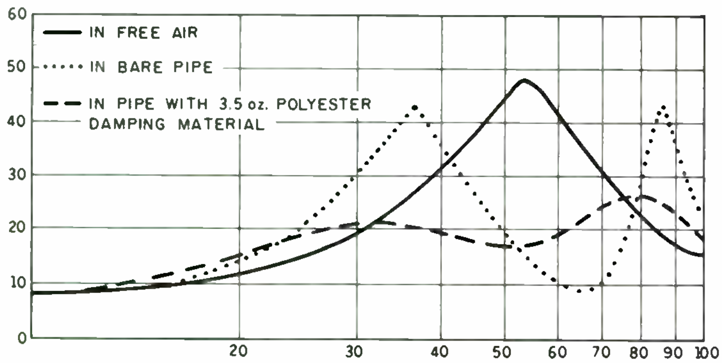


**FIGURE 4: Impedance curves of 4" woofer in free air and in pipes.**

*Continued from page 20*

**TEST RESULTS.** In *Fig. 3*, the impedance curve of the 4" woofer in a bare pipe looks similar to the same model woofer in a reflex. As I discovered later, the woofers behaved significantly differently in some tests. Some writers call the impedance peaks in tapered pipes "resonances," but Baldock claims the term resonance isn't appropriate because a reactive component is present[5] When I ran an oscilloscope test on bare pipes, the ellipse closed to a straight line at the peaks and valleys of the impedance curves. After I added damping material, the ellipses remained open. The lower peak was likely produced by cone loading from the air in the pipe, while the upper one was likely caused by enclosed air stiffness behind the cone.

If you analyze the curves for the woofer in free air and in the pipe, you'll see the lower peak in the pipe occurs at about 37.5Hz, whereas in free air, the single peak is at 53Hz or 1.41 times that of the pipe. This ratio tells you the air load mass in the pipe is approximately equal to the mass of the cone itself.

On listening tests, the bare pipe produced a hollow, ringing sound, particularly on male voices. When I conducted a rough frequency test, significant peaks and valleys occurred in the region below 500Hz. So, I lined the pipe area behind the woofer with a 1" layer of acoustical fiberglass on the sides, back, and under the top of the enclosure. I then noticed some improvement in the listening tests, and a considerable difference in the 200–500Hz frequency response test.

I added loose wads of polyester batting to the pipe, placing ¼ oz. in the throat, filling the space behind the woofer, and then gradually increasing the amount in the large section of the pipe. With each addition, I monitored the progress with impedance and listening tests.

You can stuff the pipe until the impedance curve shows a single peak, like that of a closed box speaker. The peak frequency, however, will be lower in the pipe than in a closed box. In my case, the single peak occurred at 60Hz. After stuffing and restuffing, I arrived at what seemed to be a good compromise between under and overstuffing: a total of 3½ oz. of batting. My pipe's internal volume is 0.5 $ft^3$, not including the volume occupied by drivers, partitions and cleats. Therefore, the rate of fill is probably about 8 oz. per cubic foot. The impedance curve for that condition is shown in *Fig. 4*.

After the final stuffing, I compared the pipe speaker performance with drivers of the same model in a closed box and a reflex enclosure. The closed box was clearly superior to the other two in one respect. With a total volume of only 140 $in^3$, it occupied the least amount of room space. The reflex had a volume of 334 $in^3$ and was tuned to 60Hz.

I then conducted nearfield microphone tests, which showed the three systems performed similarly in the 80–100Hz range. Listening trials, however, told me a different story about bass range perception. The closed box speaker was clearly more limited in low frequency response. (Nearfield technique doesn't register the output of ports removed some distance from the woofer.)

When listeners compared the reflex with the pipe, most initially thought the reflex had a bit more bass than the pipe. After longer and more careful listening, however, their conviction waned because the pipe speaker performed better on the lowest frequencies. Frequen-

*Continued on page 24*

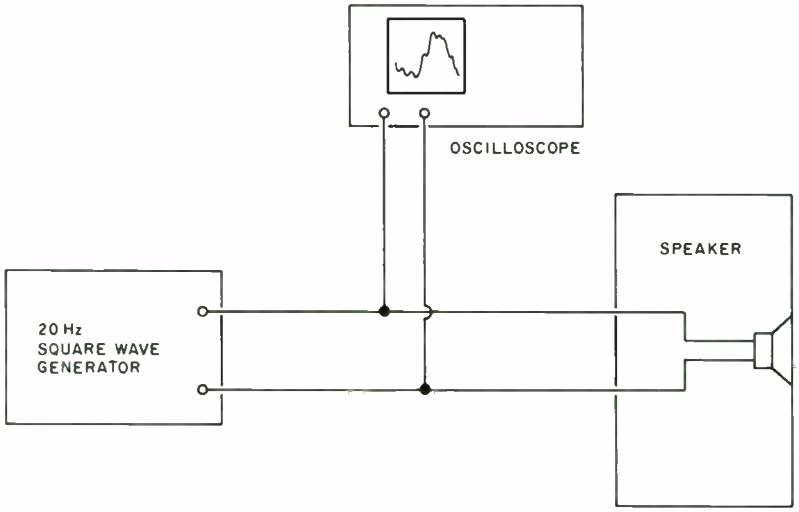


**FIGURE 5: Test setup to measure hangover.**

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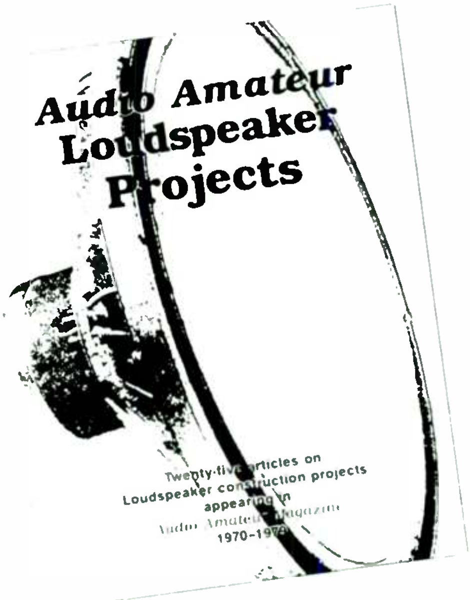


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*Continued from page 22*
cy response tests confirmed this conclusion. The reflex had slightly more output in the 70–100Hz range, while the pipe was clearly superior below 60Hz.

An interesting way to compare speaker performance is to feed a 20Hz square wave to the speakers and observe the patterns produced on an oscilloscope. This simple test setup is shown in *Fig. 5*.

The pattern will alter according to the resonance frequency of your speaker and its degree of damping. The lower the resonance frequency, the fewer the number of oscillations that will show on the trace. The oscillations' amplitude, however, will also vary. I used this method to test four identical woofer models: one in free air, then again in free air with a blob of modeling clay (whose mass was equal to the cone's mass) stuck to the cone; and three other woofers in the three types of enclosures I mentioned earlier.

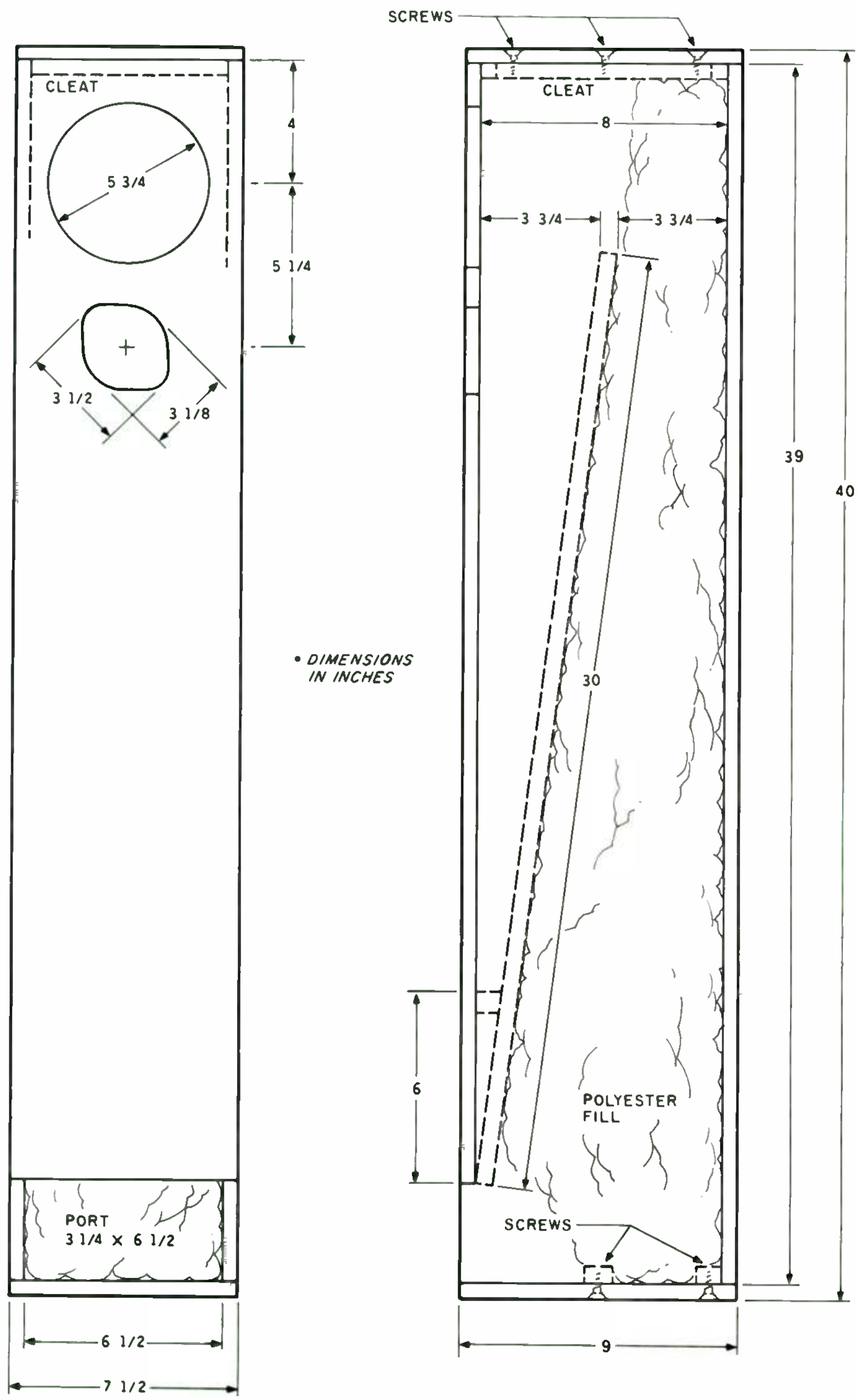


**FIGURE 6: Dimensions of tapered pipe for 6½″ woofer.**

Before you examine my various pattern photographs, be aware my oscilloscope exhibits considerable droop when it is fed a 20Hz square wave. That droop may be caused by a too small coupling capacitor, but whatever the reason, it is constant. If your instrument's controls are unchanged during the tests of various speakers, consider the comparisons valid.

The pattern in *Photos 3a* and *3b* indicate, as expected, the driver cone moves farther in free air than when installed in an enclosure. Little change occurred in the two free air tests, except for the lower resonance frequency in the one I conducted after I added mass to the cone. The trace for the closed box woofer (*Photo 4a*) shows decreasing oscillation amplitude, but the reflex's trace (*Photo 4b*) indicates little amplitude change. The pipe speaker (*Photo 4c*) shows a single pulse movement, about half a sine wave, then very little hangover. I took *Photo 4c* after I added damping material to the pipe, but the woofer in the bare pipe's trace was almost identical except for a small deflection at tail's end.

To test the woofers in the three kinds of enclosures, I fed each a 55Hz sine wave signal from an audio generator through a tone burst generator and an amplifier (*Photos 5a*–5c). In fairness to the reflex speaker, it had no more overshoot than the pipe speaker at frequencies above 80Hz, and only slightly more at 70Hz. The pipe speaker was at least equal to the others at all frequencies, except at 180Hz where it showed a bit more hangover.

After reviewing the experiments with the 4″ woofer, I decided the tapered pipe system deserved further testing with a larger woofer. With that in mind, I began to work on a pipe designed around the Peerless TP165F.

**SECOND DESIGN.** Peerless lists the TP165F's effective piston area at 12.45 x $10^{-3}$ square meters, or about 20 in.² I made the internal dimensions 6½″ x 8″, putting the pipe's theoretical maximum area at 52 in.² By using ½″ material, the minimum outside dimensions are 7½″ x 9.″ Because pipes are so compact, they occupy very little floor space and yet do not require stands. On most floor surfaces, however, you may wish to enlarge the bottom board to add stability to the tower. Extend the board beyond the enclosure's front to counter gravity's pull on the front-mounted drivers.

For easy pipe construction, place the tweeter below the woofer, as shown in *Photo 1*. Ironically, a tall enclosure, with the tweeter at ear level, should have the

*Continued on page 26*

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**CTC 300: PHONOGRAPH CARTRIDGE TEST RECORD.** Used for measuring the frequency response, crosstalk, low frequency resonance, polarity, compliance, and tracking ability of phonograph cartridges. It contains swept frequency left and right channel test bands, in both audio and infrasonic ranges, whose logarithmic frequency sweeps are compatible with the chart speeds of many graphic level recorders.

**CONTENTS: Side A.** Bands 1 Left channel sweep, 2 Right frequency sweep, 3 Left separation, 4 Right separation, 5 Lateral polarity test, 6. Vertical polarity test, 7. Left 1kHz tone, 8. Right 1kHz tone

**SIDE B.** Bands 1 Lateral sweep tone arm resonance, 2 Vertical sweep: tone arm resonance, 3 Lateral Compliance 100Hz, 4 Vertical compliance 100Hz, 5. Lateral tracking: 300Hz, 6 Vertical tracking: 300Hz.

**CTC 310: DISTORTION TEST RECORD.** Designed for measuring the non-linear distortion of phonograph pickup cartridges. Distortions are often the result of a non-linear relationship between the stylus velocity (or displacement) and the electrical output voltage from the cartridge. Part of this non-linearity may originate in the voltage generating system of the cartridge, and in part from the coupling between the stylus and the voltage generator which may be magnetic, piezo-electric, or some other element. Other distortions are fundamental to the the shape of the cutting and playback stylii, their effective vertical tracking angles, or the coupling of the stylus to the record groove. The total distortion usually increases with recorded level throughout the normal operating range of the cartridge, becoming very large as the mechanical limits of the cartridge components are approached or exceeded.

**CONTENTS: Side A.** Bands 1 Left Two-tone sweep: F2 = 1.0-50kHz, F1 = 0.685 to 49.685; (F2-F1 = 315Hz, 2 Left: 1kHz Square wave; 3 Right: 1kHz Square wave, 4 Left spots 20kHz-20Hz; 5 Intermodulation 4kHz + 400Hz in ascending level steps, phase locked, 6. Right, Intermodulation 4kHz + 400Hz in Ascending level steps, phase locked; 7 Left 1kHz reference tone, 8. Right 1kHz reference tone

**SIDE B.** Bands 1 Right, Two-tone sweep F2 = 1kHz to 50kHz, F1 = 0.685kHz to 49.685kHz(F2-F1 = 315Hz); 2 Lateral, 1kHz square wave; 3. Vertical, 1kHz square wave, 4 Right spots 20kHz—20Hz; 5 Right spots 20kHz-20Hz, 5 Lateral, Intermodulation 4kHz + 400Hz in ascending level steps, phase locked; 6. Vertical, intermodulation 4kHz + 400Hz in ascending level steps, phase locked, 7 Lateral, Noise band: 16kHz

**CTC 330: STUDIO TEST RECORD.** This was developed to assist broadcast engineers and technicians, advanced audiophiles, and other professionals in evaluating and adjusting the performance of audio disk playback equipment. It provides the range of test frequencies and levels required to measure phono cartridge sensitivity, frequency response and stereo channel separation as well as the sensitivity and frequency response of monophonic cartridges. Standard level signals can be used to set the gain in other parts of the reproducing system and to identify the left and right stereo channels. Other test bands are used to check system phase and measure or adjust the rotational speed of a turntable.

**CONTENTS: Side A.** Bands 1A Frequency response measurement (l. chan), 2A Frequency response measurement (r. chan.), 3A Lateral frequency sweep, 4A Spot frequencies (l. chan.), 5A High level sweep (l. chan.), 6A High level sweep (r. chan.), 7A Left channel reference tone, 8A Right channel reference tone

**SIDE B.** Bands: 1B. Lateral spot frequencies, 2B Right channel spot frequencies, 3B Lateral reference tone; 4B Vertical reference tone, 5A (B) Stroboscope (120Hz); 5B (B) Stroboscope (100Hz)

**CTC 340: ACOUSTICAL TEST RECORD.** Intended to be used for measuring the performance of an entire reproducing system, including the loudspeakers. The program uses random noise suitable for measurement with instruments or, in some cases, interpretation by ear. The signals are characterized as "pink" noise, which is random in nature and has equal average energy in each octave of the audio frequency range, from 20Hz to 20kHz. A spectrum analysis of pink noise using a proportional bandwidth analyzer (such as the "real time" spectrum analyzers often used for acoustical measurements) produces a linear (or "flat") frequency response. In contrast, a constant bandwidth analysis of pink noise yields a response that decreases at 3dB per octave with increasing frequency.

**CONTENTS: Side A.** Bands 1 Left sweep ⅓-octave pink noise, 20Hz—20kHz, 2 Right sweep ⅓-octave pink noise, 20Hz—20kHz, 3 Lateral sweep ⅓-octave pink noise, 20Hz—20kHz, 4 Left spots: ⅓-octave pink noise, 20Hz—20kHz; 5 Left channel wide band pink noise, 20Hz—20kHz, 6 Right channel: wide band pink noise, 20Hz—20kHz, 7 Left 1kHz Reference tone, 8 Right 1kHz Reference tone.

**SIDE B.** Bands 1 Lateral spots ⅓ octave pink noise, 20Hz—20kHz, 2 Right spots ⅓ octave pink noise, 20Hz—20kHz; 3 Left and right wide band noise (in phase), 20Hz—20kHz 4 Left and right wide band random noise (random phase), 5 Lateral 1kHz Reference tone

**CTC 350: TURNTABLE AND TONE ARM TEST RECORD.** The three basic components of a phonograph record player are the turntable, which rotates the record at (ideally) a constant angular speed, the tone arm, which supports the cartridge in a correct geometrical relationship to the record groove, and to the cartridge, whose stylus traces the record groove and which generates voltage analogs of each groove wall modulation. Because the components of a record player system interact to some degree, it is difficult to measure or specify the performance of one without considering the properties of the other two. In most cases it is acceptable to specify the makes and model numbers of related parts of the system (e.g. the cartridge or tone arm), and their operating conditions where applicable, when measuring the performance of another part.

**CONTENTS: Side A.** Bands 1 Left 1kHz reference tone, 2 Right 1kHz reference tone, 3 Wow and flutter, 4. Lateral sweep 2—100Hz, 5 Vertical sweep 2—100Hz

**SIDE B.** Bands 1 Rumble Reference tone, 2 Rumble Quiet grooves, 3 Lateral 1kHz Reference tone, 4 Vertical 1kHz Reference tone, 5A Vertical strobe 120Hz, 5B Vertical strobe 100Hz

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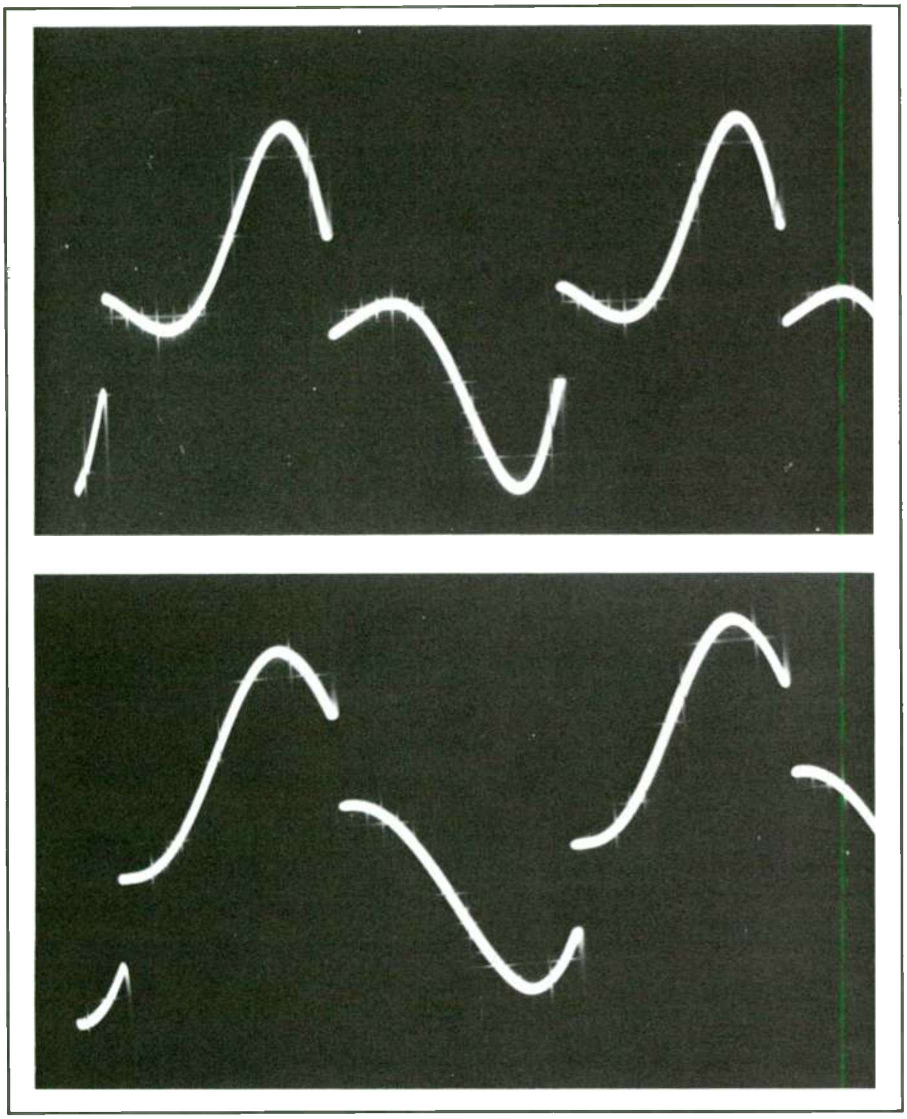

**PHOTO 3: Square wave test on a 4″ woofer in free air (a), and with mass added to cone (b).**

*Continued from page 24*

tweeter above the woofer. When I planned my design, I overlooked one effect of placing the tweeter below the woofer. Although, as you will see, I later solved the problem to my satisfaction, consider placing the tweeter outside the pipe, on top of the tower. This decision will depend on your personal preferences as you weigh performance against appearance and construction ease.

I increased the larger pipe's length in an attempt to extend the bass range. Except for loading the driver to a lower frequency, however, I doubt whether it made much improvement. Again, I had to do some arithmetic to estimate the optimum drive point. My figures indicated I should begin the throat about 6″ above the mouth, with an initial area of slightly under 5 in.$^2$ You can do this by gluing a ¾″ strip across the front of the partition, at the point where it will make contact with the rear of the front panel.

I ran the usual tests with various amounts of damping material and settled on a lighter optimum packing density than I had used on the smaller pipe. The final amount was a square foot of 1″ fiberglass on the surfaces behind the woofer, an ounce of polyester batting in the throat, and 5 oz. in the remainder of the pipe. Although the density was lower than for the smaller pipe, the impedance curve was similar (*Fig. 7*).

The larger system's bass performance was promising from the start, but something about the sound wasn't right. At first, I was tempted to blame the tweeter, a German-made MBM25S I obtained from McGee Radio, but later tests proved me wrong. As it turned out, I had several problems, some of which involved the crossover network.

**EVOLUTION OF A CROSSOVER.** My first crossover network choice was a second-order APC (all-pass crossover) described by Bullock.[6] My enclosure design had placed the woofer-tweeter center-to-center distance at 5¼″, so I planned to put the crossover frequency below the point where that distance was equal to a wavelength, or under 2.6kHz. With that in mind, I chose a crossover frequency of 2.2kHz.

At first, the system sounded loud—an ominous symptom. The problem turned out to be a rise in output between 800Hz–1.8kHz, so I made a filter to smooth the woofer's response in that range. The filter transformed the sound, helping the speaker to wear well in long listening sessions. But one abnormality remained: the vertical response was not uniform when I stood in the sound field and then sat down. Even people who liked the anomaly mentioned it. "It sounds sweeter when you sit down," one person remarked.

When I checked the frequency response at various elevations, I found a null in the 2.6kHz region at ear level while listening from across the room. I realized that when I placed the tweeter below the woofer, it tipped the axis upward (*Fig. 8*). At standing ear level, the null was replaced by a smooth response. Surely, the crossover network was in-

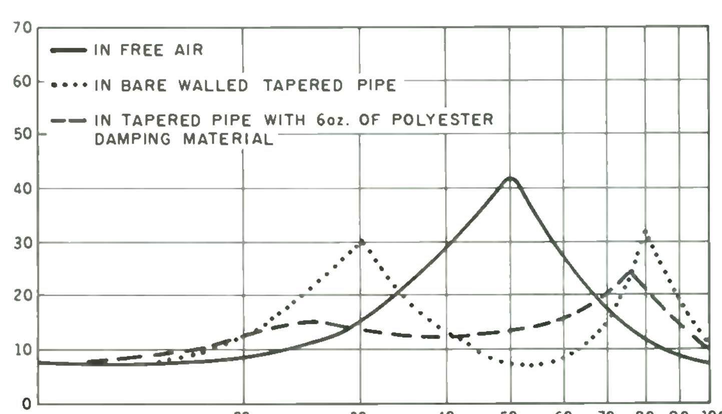


**FIGURE 7: Impedance curves of 6½″ woofer (Peerless TP165F) in free air and in tapered pipes.**

volved, but the problem was caused by a combination of factors that made the woofer's realized rolloff rate different than that predicted by the second-order filter.

I measured each driver's response, substituting a resistor in the crossover network for the muted driver. In the 2–3kHz band, the woofer's and tweeter's output were equal.

I had several solutions to choose from. An obvious one was to use a higher order crossover. Having already made up a set of crossover networks, I tried to salvage them with notch filters, tuned to 3kHz, in the woofer circuit. The first filter, made with a 0.25mH coil and an 11μF capacitor, did not provide enough cut. So, I made up another set, using 1mH coils and 3μF capacitors. They solved the problem, but I wondered whether a simpler method could give me an acceptable performance.

Throughout the tests, I tried nearly every option, including reversed driver polarities. Finally, a single, larger-than-normal inductance in the woofer circuit offered promise. So, I reduced the crossover frequency to 2kHz and made up a third-order filter for the tweeter. When you consider the woofer's impedance with the Zobel in its circuit is about 6Ω, the calculated value for a single inductance is about 0.48mH. Instead of deriving the value from calculations, I set up the speaker outdoors on a quiet day and ran microphone tests with various values of inductance. The one that produced the flattest curve turned out to be 1mH, more than twice the indicated value. This, however, is not a rare conclusion.

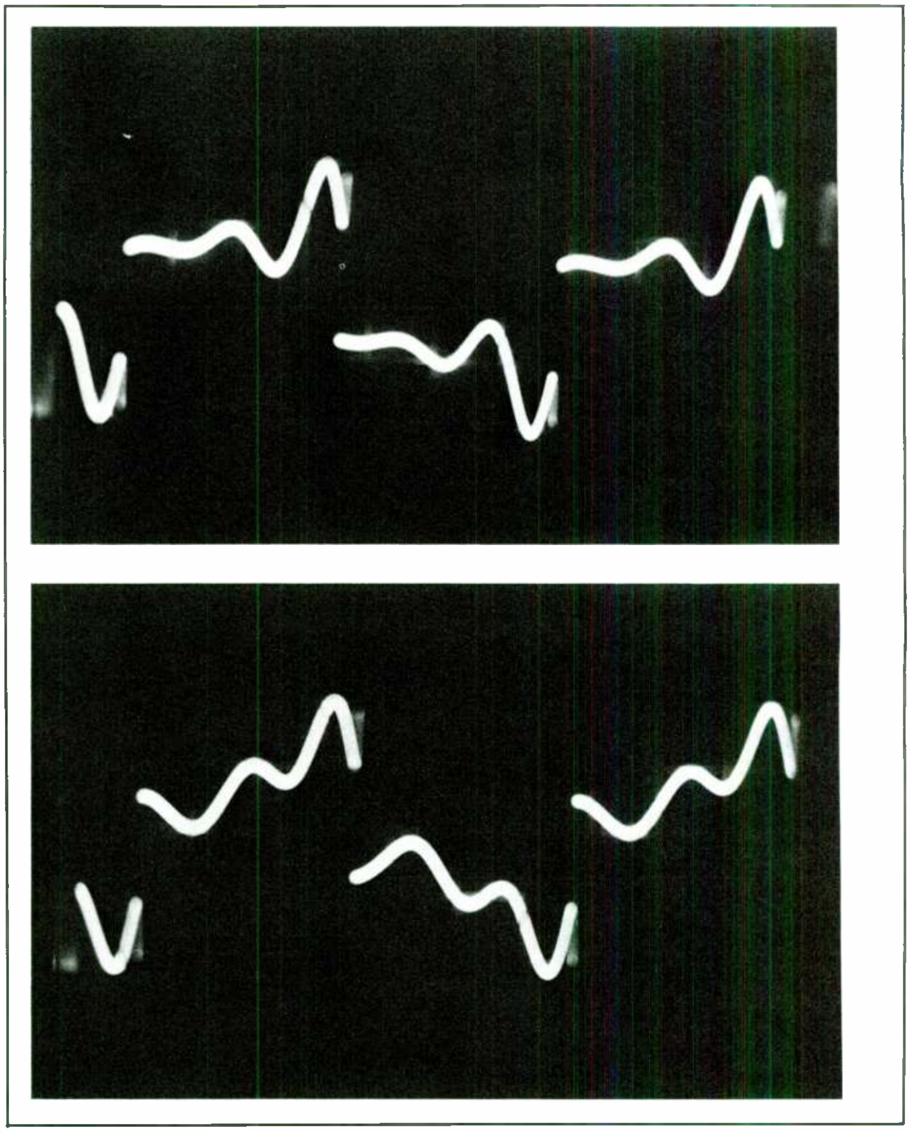

**PHOTO 4: Square wave test on 4″ woofer in closed box (a), in reflex (b), and in tapered pipe (c).**

**RESONANT PEAK FILTERS.** The MBM25S gave a good performance, but I found it had one peculiar trait. Even with a third-order network, a considerable output from the tweeter occurred at 1.1kHz—its resonance frequency. The obvious fix seemed to be a resonant peak filter.

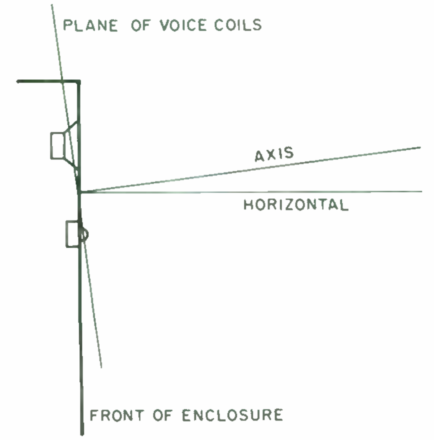


**FIGURE 8: When the tweeter is installed below the woofer, as shown in the plans, the axis tips upward.**

The *Loudspeaker Design Cookbook*[7] (LDC) lists the following formula for resonant peak filters:

$C = 1/15.2\ f$
$L = 1/39.48\ f^2C$
R = approximate rated driver impedance.

Filters produced by this formula are effective in removing the peak, but they can also effect a lower than desired impedance in the octave above resonance. In those cases, you need a higher Q. A formula that usually works well is:

$$C = 1/50f$$

You can compute the other values from the *LDC* formulas. If possible, choose the final values by experimenting. I used the formula above to make up the RPFs (resonant peak filters) suggested in the crossover diagram (*Fig. 9a*). You will need a different RPF (*Fig. 9b*) for the MCD25M, the more expensive titanium tweeter made by the MBM25S's German manufacturer. For either tweeter model, the combined impedance of the RPF, L-pad and tweeter is about 6Ω, so the same crossover values will work on both.

While the wiring diagram shows reversed polarities for the drivers, putting the drivers in phase will make for a slightly more uniform response at crossover. Unfortunately, it also raises the level a bit at 1.8kHz. Going back to the reversed condition will shift the boost upward on the axis, leaving a slight dip at crossover at ear level when you're listening while sitting. Unlike the earlier problem, however, the dip is too shallow to be noticed by most listeners. Another

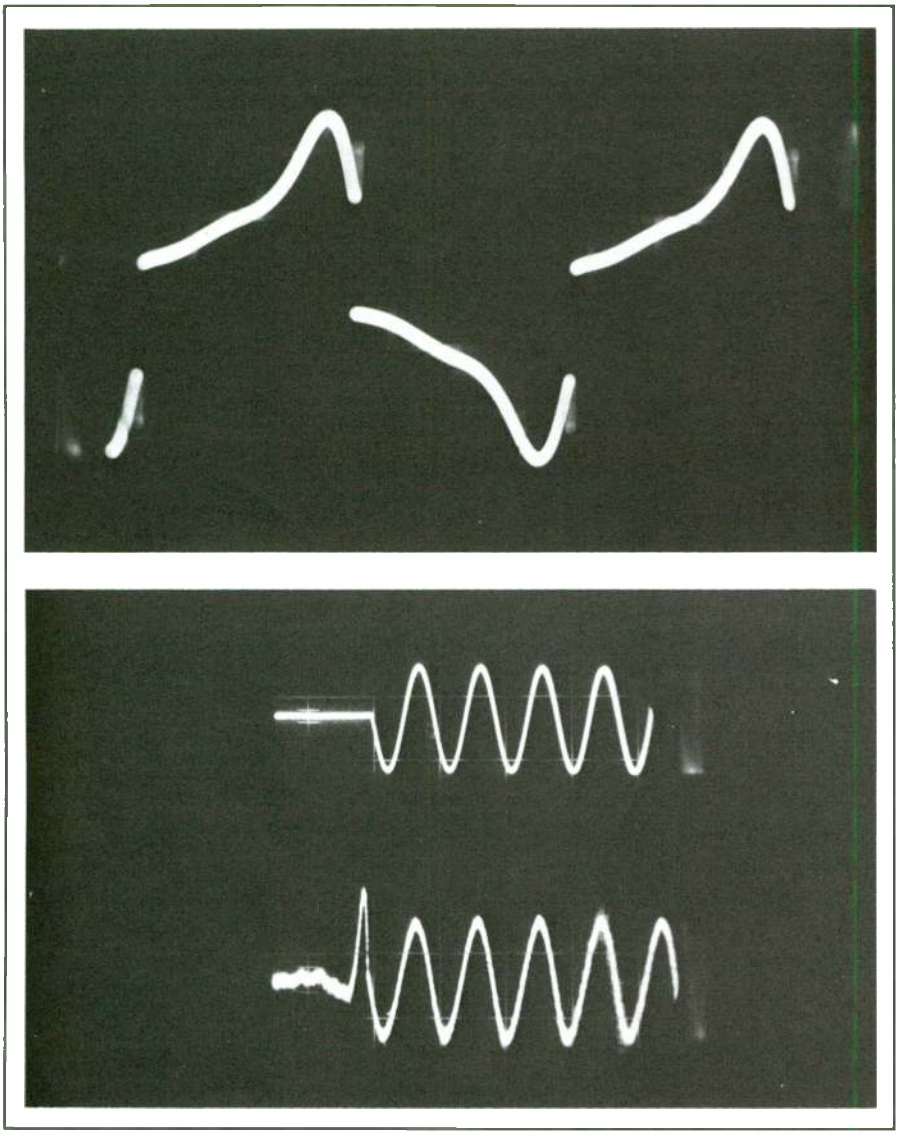

**PHOTO 5: Tone burst test on 4″ woofer in closed box (a), in reflex (b), and in tapered pipe (c).**

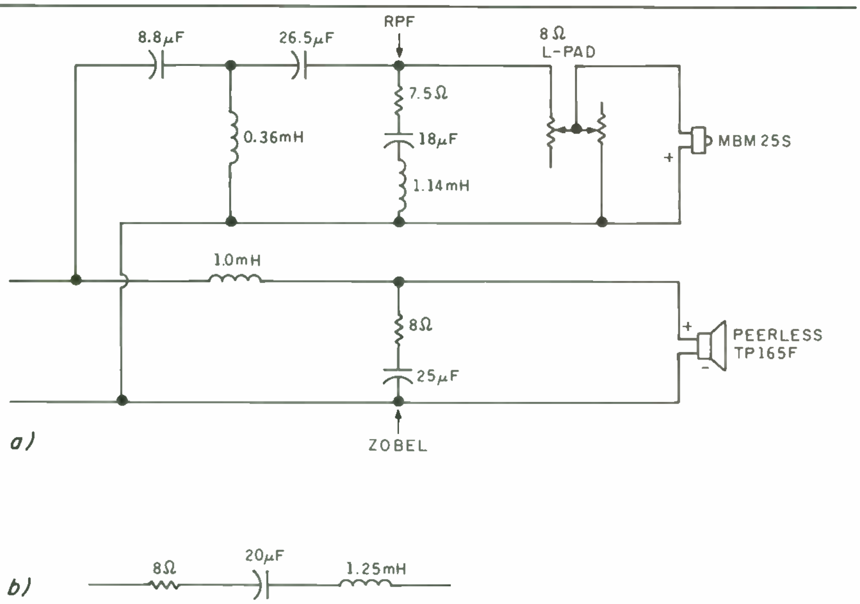


**FIGURE 9: Crossover network for Peerless TP165F woofer and MBM25S tweeter [crossover frequency, 2kHz] (a), and the RPF you should use with an MCD25M titanium tweeter (b).**

solution is to move the tweeter to a position on top of the enclosure. A small baffle here should give you improved horizontal distribution.

The tapered pipe, with its low cost and construction ease, should be a contender in the enclosure race. It does require a bit more tinkering with damping material than a simple box, but challenge is part of the game.

The damped pipe has several advantages over the bare pipe, including one that might not be obvious. In 1963, Alan Lovell, news editor of the British magazine *The Tape Recorder*, heard unusual noises coming from one of his tapered pipe enclosures. After analyzing the sound, he found his cat had crawled into the pipe and was stuck. He called and called, the cat howled and howled, but it didn't move. Mr. Lovell had to take his enclosure apart to get the cat out. Another builder reported the first time his cat saw a tapered pipe, it went right into it.

As a result of these incidents, warnings were issued for builders to place screens over their pipes' ports. So, if you run into trouble, don't expect to use the discarded pipe as a cat residence. But, if you know anyone who has a pet snake...

## ABOUT THE AUTHOR

*David B. Weems began experimenting with speakers in the 1950s and has published speaker projects in many periodicals including* Popular Mechanics. *He is the author of* How to Design, Build & Test Complete Speaker Systems *(Tab Books #1064, out of print) and* Designing, Building & Testing Your Own Speaker System, *2nd Ed. (Tab Books #1964).*

## REFERENCES

1. Weems, David B., "How To Design, Build and Test Complete Speaker Systems," TAB Book 1064, 1978 (out of print).
2. Galo, Gary, "Transmission Line Loudspeakers," *SB* 1/82.
3. Baldock, R.N., "A Preliminary Paraline Postscript," *Hi-Fi News*, June 1963.
4. Baldock, R.N., "Now, A Tricolumn," *Hi-Fi News*, April 1961.
5. Baldock, R.N., "A Preliminary Paraline Postscript," *Hi-Fi News*, June 1963.
6. Bullock, Robert M. III, "Passive Crossover Networks," *SB* 1/85.
7. *Loudspeaker Design Cookbook*, Speaker Research Associates, second edition, 1978. (Out of print)

# THE DEMONSTRATOR:
## A VENTED, COMPOUND SPEAKER SYSTEM

BY JOHN COCKROFT

Since my first article on compound speakers[1], I have received several queries on how to apply the principle to a vented configuration. I knew it could be done because one reader told me a patent had been granted for such a speaker.

Still, I didn't really know how to begin. After considerable thought, however, I decided to start with a conventional vented system, modify it, and then compare it to the original. This concept led to my design of The Demonstrator.

I used Radio Shack's #40-1022 4″ speaker because I liked its performance and already owned several of them. I designed my enclosure based on Hoge's version of Keele's approximate box method[2], a somewhat shorthand version of Thiele/Small computations. Keele's method is actually in two parts. The first computes an enclosure that is close to optimum, and the second designs a system by starting with an arbitrarily-sized box. If you are unfamiliar with these procedures, use the following equations. Use the first set to obtain an optimum (or close to it) system for your speaker. Keele presumes your speaker has a $Q_{ts}$ of 0.8 or less (less is preferable):

$$V_b = 15\,V_{as}(Q_{ts}^{2.87}) \qquad (1)$$

$$F_3 = \frac{.26F_s}{Q_{ts}^{1.4}} \qquad (2)$$

$$F_b = \frac{.42F_s}{Q_{ts}^{.9}} \qquad (3)$$

Before you use the second set of equations, first decide upon the volume of your enclosure:

$$F_3 = F_s\frac{(V_{as})^{1/2}}{V_b} \qquad (4)$$

$$F_b = F_s\frac{(V_{as})^{.32}}{V_b} \qquad (5)$$

$$R = 20\log_{10}\left[2.6\,Q_{ts}\frac{(V_{as})^{.35}}{V_b}\right] \qquad (6)$$

where R = the response ripple magnitude due to the non-optimum box. If the figure is negative, it indicates a dip; otherwise, you have a peak.

I altered the equations so you can use them a third way: to design a system for a given speaker to a desired $F_3$. As with the second method, use equation (6) to make sure the ripple isn't greater than you desire:

$$V_{as}/V_b = (F_3/F_s)^2 \qquad (7)$$

$$V_b = V_{as}/(V_{as}/V_b) \qquad (8)$$

$$F_b = F_s\,(V_{as}/V_b)^{.32} \qquad (9)$$

All of the above equations assume Q1 = 7. You can use all of the above with any measuring unit, including liters, cubic feet, or cubic inches.

When designing The Demonstrator, I used the second method and chose a box size of 450 cubic inches net (0.2604 cubic feet). The published specs for the Radio Shack #40-1022 are: $F_s = 55$Hz, $Q_{ts} = 0.35$, $V_{as} = 0.23$ cubic feet. Using equations (4) and (5), I found $F_3 = 51.7$Hz and $F_b = 52.8$Hz. From equation (6), I found a ripple of −1.2dB.

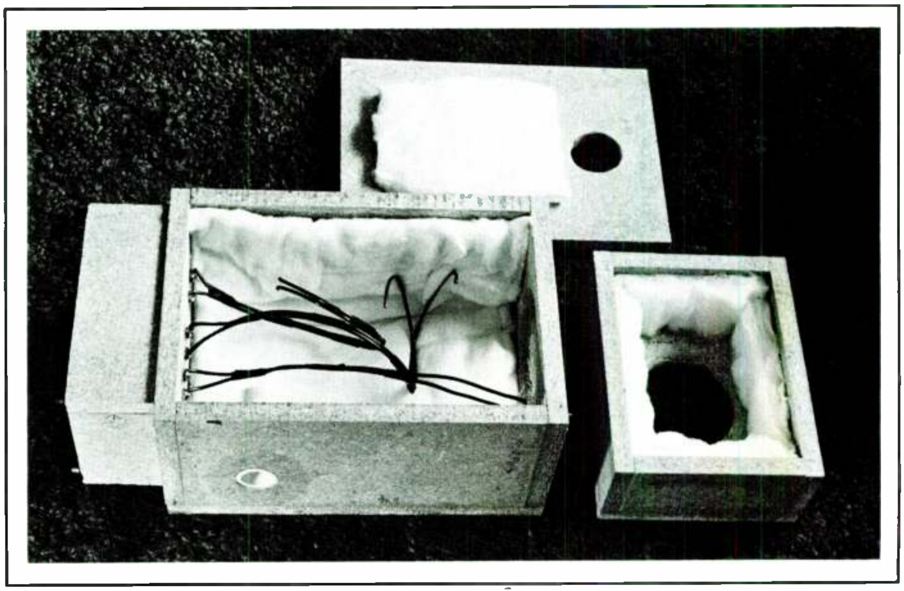

**PHOTO 1: Internal view of box and superstructure showing location of the duct and tweeter hole. Note: the author later replaced the cotton batting with 1″ fiberglass. See text.**

**BULLOCK'S WISDOM.** Next, I constructed the box. In the accompanying photos, you can see cotton batting in the enclosure. Before I completed the box, however, I changed the batting to 1″ fiberglass after reading an article in *SB* by Robert Bullock. He said fiberglass is a more reactive acoustic material and has less effect than such resistive materials as polyester and, I presume, cotton in the lower frequency range. In a vented system, the acoustic material merely damps the midrange reflections, and in the lower regions, it actually interferes with the reflex action.

In my *SB* 3/85 article, I opined that compound speakers were squat and ugly. I was determined to change these characteristics while designing The Demonstrator. I came up with a box in which the bass/midrange speaker faced out from the top, *a la* Roy Allison's loudspeaker designs. (Note: for best results, this arrangement works best when the speaker is close to the rear wall [½″ or less away], and away from corners.)

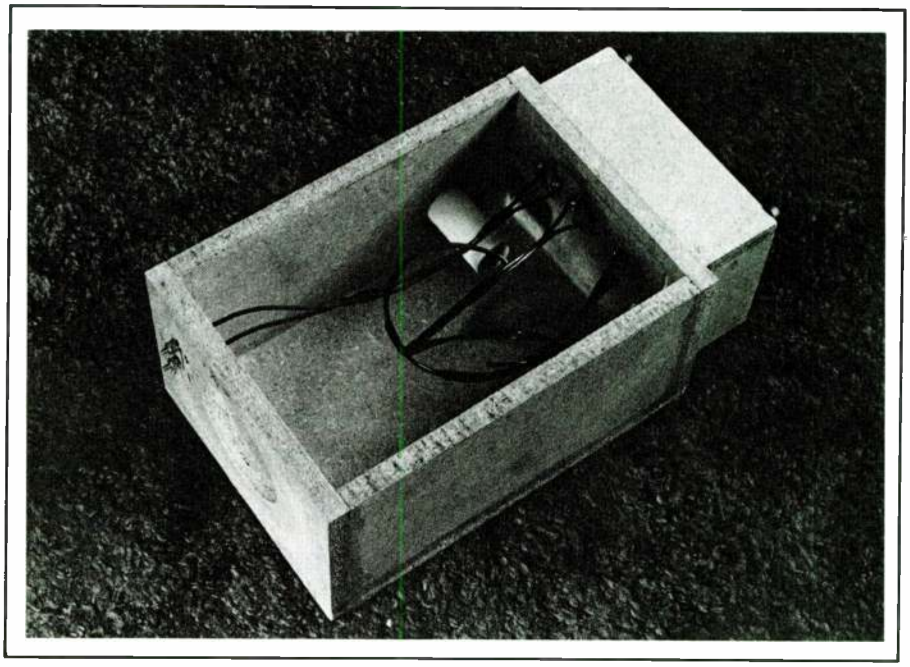

**PHOTO 2: Internal view of box before the acoustic material is added. Note wiring feedthroughs and duct.**

In the compound version, I placed the superstructure for the second speaker directly above the first speaker, which made the enclosure look even taller and slimmer. I placed the crossover in the pedestal beneath the enclosure, in the manner of my Project One (*SB* 3/85), thus completing a new, trimmer look that featured a sound, sonic design. To enhance the Allison concept, I placed the duct on the side, in a low position toward the rear, to more closely couple it to the room. I placed the tweeter in a conventional location, at the front of the enclosure.

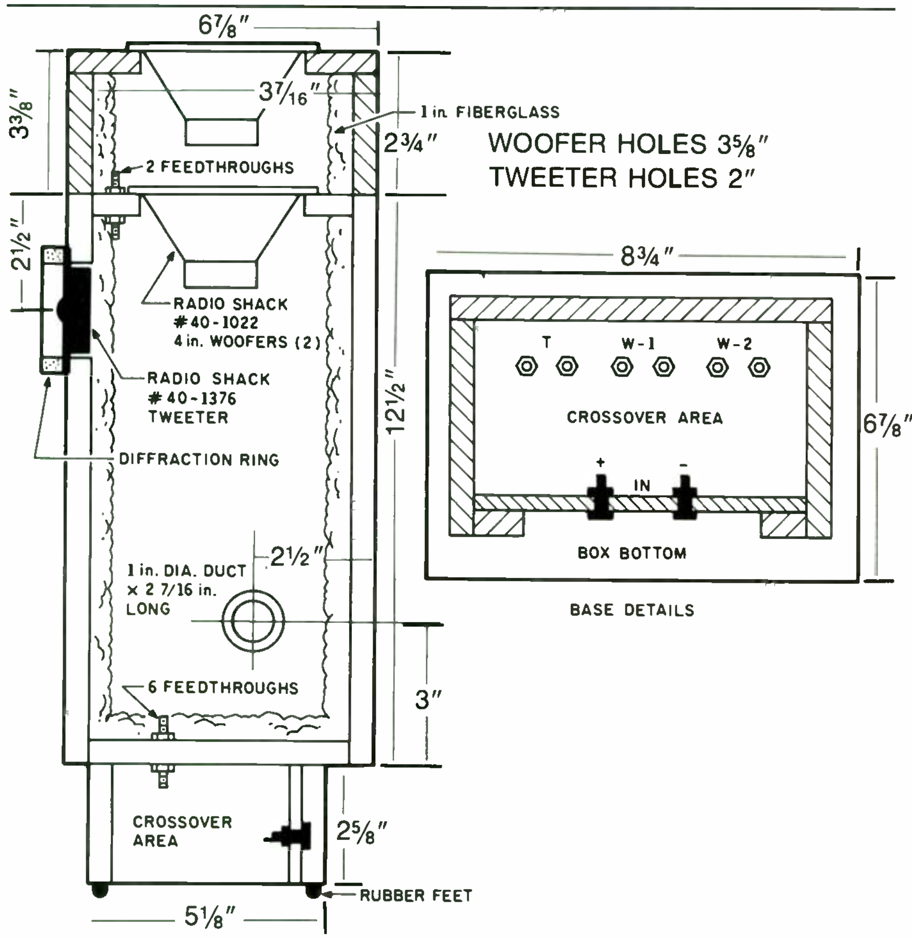


**FIGURE 1: Cross section of The Demonstrator showing details of the enclosure and locations of the drivers and the duct.**

**DUCT MEASUREMENTS.** Once I constructed the box, I made measurements, the most satisfactory of which I made with a 1″ diameter, 2 7/16″ long duct. (I have always had trouble calculating ducts from the various available equations. They never seem to come out right. I now resort to using a factor of about 0.7, and then proceeding by trial and error.)

Using the above-mentioned duct, $F_h = 79Hz$, $F_l = 31Hz$. $F_h$ and $F_l$ are the two impedance peaks at the low end of the frequency range of a vented system. With the duct removed and the hole sealed, $F_c = 70Hz$. From the above measured information, I calculated $F_b$, $F_{sb}$, $V_{as}/V_b$ and $V_{as}$:[3]

$$F_b = (F_l^2 + F_h^2 - F_c^2)^{1/2} \qquad (10)$$

$$F_{sb} = \frac{F_l F_h}{F_b} \qquad (11)$$

$$V_{as}/V_b = \frac{(F_h + F_b)(F_h - F_b)(F_b + F_l)(F_b - F_l)}{F_h^2 F_l^2} \qquad (12)$$

$$V_{as} = (V_{as}/V_b)\, V_b \qquad (13)$$

In this case, $F_b = 47.9$Hz, $F_{sb} = 51$Hz, $V_{as}/V_b = 0.8774$, and $V_{as} = 0.2285$ cubic feet. $V_{as}$ is very close to the published spec of 0.23 cubic feet. $F_{sb}$ is the resonant frequency of the speaker in the ducted enclosure, so I substituted it for the free air resonance (given as 55Hz).

With some actual figures to work with, I again used equations (4), (5) and (6) to see how the constructed system compared to Keele's model. Keele says $F_3$ should be 47.8Hz when $F_b$ is 48.9Hz, and the response ripple should be −1.22dB. The actual $F_b$ is off 1Hz from the model, or a factor of 2.1%. I doubted whether my measuring instruments were that accurate, so I stopped here.

I spent a week or so listening to my speaker, trying all kinds of music on the FM band, and testing several Telarc CDs featuring organist Michael Murray. The speaker performed very well, regardless of music category, and I was impressed by the organ CDs when played at reasonably moderate levels. It convinced me, once again, of the soundness of Keele's equations.

The speaker, as it stood at this point, would be useful for many small speaker applications. I now had a quality module I could modify.

**MODIFICATION STEPS.** The superstructure was built. It's dimensions were the same as the top of the enclosure, and its height raised the second speaker 2¾″ above the first one. I lined it on four sides with 1″ fiberglass, and soldered two speaker leads to feedthroughs which I mounted in the enclosure's top panel during the initial construction. (I wired all speaker units individually down through the enclosure's bottom panel. This makes it easy to alter the connections.)

Next, I mounted the upper structure to the enclosure with a ¼″ bead of silicone adhesive. I pressed down the top until about 1⁄16″ of rubber separated the parts. After letting it dry for 24 hours, I used a single edged razor blade (frequently dipped in water) to trim the excess rubber. I mounted the driver and wired the two in series, making sure both cones moved forward on a positive pulse (checking the inner one before mounting the outer). Finally, I combined the double-driver speaker into the crossover.

At this point, I again measured the system. I started with the same duct, which proved to be good for both versions. I ran into a problem, however, while trying to measure $F_c$: I obtained a different reading when I swept down the frequency range than when I swept up. I finally settled on the average of three down and three up sweeps, which was 68Hz. The rest was trouble-free: $F_h = 75$Hz, $F_l = 33$Hz, $F_b = 45.7$Hz, $F_{sb} = 54.1$Hz, $V_{as}/V_b = 0.577$, and $V_{as} = 0.1503$. The response ripple worked out to −2.49dB.

Once more, I plugged into Keele's equation, which said with $F_b = 45.4$Hz, $F_3$ should be 41.1Hz. Perhaps it was merely coincidental how well the same duct worked for both systems.

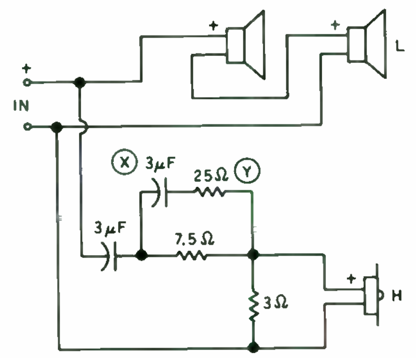


**FIGURE 2: Crossover network for The Demonstrator. "X" and "Y" add a boost of 1.75dB above 14kHz. This boost is optional**

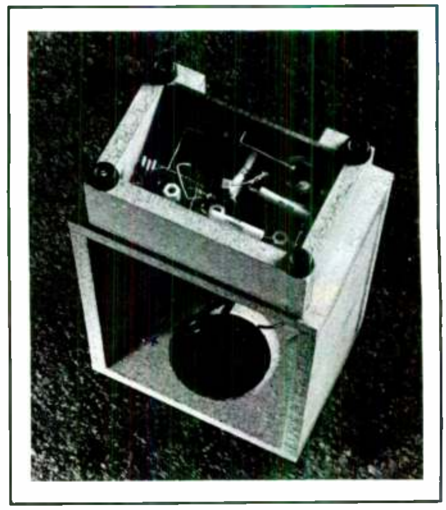

**PHOTO 3: Bottom view of unfinished speaker box showing crossover area. Although the author modified the crossover before completing the project, the general idea remains the same.**

**LITTLE COMPETITION.** I designed my system in a rather arbitrary way, and the data tells you only about my system. Other approaches, and other drivers with, for example, different values of $Q_t$, will undoubtedly produce different results.

I believe The Demonstrator's sound will hold its own in just about any competition with speakers of similar size, as well as with many larger, more complex designs. With excellent source material, it produces an open,

**TABLE 1**

**PARTS LIST**

| | |
|---|---|
| **5/8″ particle board (a 6′ shelf sold at department stores):** | |
| 2 | 12½″ x 8¾″ front and back (A*) |
| 2 | 5⅝″ x 8¾″ top and bottom (B) |
| 2 | 11¼″ x 5⅝″ sides (C) |
| 1 | 8¾″ x 6⅞″ superstructure top (D) |
| 2 | 8¾″ x 2¾″ superstructure front and back (E) |
| 2 | 5⅝″ x 2¾″ superstructure sides (F) |
| 1 | 7½″ x 2⅝″ base front (G) |
| 2 | 4½″ x 2⅝″ base sides (H) |
| 2 | 2⅝″ x 1″ base cleats (I) |
| **¼″ masonite or plywood:** | |
| 1 | 6¼″ x 2 ⅝″ base back |
| **Capacitors and Resistors** | |
| 2 | 3μF 50V Mylar capacitor |
| 1 | 7.5Ω 5 or 10W resistor (2 15Ω in parallel) |
| 1 | 3Ω 5 or 10W resistor |
| 1 | 25Ω 5 or 10W resistor |
| **Miscellaneous** | |
| 8 | electrical feedthroughs (these may be brass 10-32 x 1¼″ machine screws, or lengths of threaded rod with nuts, washers and lugs) |
| 2 | input connectors (I used banana jacks) |
| 4 | rubber feet (optional) |

*letters denote part on cutting guide

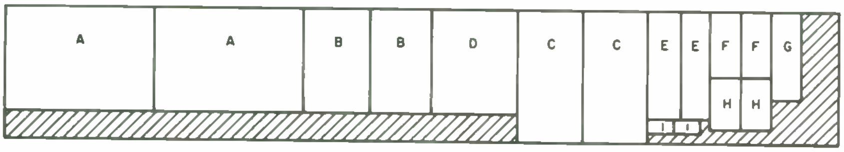


**FIGURE: 3 Cutting guide for 6′ x ⅝″ particle board shelving material. See parts list for identificaton. Shaded areas indicate scrap material.**

solid, and intact sound that is a joy to hear.

Since I have constructed only one of these drivers, I have yet to hear my system in stereo. I have, however, constructed other systems of this size and configuration that give excellent stereo performance. When I have built other systems using a single Radio Shack #40–1022, I have noticed a very slight heaviness imparted to voices. To eliminate this problem, I set an equalizer to about −2dB at the 240Hz setting. The Demonstrator, however, didn't show this slight aberration. Since I used the same speaker units I had used in my previous designs, I believe the compound configuration caused the damping of the slight peak.

Out of curiosity, I took The Demonstrator's meaurements and substituted $V_{as}$ for a single rather than compound speaker. $V_b$ came to 684 cubic inches for an enclosure that performs as well as The Demonstrator (I only calculated the enclosure, I didn't build one).

Though I wrote this article to show how I designed a vented, compound loudspeaker system, I have included sketches and a parts list should you wish to build one for yourself.

**REFERENCES**

1. Cockroft, John, "An Isobarik System," *Speaker Builder* 3/85, pp. 7- 12.
2. Hoge, W.J.J., "Confessions of a Loudspeaker Engineer," *Audio*, August 1978.
3. Small, R.H., "Vented Loudspeaker Systems," Part II, *JAES*, July/August 1973.



## If you should have a ***TECHNICAL QUERY...***

about an article appearing in this magazine, write it clearly, leaving space for a reply and referencing the magazine, the article and the page about which you are inquiring. Enclose a self-addressed stamped envelope and send these to *Speaker Builder*, Technical Dept., P.O. Box 494, Peterborough, NH 03458.

Help us by not calling in your question. We have neither the staff nor the time to respond to technical questions by phone.


# WE VISIT BOSTON ACOUSTICS

BY BRUCE EDGAR
Contributing Editor

**Speaker Builder:** *How did you get started in the speaker business?*

**Andy Petite:** I started as a consumer. And then I went to work in a hi-fi store, working parttime while I was in college. After I graduated, I managed the store. A few years later in 1964, I went to work for KLH when Henry Kloss was there. I didn't do any speaker work until the very end of my stay at KLH.

**SB:** *What was your job at KLH?*

**AP:** I was Assistant Sales Manager. When Advent started up in 1969, I was one of its charter employees. Henry Kloss was working on the original Advent speaker, and I asked him if I could become involved in the final sound of the loudspeaker. He said yes, and I did. The speaker was a commercial success. As a result, with the next product, the Smaller Advent loudspeaker, he involved me at an earlier stage and let me do the crossover network. It was also a success. He let me collaborate on the next product at an even earlier stage. I apprenticed, you might say, with Henry Kloss. When he left Advent, I took over all the speaker design work. The three products that were totally my designs were the Avent/3, the New Advent LoudSpeaker, and the Advent/1.

**SB:** *In the beginning there was only the one Advent loudspeaker without the usual model number. What was the marketing reasoning behind that choice?*

**AP:** Henry felt that the cost difference between a good speaker and one that covered the full range of music was not that great. The original Advent loudspeaker was designed to cover the full range. And since there wasn't a more expensive speaker in the product line to protect, you put everything into that one loudspeaker product. By utilizing a combination of factors like more modern materials and certain manufacturing efficiencies that arise from one model, he found that it was possible to produce a quality speaker at a lower price. The public responded enthusiastically to the Advent Loudspeaker, and it was very successful.

**SB:** *It is interesting that you were not trained technically for loudspeaker work, but that you gradually worked your way into it.*

**AP:** You can say that I apprenticed in speaker design in an old world sense. In fact, I would say that most of the people who have been historically involved in speaker design have gone through an apprenticeship period.

**SB:** *You could not have found a better person than Henry Kloss to learn from.*

**AP:** I have often claimed that Henry has forgotten more about loudspeakers than most other people knew. I have had instances in the past where I've run into design problems and have consulted with Henry. He had invariably run into the same problem and usually had a superb solution that got to the heart of the problem.

**SB:** *At what point did you break off from Advent?*

**AP:** I left Advent in 1978 because I had become disillusioned with the direction the company was taking. I didn't know what I was going to do, but Frank Reed talked me into forming Boston Acous-

**FIGURE 1: Andy Petite.**

tics within three weeks after I left Advent.

**SB:** *How long did it take to start Boston Acoustics?*

**AP:** Well, the company was officially incorporated in February 1979, and our first product was in production the last week of July. I had a lot of pressure.

**SB:** *That's a six month start-up. Were you prepared for that schedule?*

**AP:** Not really, I didn't have a product design in my back pocket, and most people who break off from a company and start a new business usually have a product design to go with.

**SB:** *What was your first product for Boston Acoustics?*

**AP:** It was the A200, which was a three-way speaker with a large front baffle and thin cabinet.

**SB:** *That is an interesting choice of box proportions. What made you go with that approach?*

**AP:** The rationale for it was actually espoused by our industrial designer who is also an engineer and is experienced in acoustics. The idea behind the A200 was to deal with the change in radiation pattern that occurs with almost any loudspeaker because of the "wall dip" that Roy Allison [*First editor of* High Fidelity *and founder of Allison Acoustics.—Ed.*] first described. The dip occurs where the wavelength of the sound being generated is larger than the front baffle but small in comparison to the projection of the cabinet from the wall. A dip in the mid-bass radiated power will result. The thinking behind the A200 was to make the cabinet very thin so as to "fool" the woofer into thinking that it was mounted in the wall. The path length from the woofer to the wall is short so that there are no significant disruptive reflections off the rear wall. Then you only have to worry about the floor reflection. In the A200 we put the woofer very close to the floor and the crossover frequency just below where floor reflections become significant.

**SB:** *When did you get into the small speaker game?*

**AP:** Our first small speaker was the A40 which was the sixth product in our line. For small speakers such as the A40, there is no real advantage in making the cabinet wide and shallow. The real genesis of the A40 started from my belief, probably inherited from Henry, that most of the good sounding mini-speakers on the market were overpriced. I wanted to make a mini-speaker that sounded as good as those expensive speakers but one that was more realistically priced. The lower price would be based on the assumption that we could sell it in large enough quantities to create manufacturing and purchasing efficiencies. And also by making the drivers as good as possible, we would not need a costly and elaborate crossover network to correct for driver deficiencies. (*Fig. 2.*)

**SB:** *As a matter of historical trivia, why do you use the letter "A" in your model number?*

**AP:** Coming up with model numbers is a really arcane endeavor. Actually, it was the idea of our industrial designer, who thought a letter prefix before the number was better than just a number. Several approaches have been used in coming up with model numbers. One familiar one is to number models chronologically, like the AR-1, AR-2, etc. KLH did the same. However, after a while it becomes awkward because you have models like the AR-4 which was less expensive than the AR-3. And it becomes confusing to the consumer who hasn't followed the chronological progression. At Boston we tried to avoid this problem by using a system whereby a higher model number implies a more expensive speaker.

**SB:** *At the start up of Boston Acoustics, were you making your own drivers or buying from other manufacturers?*

**AP:** We started off by making all of the woofers, and in the case of the A200 we bought the midrange and tweeter. Eventually, we introduced our own tweeter in 1982. (*Figs. 3–10.*)

**SB:** *Boston Acoustics differs from most of the other mainstream US speaker manufacturers in that you have changed from buying your drivers to making them. Why did you choose that course?*

**AP:** There were three reasons: quality, price and reliability. When we were buying tweeters from outside vendors, we started to find inconsistencies in the quality and performance. Basically we decided to make our own tweeters in order to control consistency. If you have rejects in a batch of imported tweeters, you have to send them back, and that becomes expensive. Whereas, if you are making your own tweeters and run into a problem, you can fix it on the spot and go on. (*Figs. 11–15.*) The second reason was cost. We felt that we could make a better quality tweeter for less money than it would cost us to buy it. A third reason, which became as large a factor, was reliability. We found the field failure rates for our purchased tweeters were too high. To improve reliability, we wanted high temperature voice coils, but it was inconvenient for the vendor to incorporate high temperature voice coils, because they were burning back the insulation on the voice coil leads by putting them in a solder bath. You can't do that with high temperature insulation. But high-temp wire makes a big difference in field reliability.

**SB:** *What do you do to your tweeters to increase their reliability in the field?*

**AP:** We use flexible lead wire with a rigid attachment point and a high temperature voice coil. We also use high temperature adhesives. None of these factors by themselves are directly responsible for increasing reliability in a large way, but it is a combination of them all.

Instead of using normal tolerances of ±0.001″ on our tweeter parts, we specify ±0.0005″. Thus we can make the tweeter gap 0.002″ smaller. That's fairly significant when you are talking about a very narrow gap to begin with. We are able to achieve a slightly higher efficiency, which means the tweeter

**FIGURE 2: The Boston Acoustics A40, one of the better small affordable speakers on the market.**

FIGURE 3: Woofer production begins by attaching the magnet-pole piece assembly to the basket with a quick-setting adhesive.

FIGURE 4: To assure alignment in prouction, the voice coil is inserted on a precision-formed plug.

doesn't have to work as hard. You get the same acoustic power out for less power into the tweeter. All of this contributes to the additional reliability.

**SB:** *Who is in charge of your driver design and development?*

**AP:** Moses Gabbay.

**SB:** *What is his background?*

**AP:** Moses is a physicist by degree. He was born in Egypt and came over to this country about 15 years ago. He worked at AR in the late '60s and early '70s and then went to AVID as their chief engineer. (*Fig. 16.*)

Another engineer, Winslow Burhoe, has recently joined us. He was founder of EPI, but his experience dates back to the early days of AR when he was involved in the development of the AR-3a midrange and tweeter.

**SB:** *It's almost as if the New England "school of speaker design" has come home to roost at Boston Acoustics. In your driver production, what parts do you make in-house and what parts do you buy from vendors?*

**AP:** We really don't make any component parts in-house. The baskets, for example, are made for us by other manufacturers with our tooling. Our diaphragms, surrounds, voice coils, etc. are becoming custom parts that we specify to a vendor.

**SB:** *Are your vendors in the US or "off shore?"*

**AP:** Both. Wherever we can obtain the best part, we go there. For example, the pole plates come from the Orient. They make precision pole pieces by cold forging; nobody in the US currently does.

**SB:** *What material do they use for the pole pieces?*

**AP:** They use soft iron which is magnetically superior to the harder alloys that are often used. Before the final strike it is annealed in order to grain-orient the iron molecules. The surfaces of the finished pole pieces are so flat that we can achieve much more efficient magnetic circuits than we could with the parts we were using before.

**SB:** *Even though you make the drivers, don't you have a potential quality control problem because of multiple suppliers?*

**AP:** It's a constant problem. We share information with one Japanese driver manufacturer. They have the same problem, and most of their parts are sourced in Japan. And the Japanese vendors have a legendary reputation for consistency. But you talk to anybody that makes drivers, and they all tell the same stories.

**SB:** *Do some parts have more problems than others?*

**AP:** We rarely have driver rejects because of voice coils or magnets. The biggest problem is with diaphragm variations.

**SB:** *Even with polypropylene?*

**AP:** That's one of the motivations for using polypropylene diaphragms. They are more consistent.

**SB:** *Do you still use paper cones?*

**AP:** Yes, although I should add that our "paper" cones don't have any paper in them. The "paper cones" are made from a combination of felted wool and cotton rag. The 8″ driver in the A70, for example, uses a "paper cone." (*Fig. 17.*)

**SB:** *Why is the largest driver in your product line only a 10″?*

**AP:** The common assumption is that the larger the woofer, the better is the bass. That's not necessarily true if you

FIGURE 5: The operator on the left lays down two beads of adhesive on the basket for the outer edges of the spider and cone and inserts the spider and voice-coil alignment plug. The operator on the right lays in the cone and applies the adhesive for the spider-cone and voice coil interfaces in a two-step process.

**FIGURE 6: The woofers travel on a slow moving conveyor belt through an oven to cure the adhesives. The woofers shown in this photo have just emerged from the oven and are beginning to cool.**

are working with a given size enclosure. If you put a larger woofer into a box of about 1.5 cu.ft., the general results are that the system resonance increases and the bass response decreases because you've increased the stiffness. To bring up the bass response you have to add more mass and then more magnet and pretty soon you will end up with an AR-3 woofer. The 12″ AR-3 actually had no better bass than the Advent which had a 10″ driver and cost less to make. The only advantage the larger cone gives you, all other things being equal, is slightly higher maximum output level.

**SB:** *So your largest is a 10″ driver?*

**AP:** Well, actually at present our largest is a dual 8″ which has the same radiating area as a 12″.

**SB:** *You've been very consistent in using acoustic suspension in your speakers. Does that come from your KLH and Advent heritage?*

**AP:** Yes, but there are basically two reasons. The first is that below cutoff a vented box is undamped. Particularly with records you have infrasonic energy problems and resultant intermodulation distortion. This problem may decrease as people go to CDs rather than records. The other problem is that the group delay is higher for a vented box, although I don't know how audible it is.

**SB:** *Do you believe the acoustic suspension enclosure is easier to mass produce?*

**AP:** Everybody knows that a woofer suspension changes over time, i.e., it becomes more compliant. In the case of a sealed box that increased compliance means no change in bass response.

In the case of a vented system, you may go from an optimally tuned design to one that is mistuned.

**SB:** *What type of crossover do you typically use in your systems?*

**AP:** Whatever works.

**SB:** *What do you use in the A40?*

**AP:** The original A40 had a 6dB/octave network. In the new A40 series II, we use a two-pole crossover on the tweeter to increase power handling and decrease distortion near tweeter resonance.

**SB:** *What about the woofer?*

**AP:** We use anywhere from no crossover in some of our products to a three-pole crossover. It depends on what works in a particular application. In the case of the A400 with two woofers, we used a three-pole at 30Hz because slower roll-offs caused interactions between the woofers and midrange. We did try a two-pole which meant that you would have to invert the polarity of the midrange to avoid a hole in the crossover region.

**SB:** *Do you find inverting the polarity of the driver to be a problem?*

**AP:** If you really want to know how I feel about inverting polarity, I try to avoid it. On the other hand, some very highly regarded speakers have drivers with inverted polarity. The LS3-5A, which is a legend in the business, has a tweeter with inverted polarity, and nobody has complained about phase problems with that speaker. So intellectually I don't have a problem with inverted polarity although I do have a bias against using it in our speakers. Out of all the speakers that I've been involved with, only one has had a driver with inverted polarity.

But sometimes you have to invert polarity to get the response right. Recently, some B&K representatives gave us a demo of their two channel FFT, which will give a plot of acoustical phase. One B&K engineer, said it was impossible for a system with an "out of phase" driver to have good phase response. So I asked him to test a two-way system in which the tweeter polarity was inverted and tell me which plot looked better: inverted or uninverted. He picked the phase plot with inverted polarity and was absolutely shocked.

**SB:** *Do you have any other thoughts on crossovers?*

**AP:** Yes, I do. We try to design our drivers to be as smooth as possible including the way they roll off so that they don't require elaborate crossovers to correct for their deficiencies. If you have a driver that is very flat in the passband but has horrendous peaks above the passband, there is no way that you can properly use a single-pole crossover or even a two-pole.

If you measure our woofers, you will find that they are all very smooth and well behaved at the top end. That is why we use a "paper cone" without high Q cellulose material in it. Our polypropylene cones also damp out the break-up modes in the material. The smooth roll-off allows you to put two drivers in a two-way system and not have to resort to a fancy crossover network which is expensive and often causes weird impedance variations that amps find hard to drive.

**SB:** *Do you find you prefer iron-core or air-core inductors?*

**AP:** No, I don't have a preference for one or the other. You can get ferrite cores that have extremely high current capabilities without saturation. But it really boils down to what you are trying to achieve in the crossover region. If you have a 4Ω woofer and have a 3mH (air-core) choke in series with it, you will lose bass response below resonance, due to power losses in the choke. You may give up several dB by using an air choke. So if you have to use a large choke, either you make the woofer with a higher impedance, or you use a ferrite- or iron-core inductor. In the case of a parallel choke, there are times when you want to use a lossy inductor, and you can use an air-core choke with small gauge wire. So we use both air core and iron-core inductors selected on the basis of performance.

**SB:** *What type of capacitors do you use in your crossovers?*

**AP:** We use high-grade, non-polarized electrolytics. The one criticism I've heard of electrolytics is that they don't

**FIGURE 7: The alignment plug is removed, and flexible leads from the voice coil are attached to the input terminal strip.**

age gracefully. I haven't found that to be true. I have some original Advent loudspeakers that are now 17 years old. I check them periodically and they have stayed pretty much the same. We buy our capacitors with a low dissipation factor and where possible with a high temperature rating of 85°C or more for which we pay a premium.

**SB:** *Have you tried Mylars in your crossovers?*

**AP:** Recently, I did some very careful comparisons between Mylars and electrolytics. They are different in that the Mylar is closer to an ideal capacitor. If you look at the voltage across the driver when you use Mylars, it's higher at turn on. Another way of looking at the comparison is that you can get the same voltage across a driver with an electrolytic as a Mylar by choosing a larger electrolytic. In practice we find we use a smaller value Mylar than we would be if we were using an electrolytic.

**FIGURE 9: Bruce Edgar and Andy Petite examine a completed driver which has been stored on the racks awaiting installation.**

**SB:** *Do you hear any audible differences between Mylar and electrolytic capacitors?*

**AP:** I haven't really set up two speaker systems with the same response but using Mylars in one and electrolytics in the other. But I think people who say that if you replace an electrolytic with a Mylar, you will hear a different sound—they're right. However, the difference results from having a different frequency response. And the general perception is that if you immediately hear a change, it must be better. But that conclusion doesn't always follow, because the change may not be what the designer intended.

**SB:** *What do you do in the way of testing of individual drivers and systems?*

**AP:** We test every finished speaker system for frequency response and for rattles, buzzes and distortion. We also test for system resonance frequency. (*Fig. 18.*)

**SB:** *Do you test individual drivers before they are installed?*

**AP:** Yes, we do. We also test crossover networks to a tolerance of ±0.5dB against a reference network. We follow that procedure because we have a stringent tolerance on the frequency response on the whole system. Also if there is anything wrong with the final frequency response, we want to eliminate the crossover network as a possible cause because it is a real pain to repair a crossover once it is in a box.

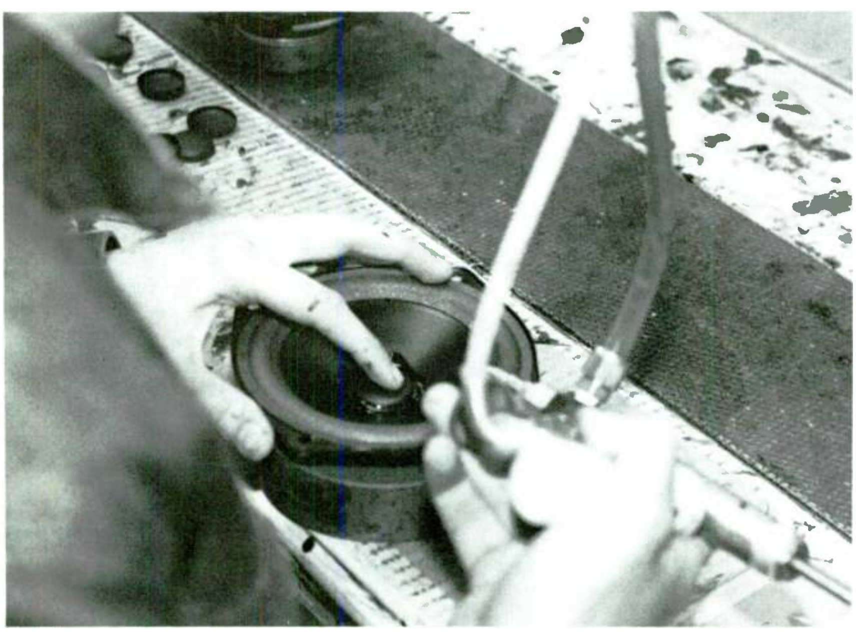

**FIGURE 8: In a final step, the dust cover is glued on.**

**SB:** *Do you find that you have to match pairs of speakers?*

**AP:** Not really, because all the drivers must be matched to a reference with a tolerance of ±1dB.

**SB:** *Is that an empirical standard?*

**AP:** No, those tolerances are what we want to maintain. There is much mysticism about loudspeakers, but it is true if you control most of the physical parameters of a loudspeaker, it will come out right. The parameters that are important are the voice-coil characteristics and the diaphragm mass, especially with the tweeter. The latter also includes the control of the correct amount of adhesives since they will affect the mass. The only factor you really can't measure before speaker assembly is the consistency of the diaphragm.

**SB:** *Do you check samples in a batch of diaphragms for mass consistency?*

**AP:** Yes, we do at incoming inspection.

**SB:** *In testing your drivers, where do most of your rejects come from?*

**AP:** There is more potential for rejects with tweeters than with woofers because the gap is very tight in tweeters and dust particles can cause problems. About 90% of our tweeter rejects come about because they have a reduced output near resonance which indicates a rubbing voice coil. The other 10% of the rejects come from a falling high frequency response.

**SB:** *I notice you have recently introduced* *a new top-of-the-line speaker system, the T1000, which seems to run counter to your low-cost speaker philosophy.*

**AP:** Well, for what performance it offers, it is "low cost." The T1000 competes wih speakers costing 40% more.

**SB:** *What does the T1000 do better than the $1400 class of speakers?*

**AP:** Two aspects of the model make it better than the competition. One is price versus performance. The second is that we are taking a rather different approach with the woofer/midrange crossover point. If you use a 2″ dome for a midrange, you tend to be reluctant to place the crossover point below 500Hz because of power handling restrictions. We use a 6½″ driver that allows us to crossover at 250Hz which is well below the normal woofer/midrange crossover point. As a result most of the fundamental midrange information in the T1000 is coming from a location on the cabinet at ear level.

In a conventional design the woofer reproduces much of the fundamental midrange, and usually it is located at knee level.

**SB:** *Are there any other different aspects about the T1000?*

**AP:** It has a new generation 1″ dome tweeter with a redesigned faceplate which helps to smooth out the extreme top end. We are also using Mylar capacitors in the crossover.

**SB:** *A big part of your market now is automobile speakers. What do you do differently for auto installations?*

**AP:** Everything has to be different. The first difference is that you have to make them as small and compact as possible to fit in all the nooks and crannies. You have to make them reasonably weatherproof. They have to have a higher free-air resonance than a speaker designed or an enclosure.

**SB:** *Do rear cavity effects vary from car model to model?*

**AP:** They are remarkably similar. The only exception that I can think of was an older Camaro model, which had an unusual angle on the rear window.

**SB:** *How do you simulate the interior of a car in your lab?*

**AP:** Basically, a car's interior is best simulated as if it were an infinite plane baffle because the speaker tends to be mounted either on a door or on the rear package shelf. We find you get a very good correlation between testing car speakers on a infinite plane baffle and the way they perform in a car.

**SB:** *Where do you see Boston Acoustics going in the market?*

**AP:** I think there are a number of trends going on with consumers. They view smaller as better which is unfortunate because that's working against the

**FIGURE 10: The drivers are installed in cabinets on the assembly line.**

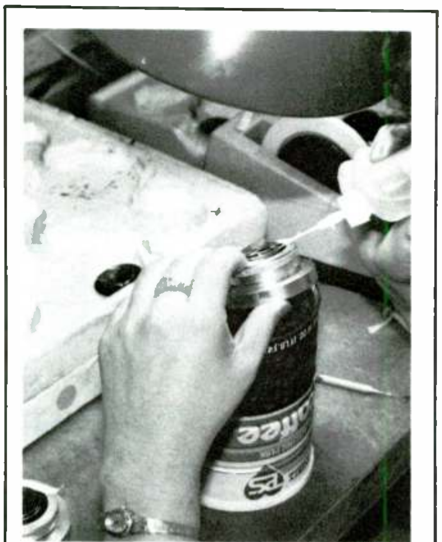

**FIGURE 11: The tweeter production begins with attaching the voice coil to the diaphragm with a specialized adhesive.**

**FIGURE 12: Andy Petite holds the voice coil/diaphragm assembly after it has been affixed in the tweeter frame.**

laws of physics. Built-in speakers are becoming an area of great interest. That is a whole world of business that we were previously unaware of. It comes from designers, interior decorators, architects, etc. Many of our dealers are doing that type of business now, and they tell us that it's growing.

If you ever read *Home Entertainment*, you will notice that speakers are either hidden or out of the frame of the camera. I think that interior designers consider speakers the eyesore of the whole system. Everything else seems to be elegant and very integrated into the room, and speakers don't. Built-in's may be a solution.

**SB:** *What is your new built-in unit?*

**AP:** The System 360 is a two-way speaker with a 6½″ woofer and a 1″ dome tweeter.

**SB:** *What do you do with the back volume?*

**AP:** The woofer is designed to perform properly with almost any size volume behind it. The way you do that is to make the woofer with a higher free-air resonance and Q than you would if it were being used in an enclosure. Normally, for a sealed box enclosure you would design it for a very low free-air resonance.

**SB:** *What about video applications?*

**AP:** Built-in speakers are terrific for video, especially the media room. The most exciting development to come along in a long time, the compact disk not withstanding, is the "video theater" with surround sound. I think as more people experience video systems with surround sound, they will want it for their homes. One problem is what do you do with the additional speakers. Where do you put them? Built-in speakers will have the advantage in those applications.

**SB:** *You seem to rely heavily on a broad base of empirical knowledge rather than a theoretical approach. Is that a correct assessment?*

**AP:** Yes, and I have observed that there is a real dichotomy between empirical and theoretical people. If you hear the loudspeaker papers at an AES convention given by academics, you can frequently hear the designers in the room snickering because it is clear that theoreticians have never tried to design and build a loudspeaker system.

My favorite example is Marshall Leach, who recently presented a paper on designing crossover networks for real-world loudspeakers. For years he has given papers on phase correct crossover networks. He and a grad student decided to design a loudspeaker system, and lo and behold they found out that none of that theory worked because a driver is not a resistor.

**SB:** *Where does Boston Acoustics see itself in today's world?*

**AP:** I think we see ourselves as catering to people who like music in their homes but don't want to readjust their lives around their music. In other words, they integrate their music around their lives. By that I mean that we don't make speakers that need to be moved well away from the walls because most people don't want to put up with that inconvenience in their homes. Most people want to place loudspeakers in a location that is convenient and mainly near a boundary. So we design our speakers to sound good in such a location.

This attitude comes from my experience with the Advent loudspeakers. Well over 90% of them were used on the floor, according to the warranty cards. But they never sounded their best on the floor. So this time around, we took that experience into consideration and designed our A100 and A150 mod-

**FIGURE 13: A precise amount of Ferrofluid is injected into the tweeter magnet/pole-piece assembly.**

**FIGURE 14: The magnet assembly is then inserted into the tweeter frame that has tabs which provide precise alignment.**

**FIGURE 15: A computerized test fixture is used to check the tweeter's response against the tolerance specifications.**

**FIGURE 16: Moses Gabbay, Boston Acoustics' driver designer.**

els so that they would sound right on the floor.

One of Henry's least successful designs was the KLH 12 which was a three-way floor standing model. One of the reasons that it was unsuccessful was that it didn't sound good where it had to go on the floor. Another manufacturer that I know of designed a floor standing speaker and then found that nobody liked it on the floor. It sounded better up in the air away from room boundaries. So he was smart enough to make it into a positive factor and sold an expensive speaker stand for it.

**SB:** *I detect from your attitude that you don't like speaker stands.*

**AP:** Yes, because a $200 speaker is no longer a $200 speaker with a stand. Instead, it becomes a $250 speaker.

**SB:** *Let me confirm something here. One of the impressive things about Henry Kloss was that he had a Henry Ford attitude about equipment. He wanted to design a piece of gear in such a way that it was as good as it could possibly be for as little as it could possibly cost because he thought many people would buy it.*

**FIGURE 17: Every completed speaker is checked in a chamber for frequency response.**

**FIGURE 18: The operator then turns the speaker around and checks the response sweep with his ear for rattles, buzzes, and any other distortion.**

**AP:** That's true.

**SB:** *It seems that you've absorbed that idea of audio not being an esoteric business for the few.*

**AP:** I have to admit that I have much the same feeling. I think what turned Henry on and what turns me on is designing an $80 speaker. Any designer who knows his business should be able to design a $1000 speaker. It takes much skill to design a good sounding $80 loudspeaker. And that's what turns me on.

# Book Report

**LOUDSPEAKER DESIGN COOKBOOK**, Third Edition, Vance Dickason (Marshall-Jones Co., order from Old Colony Lab, Part No. AA2, $14.95)

I recommended the second edition of Vance Dickason's *Loudspeaker Design Cookbook* (*SB*: 2/80, p. 30), and am even more enthusiastic about the third edition. This book should be on your shelf if you are serious about designing your own systems, even if you have a great deal of experience and expertise.

Mr. Dickason has collected a wealth of valuable information and organized it as a basic reference on loudspeaker design. In the unlikely event that he does not address a topic or covers one insufficiently, the extensive references should facilitate your search for more information. You would have to spend months gathering references to accumulate a comparable amount of information on your own.

This edition is organized somewhat differently than its predecessor. The chapter titles are now as follows: Closed-Box Low Frequency Systems, Vented-Box Low Frequency Systems, Passive Radiator Low Frequency Systems, Transmission Line Low Frequency Systems, Cabinet Construction: Shape and Damping, Mid- and High-Frequency Drivers, Passive and Active Crossover Networks, and Loudspeaker Testing.

Mr. Dickason has extensively revised all of his material to incorporate recent references. He has given some topics expanded treatment and included others for the first time. The new chapter on crossover networks, for example, replaces two chapters in the first edition. It contains much more information because of recent developments in the field.

The new book discusses the augmented passive radiator, a system that did not exist when the first edition was published. The first four chapters discuss implementing the various low-frequency loading techniques. Each of these chapters includes relevant notation and terminology, a brief history, and a description of the system and its strong and weak points.

Chapter 1 contains Small's classic alignment procedures and formulas for a closed box. In addition to giving the design formulas, the chapter presents many alignments in a convenient tabular form. Mr. Dickason includes a good discussion of enclosure-filling and dual-woofer configurations, addresses the question of power handling, and gives pertinent references on the topic of equalized alignments for closed boxes.

Chapter 2 includes a full set of tables for the standard fourth-order alignments and gives complete design procedures. Mr. Dickason discusses many details of vented-box design, and mentions others in references. The chapter is a thorough treatment of standard alignments, and provides references for higher-order alignments.

The third chapter follows the same format as the first two. It includes a discussion of the augmented passive radiator and gives an alignment table.

Chapter Four is on transmission lines, a design for which nobody has come up with a good enough mathematical description to derive the various alignments. Mr. Dickason gives the standard directions and specific driver recommendations. My research leads me to believe that woofer characteristics are not as critical for transmission lines as they are for other types of loading, and that any driver that is satisfactory for closed or vented boxes could probably be used with satisfactory results in a transmission line.

Chapter Five gives various ideas on the optimal shaping of rectangular cabinets and methods of enclosure damping. It includes detailed descriptions of two interesting alternatives to the traditional box-shaped enclosure.

The sixth chapter deals mainly with bandwidth and radiation pattern considerations for midrange and high frequency reproduction. This material, together with Colloms' section on midrange and high-frequency drivers in *High Performance Loudspeakers*, should give you a good feel for the problems in this field.

The seventh chapter covers crossover networks much more thoroughly than the first edition did. It includes a detailed description of the types and orders of networks that are available and the tradeoffs they represent. Mr. Dickason brings up the problem of driver-crossover interaction and mentions some methods of dealing with it, such as D'Appolito's (3,2) geometry. He also gives passive-network design formulas.

These formulas cover a variety of two-way networks for general crossover frequencies, but the three-way formulas are restricted to all-pass networks with two possible crossover frequency spreads. Mr. Dickason gives references for other cases.

The chapter also covers equalization and compensation techniques, passive component selection and inductor winding, a discussion of the advantages of active crossover networks, and a mention of computer-aided design.

The book ends with a short chapter on the measurement techniques that you can use to find the requisite parameters for low-frequency design and make system-impedance-magnitude and frequency-response plots.

The wide coverage and extensive bibliography of the *Loudspeaker Design Cookbook* make it an important reference for speaker designers of any level of expertise. For the inexperienced home builder, it is the best single reference available.

Robert M. Bullock III
*Contributing Editor*

# AN INTRODUCTION TO FREQUENCY RESPONSE AND LMP

## Part II: THE RESPONSE STEP

BY RALPH GONZALEZ

An ideal driver will exhibit a 6dB step (a rise which levels off at higher frequencies) in its response above the frequency where the wavelength equals the cabinet width.[1,2,3] The response will begin to rise at about ¼ this frequency, where the cabinet face begins to re-radiate an appreciable amount of the speaker's off-axis radiation. This phenomenon may be viewed as a 6dB step down in response at relatively low frequencies, in which case it is termed "diffraction loss."

The frequency whose wavelength equals your cabinet width (w in inches) may be calculated from:

$$f\ (\text{Hz}) = 13{,}500/w$$

If this frequency is below or more than two octaves above the range of frequencies in which a driver is being used, you may choose to ignore the effect of diffraction loss on this driver. Otherwise, mount the driver asymmetrically on the baffle, bevel the cabinet edges, or plan on off-axis listening to ensure a smooth response step.[3,4]

To include the response step of your driver in the LMP model, first determine the driver's sensitivity without aid of the response step. Since the manufacturer is likely to measure the driver's sensitivity near the highest SPL on its frequency response curve (given a 2.83V input and 1 meter microphone distance), it is helpful to find how the curve was obtained.

If (1) these measurements were made with the driver mounted in a (relatively) large baffle or in a wall, the step may have occurred at a very low-frequency and may not be visible as such. If (2) the driver was mounted in a small baffle or enclosure, the driver's frequency response may exhibit a rising trend due to the response step. Finally, if (3) the driver was mounted in an enclosure with a particularly small or narrow front baffle (relative to the wavelengths in the driver's frequency range, where wavelength (inches) = 13,500 / f (Hz)), the response step may not occur until a frequency near the driver's high-frequency rolloff point, and may serve to boost the driver's otherwise falling response here without affecting the remainder of the driver's frequency range.

In cases (1) and (2), assume an effective sensitivity 4–6dB lower than the driver's rated sensitivity. If you believe the driver was measured as in (3), you may not need to decrease its rated sensitivity in your LMP model. If you are uncertain about the effective sensitivity of your drivers, fine-tune your system during the final listening tests with L-pads on the individual drivers (see "Construction" below).

The response step would have an ultimate height of 6dB if your driver were a point source mounted on a fully-reflective baffle. You can, however, treat your baffle to reduced diffraction by covering the area around the driver with foam or soft rubber, or by restricting the tweeter's dispersion with a foam "donut."

In these cases, choose a higher frequency for the step in your LMP model (assume a narrower effective baffle width). On the other hand, if your driver is nearly as large as the baffle width—as is often the case with the woofer, the response step may be smoothed further. (I would be interested in any experimental or theoretical information *SB* readers might share regarding the effect of diffraction on the on-axis response of a piston mounted in the end of a tube, as opposed to a point source on a baffle.) Such a driver, however, is likely to be breaking up at the relevant frequencies, so the best approach may be to choose a step (and high frequency rolloff damping ratio) which produces a model appearing to match the manufacturer's curve near the "theoretical" step frequency.

**FOURTH-ORDER ROLLOFFS.** This article was written assuming most drivers can be modelled as having a second-order high-frequency rolloff. An exception is the SEAS 11F-GX woofer, discussed in "Executing LMP." This driver's rigid cone exhibits a rapid reduction in response above 6kHz. A fourth-order low-pass filter appears to provide a better model for its high-frequency rolloff. Since other drivers have this characteristic, I have included a prompt in LMP for a fourth-order high-frequency rolloff. If the rolloff is particularly jagged, however, you must judge for yourself whether an LMP model will be sufficient. (How closely can you duplicate the manufacturer's curve with LMP?)

If your driver appears to have an ultimate high-frequency rolloff slope closer to 24dB/octave than to 12dB/octave, you may wish to use the fourth-order model. LMP's fourth-order rolloff consists of identical, cascaded second-order filters. To find the corner frequency and damping ratio for the high-frequency rolloff of your LMP model, use the same procedure as before except double the magnitude scale of *Fig. 5* in Part I so that 10dB becomes 20dB, −10dB becomes −20dB, and so on. For example, if magnitude is −12dB at the crossover frequency, you

```
1.  NUMBER OF DRIVERS? 2

    DRIVER # 1 DRIVER INFORMATION

2.  CORNER FREQUENCY (HZ) OF DRIVER LOW-FREQUENCY ROLLOFF (OR 0)? 0
3.  CORNER FREQUENCY (HZ) OF DRIVER HIGH-FREQUENCY ROLLOFF (OR 0)? 3000
4.  HIGH-FREQUENCY ROLLOFF DAMPING RATIO? .7
5.  ORDER OF HIGH-FREQUENCY ROLLOFF (2 OR 4 ONLY)? 2
6.  POLARITY INVERSION (Y OR N)? N
7.  SENSITIVITY (DB)? 90
8.  DEPTH DISPLACEMENT (INCHES)? -1.5
9.  FREQUENCY OF RESPONSE STEP (OR 0)? 0

    DRIVER # 1 CROSSOVER INFORMATION
10  IDENTIFICATION NUMBER OF CROSSOVER (OR 0)? 0

    DRIVER # 2 DRIVER INFORMATION

11.  CORNER FREQUENCY (HZ) OF DRIVER LOW-FREQUENCY ROLLOFF (OR 0)? 1700
12.  LOW-FREQUENCY ROLLOFF DAMPING RATIO? .8
13.  CORNER FREQUENCY (HZ) OF DRIVER HIGH-FREQUENCY ROLLOFF (OR 0)? 18000
14.  HIGH-FREQUENCY ROLLOFF DAMPING RATIO? .6
15.  ORDER OF HIGH-FREQUENCY ROLLOFF (2 OR 4 ONLY)? 2
16.  POLARITY INVERSION (Y OR N)? N
17.  SENSITIVITY (DB)? 90
18.  DEPTH DISPLACEMENT (INCHES)? 0
19.  FREQUENCY OF RESPONSE STEP (OR 0)? 0

     DRIVER # 2 CROSSOVER INFORMATION
20  IDENTIFICATION NUMBER OF CROSSOVER (OR 0)? 6

21  VALUE OF COMPONENT K 0 (ENTER -1 WHEN DONE)? 7
22  VALUE OF COMPONENT K 1 (ENTER -1 WHEN DONE)? .25E-3
23  VALUE OF COMPONENT K 2 (ENTER -1 WHEN DONE)? 4E-6
24  VALUE OF COMPONENT K 3 (ENTER -1 WHEN DONE)? -1

    *****  GRAPH CHARACTERISTICS  *****

25  PAGE WIDTH < 80 >?
26  GRAPH OF MAGNITUDE (Y OR N) < Y >?
27  GRAPH OF PHASE ANGLE (Y OR N) < Y >?
28  LOWEST FREQUENCY OF GRAPH (HZ) < 20 >?
29  HIGHEST FREQUENCY OF GRAPH (HZ) < 20450 >?
30  NUMBER OF DIVISIONS PER OCTAVE < 3 >?
```

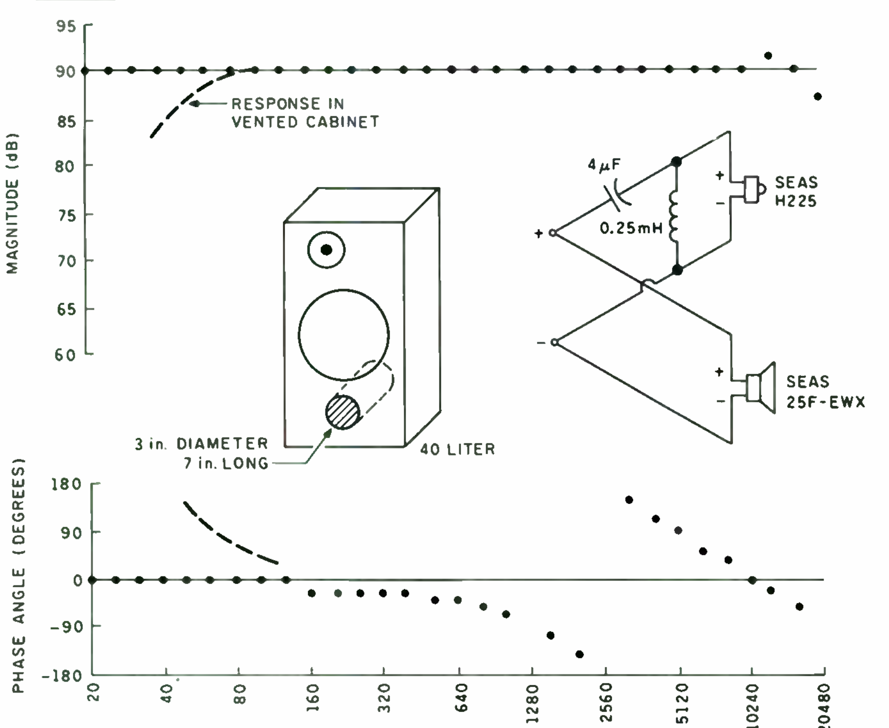


**FIGURE 1: (a) sample execution of LMP; (b) output of sample in Fig. 1a, crossover and cabinet design.**

may specify a damping ratio of 1.0 and choose the fourth-order rolloff. Notice the phase angle of this filter goes to −360° or 0° at high frequencies.

**MODELLING THE CROSSOVER.** LMP comes in three flavors (or versions), each providing a different way of entering the crossover model for your simulation. If you are familiar with circuit theory and can calculate your own transfer functions, use the LMP2 or LMP3 versions. LMP2 allows you to enter the information pertaining to the *completely factored* transfer function for each driver's crossover. You may also provide a more complex driver model by incorporating it into this transfer function. With LMP3, enter the coefficients of the numerator and denominator polynomials of the transfer function after optionally factoring out the gain and derivative factors. These two methods are equally suited to active crossovers.

LMP is the easiest version to use. First, choose a passive crossover configuration and calculate the element values by means of another source, such as Robert Bullock's 1985 *SB* series. LMP contains subroutines which calculate the transfer function of each driver's crossover for the first-, second-, third-, and fourth-order high, low, and bandpass parallel crossover sections shown in *Figs. 4–7.* For each driver, enter the identification number of the crossover section which matches the one you have chosen and the element values K0, K1, K2 and so on. I have defined K0 to be the driver impedance for the crossovers considered (see "Sensitivity and Impedance," *Part I*). Also, note all capacitor and inductor values should be in farads and henries, so a 4µF capacitor should be entered as:

.000004 (or: 4E−6)

If your crossover resembles one of those given, but is missing a capacitor or inductor, you can approximate the desired program inputs by using values of 0 farads and 100 henries for the missing capacitor or inductor, respectively. To replace the superfluous capacitor or inductor with a short circuit rather than an open circuit, use values of 1 farad and 0 henries, respectively, in your calculations. For example, if you want a midrange crossover with a second-order high-pass section and a first-order low-pass section, use the arrangement for the second-order bandpass filter (*Fig. 5c*), but set K1 = 0. Since formulas to calculate the values of K2, K3 and K4 will be difficult to find, use LMP to experiment with these values. As a starting point,

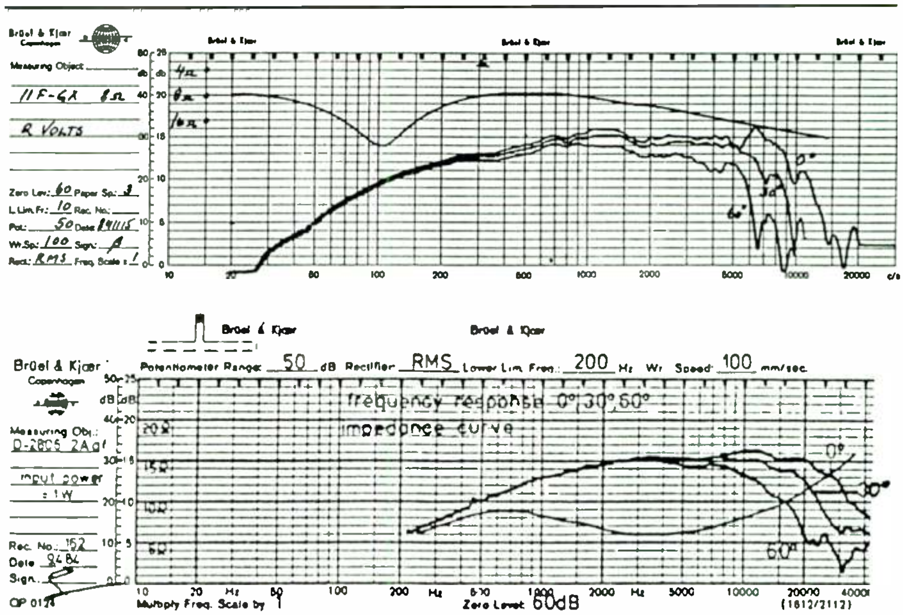


**FIGURE 2: (a) manufacturer's response curve for SEAS 11F-GX in 2.5 liter box, 0.5m distance; (b) manufacturer's response curve for Dynaudio D28af.**

calculate K2 as you would for a first-order low-pass crossover in a two-way system, and calculate K3 and K4 as you would for a second-order high-pass crossover.

If there is sufficient interest, I will derive subroutines for popular passive parallel crossover configurations other than those in *Figs. 3–7*, which may be appended to the program code at a later date. Knowledgeable readers can try calculating transfer functions of specialized crossover configurations they frequently use, and append a new subroutine, as described in REM statements in LMP's code. It should also be possible to create subroutines for popular active crossovers.

**EXECUTING LMP.** Upon execution, LMP will prompt you for information on each driver and its crossover section. The program will then plot magnitude and phase versus frequency curves for the combined system. If desired, repeat the prompting cycle and change any of the inputs (such as midrange sensitivity) to gain a more nearly flat magnitude response. (Remember, it is much more difficult to achieve a flat (0°) phase response through the crossover region, and very few commercial loudspeakers—even expensive ones—are designed this way.)

Sample executions of LMP are given in *Figs. 1* and *2*. I have numbered LMP's prompts to refer to them in the following discussion. These examples demonstrate how to use LMP to design loudspeaker systems; and should make interesting construction projects. I would be interested in any results and suggestions from *SB* readers.

The loudspeaker modelled in *Fig. 1a* is a two-way system (prompt 1) using the SEAS 25F-EWX 10″ woofer and the SEAS H225 ¾″ tweeter, whose curves are given in *Figs. 6a* and *6b* of *Part I*, respectively. Nominal power handling for the system is 60W and sensitivity is a fairly high 90dB/W/m.

Driver #1 is the woofer. I have chosen to ignore its low-frequency rolloff by specifying "0" to prompt 2. In practice, this woofer is well suited to a vented cabinet as shown in *Fig. 1b* (Reference #5 and Robert Bullock's article in *SB* 4/80). Clean bass down to 50Hz should be possible. The woofer has a smooth second-order (prompt 5) high-frequency rolloff at 3kHz (prompt 3) with a damping ratio of .7 (prompt 4) as discussed in "Modelling the Driver," *Part I*. Its diaphragm is about 1½″ (prompt 8) farther from the listener than that of the tweeter when both drivers are on a flat baffle and the enclosure is floor-mounted (see "Time Delay," *Part I* and the discussion on "lobing" in "Off-Axis Response," below).

Since the woofer is likely to be mounted in an enclosure not much wider than itself, and since the manufacturer's frequency response graph was also made in a small enclosure, I am ignoring the response step by entering "0" in prompt 9. To compare the driver model to the manufacturer's curve, specify "1" for "NUMBER OF DRIVERS," and choose crossover #0 (no crossover).

In the system considered here, I have employed the "Alternative Crossover"

```
1.  NUMBER OF DRIVERS? 2

    DRIVER # 1 DRIVER INFORMATION

2.  CORNER FREQUENCY (HZ) OF DRIVER LOW-FREQUENCY ROLLOFF (OR 0)? 110
3.  LOW-FREQUENCY ROLLOFF DAMPING RATIO? 1
4.  CORNER FREQUENCY (HZ) OF DRIVER HIGH-FREQUENCY ROLLOFF (OR 0)? 6500
5.  HIGH-FREQUENCY ROLLOFF DAMPING RATIO? .55
6.  ORDER OF HIGH-FREQUENCY ROLLOFF (2 OR 4 ONLY)? 4
7.  POLARITY INVERSION (Y OR N)? N
8.  SENSITIVITY (DB)? 80
9.  DEPTH DISPLACEMENT (INCHES)? .3
10. FREQUENCY OF RESPONSE STEP (OR 0)? 2250
11. HEIGHT OF RESPONSE STEP (1 TO 6DB)? 6

    DRIVER # 1 CROSSOVER INFORMATION

12. IDENTIFICATION NUMBER OF CROSSOVER (OR 0)? 1

13. VALUE OF COMPONENT K 0 (ENTER -1 WHEN DONE)? 8
14. VALUE OF COMPONENT K 1 (ENTER -1 WHEN DONE)? 17
15. VALUE OF COMPONENT K 2 (ENTER -1 WHEN DONE)? 1.2E-6
16. VALUE OF COMPONENT K 3 (ENTER -1 WHEN DONE)? .5E-3
17. VALUE OF COMPONENT K 4 (ENTER -1 WHEN DONE)? -1

    DRIVER # 2 DRIVER INFORMATION

18. CORNER FREQUENCY (HZ) OF DRIVER LOW-FREQUENCY ROLLOFF (OR 0)? 800
19. LOW-FREQUENCY ROLLOFF DAMPING RATIO? 1
20. CORNER FREQUENCY (HZ) OF DRIVER HIGH-FREQUENCY ROLLOFF (OR 0)? 20000
21. HIGH-FREQUENCY ROLLOFF DAMPING RATIO? .8
22. ORDER OF HIGH-FREQUENCY ROLLOFF (2 OR 4 ONLY)? 2
23. POLARITY INVERSION (Y OR N)? N
24. SENSITIVITY (DB)? 77.5
25. DEPTH DISPLACEMENT (INCHES)? 0
26. FREQUENCY OF RESPONSE STEP (OR 0)? 3400
27. HEIGHT OF RESPONSE STEP (1 TO 6DB)? 6

    DRIVER # 2 CROSSOVER INFORMATION

28. IDENTIFICATION NUMBER OF CROSSOVER (OR 0)? 3

29. VALUE OF COMPONENT K 0 (ENTER -1 WHEN DONE)? 14.5
30. VALUE OF COMPONENT K 1 (ENTER -1 WHEN DONE)? 7E-6
31. VALUE OF COMPONENT K 2 (ENTER -1 WHEN DONE)? -1

    *****  GRAPH CHARACTERISTICS  *****

32. PAGE WIDTH < 80 >?
33. GRAPH OF MAGNITUDE (Y OR N) < Y >?
34. GRAPH OF PHASE ANGLE (Y OR N) < Y >?
35. LOWEST FREQUENCY OF GRAPH (HZ) < 20 >?
36. HIGHEST FREQUENCY OF GRAPH (HZ) < 20450 >?
37. NUMBER OF DIVISIONS PER OCTAVE < 3 >?
```

**FIGURE 2c: Sample execution of LMP.**

discussed in *Part I* so the woofer needs no crossover. The construction is simple, and we eliminate a potential distortion source as well as any need for impedance compensation. Again, choose crossover #0 (prompt 10) to indicate that no crossover is used.

The tweeter has a smooth response up to 18kHz (prompt 13) with good dispersion and low distortion. While the tweeter's polarity should be inverted in an ideal second-order crossover, the additional phase shift introduced by the woofer's time delay and the tweeter's low-frequency rolloff, in this case, results in a much better combined response *without* polarity inversion (prompt 16).

Again we ignore the response step (prompt 19), this time because the manufacturer obtained his graph with the tweeter mounted in a large baffle (compared to the relevant frequencies), and we are also using the tweeter on a relatively large baffle. To reduce diffraction-induced irregularities, mount the tweeter asymmetrically on the baffle (*Fig. 1b*). This tweeter has a very smooth impedance curve, so impedance compensation should be unnecessary. As indicated in "Crossover Alternative," *Part I*, I recom-

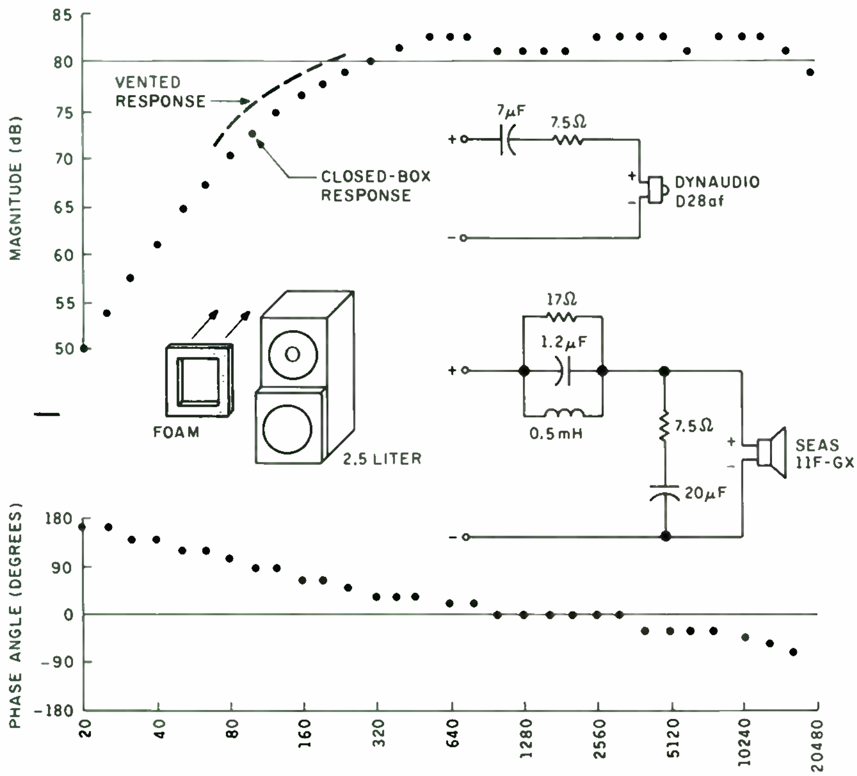


FIGURE 2d: Output of sample in Fig. 2c, crossover and cabinet design.

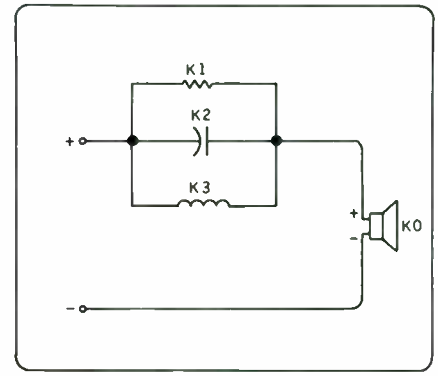


**FIGURE 3: Crossover #1: notch filter.**

mend a second-order high-pass crossover. After noting suggestions 4 and 5 in "Trial and Error," and running LMP several times, I arrived at a (non-Butterworth) 5kHz second-order crossover, with the element values of *Fig. 1b*. Choose crossover #6 (*Fig. 5b*) and the corresponding element values are supplied (prompts 21–23).

The resulting magnitude versus frequency response is surprisingly (and unusually) smooth, according to the LMP model (*Fig. 1b*). Having recently constructed a pair for my father, I know that the speakers are quite well-balanced and "powerful" sounding, and the woofer and tweeter combine convincingly. The sound varies little over the listening area in spite of the 10″ bass-midrange driver's directivity at high frequencies. The response also remains fairly smooth above and below the listening position, though if the enclosure is stand-mounted, a slight improvement may result from replacing the .25mH inductor with a .3mH one (the woofer's "Depth Displacement" is only −.75″ when listening on the system's axis).

The next example is much more compact, at the expense of bass response and sensitivity. I believe the SEAS 11F-GX woofer (*Fig. 2a*) is the same (though perhaps in a modified form) as that used in the British ProAc Tablette and (as the midrange in) the Thiel CS3—two systems of highly-acclaimed sound quality. The Dynaudio D28af tweeter (*Fig. 2b*) is also similar to those used in these speakers. The crossover is (approximately) of the minimum phase type (see "Frequency Response," Part I), which, while rarely used because of the demands it places on the drivers, offers theoretically better transient response than common passive crossovers. In addition to having a smooth response, the Dynaudio tweeter is one of the few available capable of withstanding high power levels with the chosen crossover.

The enclosure is exceptionally compact and unimposing, although you may need to mount it on a rigid stand (raising the speaker to ear height) to reduce colorations arising from the cabinet's moving in response to the woofer's excitation.

Placing the 11F-GX in a 2 liter sealed enclosure will raise the free-air resonance to 110Hz (prompt 2) and the total Q to 0.5. From "Modelling the Driver" in *Part I*, this Q corresponds to a damping ratio of 1 (prompt 3). After a few trials involving the woofer model alone without a crossover, I found a good match for the manufacturer's curve (which was obtained using a similar enclosure) required a 6dB response step (prompt 11) at 2.25kHz (6″ wide enclosure), and a fourth-order high frequency rolloff (prompt 6) with a damping ratio of .55 at 6.5kHz.

I am interpolating between the 0° and 30° off-axis curves, to approximate the 20° off-axis result of *not* "toeing in" the pair of speakers during stereo listening. A sensitivity of 80dB was chosen (prompt 8) since the 6dB response step contributed to the manufacturer's sensitivity measurement of 86dB (see "The Response Step," above). Note that the response step has been largely eliminated by the crossover arrangement.

The woofer is mounted on a ¾″ step on the front baffle. Since the woofer's diaphragm is recessed somewhat, it is only about .3″ (prompt 9) closer than the tweeter is to the listener. This .02mS time "lead" (see "Time Delay," Part I) reduces the interdriver phase shift and prevents a dip in the response around 3–6kHz. (It also causes a slight violation

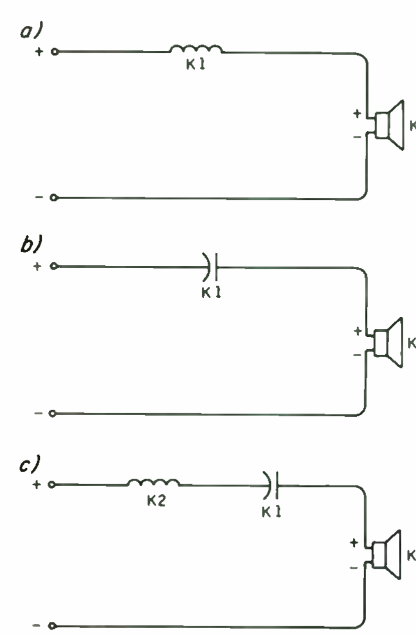


**FIGURE 4: Crossovers #2–4: (a) first-order low-pass, (b) high-pass, (c) band-pass.**

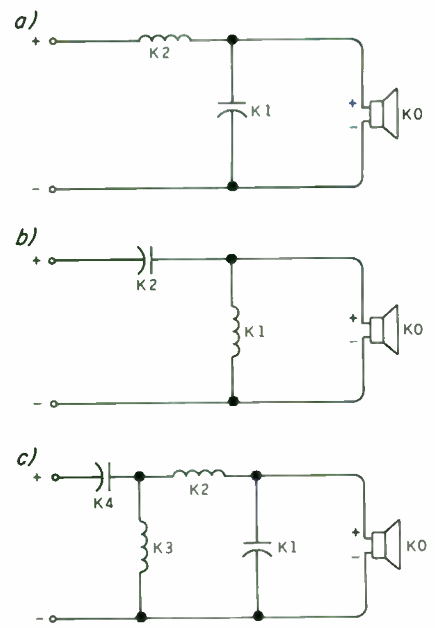


**FIGURE 5: Crossovers #5–7: (a) second-order low-pass, (b) high-pass, (c) band-pass.**

of the minimum phase condition.)

Crossover #1 (*Fig. 3*) is a notch filter. Here I used an overdamped notch at 6.5kHz to approximate a first-order roll-off at 1.6kHz. The driver impedance and the resistor, capacitor, and inductor values are entered in prompts 13–16. The woofer's impedance curve (shown inverted in *Fig. 2a*) is rising at 1.6kHz. To ensure the crossover works as modelled, we must employ impedance compensation. According to reference 5 of *Part I* (*SB* 1/87, p. 52), an extra resistor and capacitor is needed, as shown in *Fig. 2d*.

Assuming again that the listener will be 20° off-axis of the speakers, I used the manufacturer's curves of the D28af tweeter to get the values in prompts 18–22. The manufacturer probably measured the tweeter on a large baffle or wall so its sensitivity is about 84dB with a 6dB response step. If ¾″ absorptive foam is placed around the tweeter as in *Fig. 2d*, the enclosure will effectively be only about 4″ wide, and the response step should be placed at 3.4kHz (prompt 26).

I found a sensitivity of 77.5dB yielded the smoothest overall response. To reduce the sensitivity from 84dB to about 77.5dB (6.5dB), place a 7.5Ω resistor in series with the 8Ω tweeter, to yield a total 14.5Ω impedance (see "Sensitivity and Impedance," Part I). A first-order high pass crossover (prompt 28) with a 14.5Ω driver and a desired crossover frequency of 1.6kHz requires a 7μF capacitor, according to the standard formula. Referring to *Fig. 4b*, these values are entered at prompts 29 and 30.

The system will probably not satisfy people with a taste for *loud* music. Sensitivity (82dB/W/m), power-handling (40W), and bass response are limited. You could raise the maximum output level by 6dB by wiring two closely-matched woofers in series or parallel and correcting the crossover for the change in impedance and (for the parallel arrangement) in optimum tweeter sensitivity. Bass response could be extended somewhat by porting; for example: box volume = 1.5 liter, vent = 1″ dia. by 5-7″ long. You could also improve power handling and bass by experimenting with a slightly larger woofer, such as the Focal 5N401 or the SEAS P13RCY. Another alternative would be to add a subwoofer. In order to preserve the minimum-phase relationship, you could place a separate subwoofer under each satellite, with a first-order crossover around 500Hz. Finally, please note that the system described in this example has not been constructed, and must be considered experimental.

**TRIAL AND ERROR**. LMP's main advantage as a modelling program is experimentation with different drivers, crossovers, and cabinets before you actually purchase components or build an enclosure, and also without making

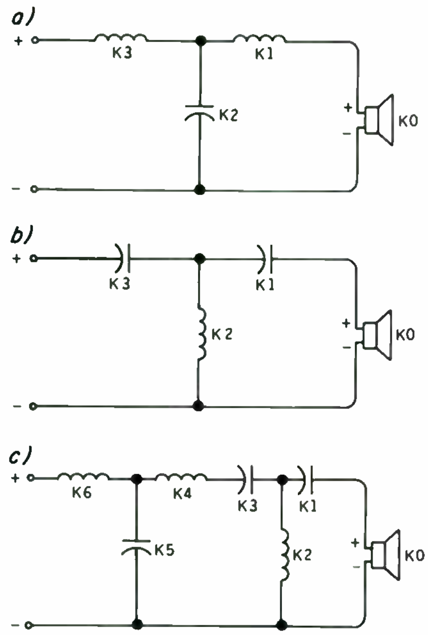


**FIGURE 6: Crossovers #8–10: (a) third-order low-pass, (b) high-pass, (c) band-pass.**

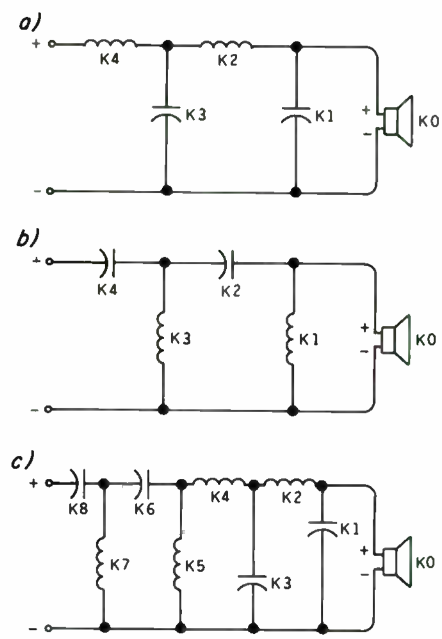


**FIGURE 7: Crossovers #11–13: (a) fourth-order low-pass, (b) high-pass, (c) band-pass.**

repeated measurements.

The following suggestions may help you obtain a smooth crossover from one driver to another. Make it your goal to get the phase angles of the driver/crossover sections within 90° of each other at the crossover frequency. (Specify "NUMBER OF DRIVERS" = 1 *or* temporarily reduce the "SENSITIVITY" of the other drivers by 60dB in order to plot an individual driver/crossover section.)

1. Invert the polarity of one driver.
2. Introduce a time delay or time "lead" to one driver relative to the other.
3. Use an asymmetric electrical crossover, such as those discussed in "Crossover Alternative," in *Part I.*
4. Use non-coincident crossover points for the drivers.
5. Use a high-pass or low-pass *second-order* filter with a lower damping ratio than the conventional Butterworth or Linkwitz-Riley filters. Try simultaneously increasing the value of the capacitor and decreasing the value of the inductor by a factor of 1.5.
6. Use a *notch filter* (*Fig. 3*) in place of a *second-order* crossover to *increase* the damping ratio (reduce the resonant peak) of a woofer's high-frequency rolloff (see "Crossover Alternative," in Part I). In this case, let $f_c$ and $d_w$ be the corner frequency and damping ratio of your woofer's high-frequency rolloff, and let dd be the desired damping ratio (1.0 for a Linkwitz-Riley crossover and 0.7 for a Butterworth [constant-power in this case reference 6, *Part I*] crossover). The values of resistor (K1), inductor (K3), and capacitor (K2) in *Fig. 3* are:

$$K1 = (dd - dw) \times K0 / dw$$

$$K3 = (dd - dw) \times K0 / (3.14 \times fc)$$

$$K2 = 1 / (40 \times K3 \times fc^2).$$

**OFF-AXIS RESPONSE.** The magnitude versus frequency response curve produced by LMP corresponds to a loudspeaker's "direct" or "anechoic" response. Many believe this should be as flat as possible for natural reproduction, as indicated by the popularity of "All-Pass Crossovers" (reference 6, *Part I*). You can, however, also attempt to smooth out the *reverberant response*: the sound reaching your ears only after having reflected from the walls and furniture.

Suppose you have a two-way system whose drivers have excellent dispersion at the crossover frequency, and are spaced so the vertical distance between their centers is greater than one half the wavelength of sound at the crossover frequency. (For a 2kHz crossover, the drivers would be more than 3″ apart—which is usually the case.) Even if your on-axis response is flat, you may have a peak or a dip in the reverberant response at the crossover frequency.

Use LMP to graph the *individual* driver/crossover sections in your completed system (temporarily reduce the sensitivity of the other drivers by 60dB). Now find the "crossover frequency:" that where both driver/crossover sections have the same magnitude (D). Let M (dB) = this magnitude relative to the overall sensitivity of the system (S); i.e., M=D−S. Then, the reverberant response at the crossover frequency will be about R=3dB + M, relative to the reverberant response away from the crossover frequency (assuming, again, that the centers of the drivers are separated by at least one half the wavelength of sound at the crossover frequency).

Notice that if the relative phase shift between the driver/crossover sections at the crossover frequency is 90° (as in an ideal odd-order crossover), then M= −3dB is required for a flat on-axis response, and a flat reverberant response will result. If the relative phase shift is 0° (as in an ideal even-order crossover), M= −6dB produces a flat on-axis response, and you will see a mild dip in the reverberant response.

In practice, the significance of these conclusions depends on the actual dispersion of your drivers at the crossover frequency, the size of your listening room, and the amount and quality of its acoustic damping materials (rugs, curtains, and the like), as well as your listening tastes.

*Lobing* is a related problem: the changes in frequency response when listening at points above and below the system's axis, due to driver interference near the crossover frequency (assuming the drivers are more or less vertically aligned on the baffle). Here, crossovers which produce 0° of interdriver phase shift have the advantage of symmetric lobing. You may use LMP to evaluate the response above or below the system's axis by calculating the additional interdriver depth displacement resulting from listening at the new position.

Reverberant response and lobing problems can be reduced by placing the drivers closer together on the baffle and/or choosing a lower crossover frequency. If, however, you are using the drivers at frequencies where their own dispersion is poor, these problems become more complicated. Try substituting the manufacturer's *off-axis* curves when evaluating vertical and horizontal lobing with LMP.

Finally, note that above the "response step," frequency, the system radiates in the "forward" hemisphere only. If you have equalized out the on-axis response step, the reverberant response may show a step *down* at this frequency. One way to change the frequency balance of the reverberant response is to add a rear-firing midrange or tweeter to your system.

**CONSTRUCTION**. Pay close attention to several details when constructing your loudspeaker system to ensure it will behave as its model predicted. Select quality crossover components, control cabinet resonances and diffraction effects, and check bass loading and room interactions to ensure a smooth, musical response.

Martin Colloms' *High Performance Loudspeakers*[4] is an excellent source of information on these problems. Also, many of the construction articles in *Speaker Builder* offer valuable suggestions and guidelines. (See References)

Finally, since no modelling program can guarantee perfect results, if at all possible you should measure the frequency response of your completed loudspeaker in an anechoic chamber or outdoors with a good microphone. As a minimal approach, use an inexpensive stereo test record and/or a graphic equalizer and your ears to locate peaks or dips in the frequency response. Fine-tune the completed loudspeaker by adjusting midrange and tweeter levels with "L-pads."

If it does nothing else, LMP will make you aware of the importance of driver characteristics in crossover design, and will help you decide whether you can improve upon the results of using "cookbook" crossover design formulas.

I am currently using LMP to design a minimum phase crossover for a compact three-way Dynaudio/SEAS system which I hope to describe in another article.

*Next time we will publish a step-by-step project illustrating use of the Author's LMP software.—Ed.*


## REFERENCES

1. Muller, G.; Black, R.; and Davis, T., "The Diffraction Produced by Cylindrical and Cubical Obstacles and by Circular and Square Plates," *JASA*, Vol. 10, No. 1 (1938), pp. 6-13.
2. D'Appolito, Joseph, "A High Power Satellite Speaker," *SB* 4/84, pp. 7-14.
3. Kral, Robert, "Diffraction–The True Story," *SB* 1/80, pp. 28-33.
4. Colloms, Martin, *High Performance Loudspeakers*, 3rd Ed., Halstead Press (John Wiley and Sons), 1985.
5. Weems, David, *Building Speaker Enclosures*, TAB Books, 1981.

## FURTHER REFERENCES

Boulais, Michael, "The Dynaudio: An Inexpensive Two-Way with Great Sound," *SB* 1/83.

Linkwitz, Siegfried, "A Three-Enclosure Loudspeaker System," *SB* 2, 3, 4/80.

Olson, Harry, *Acoustical Engineering*, D. Van Nostrand Co., 1957, pp. 17-26.

Morse, Phillip, *Vibration and Sound*, McGraw-Hill, 1948.

Hayt, W.; and Kemmerly, J., *Engineering Circuit Analysis*, McGraw-Hill, 1978.


**HOW TO GET COPIES OF LMP:** The software is available, at $12.50 per copy, from Old Colony Sound Lab, (P.O. Box 576, Peterborough, NH 03458) in three versions. The price includes author support via mail at P.O. Box 54, Dover, DE 19711.

1. Apple II: Applesoft BASIC, DOS 3.3; (5¼" SS/DD) Part No. CSK-C1.

2. Apple Macintosh: Microsoft BASIC; (5¼" SS/DD) Part No. CSK-C2.

3. IBM PC/XT/AT and compatibles (Microsoft) BASIC DOS 2.1 or 3.1 (5¼" DS/DD; 360K) Part No. CSK-C3.

## MODIFICATIONS

These are a few areas of LMP which are open to future improvement.

1. The method of Nodal Analysis has the potential of allowing LMP to accept arbitrary crossovers without need for additional subroutines; however, it may slow processing significantly and may produce complications in the context of LMP. An example can be found in "Design of Optimized Loudspeaker Crossover Networks Using a Personal Computer", P. Schusk, *JAES*, March 1986.

2. With LMP you must rely on impedance compensation circuits to avoid problems from the complex impedance of real-world drivers. If you are interested in developing LMP to model this varying impedance directly, I can provide suggestions.

3. If you wish to rewrite LMP for another computer, send me 5 stamps for a hardcopy of the code and suggestions for distributing the new version.

Please inform me of any such modifications to LMP so that significant improvements may be made available to *SB* readers. Send a SASE to Ralph Gonzalez, PO Box 54, Newark, DE 19711

# Kit Report
## Audio Control's Richter Scale

Unit Tested: Richter Scale, Series Three, Serial #P12111.

Manufacturer: Audio Control, 6520 212th St., SW, Lynnwood, WA 98036, (206) 775-8461. List Price: $349.

The Audio Control Richter Scale is, in my opinion, one of the most interesting and useful audio devices I have seen recently. It combines a two-way electronic crossover with a highly effective low-frequency equalizer and test system. The crossover uses a fourth order (24dB/octave) Linkwitz-Riley characteristic with such well known features as in-phase outputs at the crossover frequency, flat magnitude response through the crossover region, and no off-axis peaks in the radiation pattern.

Although loudspeaker designers still debate which type of crossover characteristic is best, (an issue which will probably never be resolved), I favor the Linkwitz-Riley approach. I am also a die-hard advocate of multi-amplification because it overcomes so many limitations inherent in passive crossover designs. Passive crossovers versus multi-amplification is another issue which still generates controversy. In my experience, however, those who refuse to acknowledge the superiority of bi- or tri-amplified systems are those who have not tried them.

The Richter Scale (the unit I tested was labeled Series Three, implying revisions on two previous models) has a 90Hz crossover frequency, but you can change it to many other locations. A plug-in, eight resistor DIP module determines the crossover frequency, while a manual chart gives resistor values for eleven other crossover frequencies ranging from 72–212Hz. A formula is given so you can determine resistor values for other frequencies. Since all eight resistors are the same value, the calculations are simple.

The crossover is only half the complete package. The Richter Scale also contains a six-band graphic equalizer to correct the subwoofer/room interface's response problems. The equalizer contains controls centered at 22.5, 31.5, 45, 63, 90 and 125Hz (these are ½-octave spacings) and a built-in analyzer to make an adjustment of ± 12dB. Also included is a warble-tone generator with a continuously variable center frequency from 22.5–250Hz. To make analyses, you can use the built-in metering system and supplied electret condenser microphone. The microphone appears to be the Panasonic P9932 sold by Digi-Key, although Audio Control provides a nice housing for it.

The Richter Scale also contains a few other features I found useful, including an 18dB/octave infrasonic filter, (although Audio Control has mislabeled this as "subsonic," technically, *infrasonic* information is below the frequency range of human hearing, whereas *subsonic* refers to anything which travels slower than the speed of sound). I believe an electronic crossover is a logical place to put an infrasonic filter, and I have done so in my own crossover designs. In addition, the Richter Scale has a rumble reduction circuit, which reduces the effects of turntable rumble and acoustic feedback in the audible range.

This circuit is bound to have an effect on music containing low frequency information, so its use will probably be limited to extreme cases. In my opinion, the audiophile serious enough about sound reproduction to go to the trouble of bi-amplifying a system should have solved turntable rumble and acoustic feedback problems beforehand. This control could be useful if you play a recording with an excessive amount of low-frequency noise. Both the infrasonic filter and the rumble reducer are switchable.

Finally, the Richter Scale has a mono bass output if you have a common subwoofer, while an inverted mono output is provided so you can bridge a stereo power amplifier for mono operation. These mono outputs are in addition to the stereo bass outputs. Gold-plated RCA connectors are used throughout.

Most circuitry is contained on a single, large, printed circuit board. No schematic diagram was provided, so I looked inside to see the basic circuit components. The crossover circuitry for each channel consists of a 4136 quad op amp. Film-type capacitors and 1% metal film resistors are used in the crossover circuits. Input and high frequency output buffering appear to be accomplished with 4560 dual op amps. An additional 4136 is used for low-frequency output buffering on all four low outputs (left and right stereo, mono and mono inverted). Unby-passed electrolytic capacitors are used for output coupling.

The equalizer utilizes three 4136 op amps. Four additional 4560 op amps appear to be related to the infrasonic filter, rumble reducer and analyzer circuits. The warble tone generator seems to be designed around an LM13600N dual transconductance amplifier chip. Four discrete

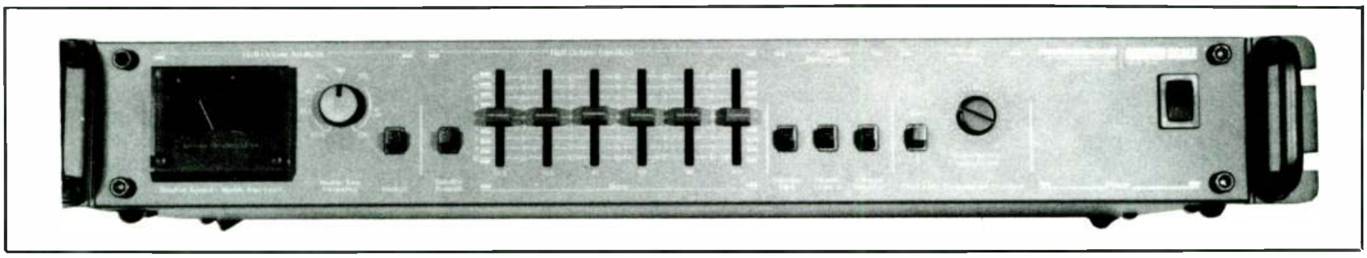

**PHOTO 1: Front panel of the Richter Scale. From left to right are the analyzer's meter, equalizer and low-frequency level control.**

transistors and an additional 4136 op amp appear to be related to the microphone preamplifier and microphone phantom powering. All resistors, other than those used in the crossover's frequency determining elements, are ±5% carbon film.

I was disappointed by Audio Control's choice of op amps. The 4136 chips were only medium performance devices when Walt Jung wrote his landmark series on slewing-induced distortion in the 1977 issues of *Audio Amateur*. In recent years they have been outclassed by many other quad devices, the most obvious example being the Texas Instruments TL-074. When I phoned Audio Control to ask why they used the 4136, I was told they do not favor BIFET input devices (such as the TL-074) because of their higher noise level. While it is true BIFET devices do have higher noise levels, this is of little consequence at line-level applications.

I have designed electronic crossovers using the TL-070 series devices and have achieved signal-to-noise (S/N) ratios of 96dB or higher, relative to a 1V output. The 4560 op amps are an improved version of the mediocre 4558 (essentially, a dual 741), but are no match for more recent dual op amps, such as the Signetics 5532. The 4560 devices are pin-out compatible with standard dual op amps, so modifications are possible. Unfortunately, the 4136 uses a non-standard pin configuration, so you cannot directly substitute a TL-074. Any modifications will, of course, void the warranty (a generous 5 years), and leave servicing in your hands. Since no schematic is provided with the unit, this could lead to problems.

I was surprised by the Richter Scale's power supply. It is an unregulated, dual ±15V type, each side consisting of (after the rectifiers) two 470μF caps and a 22Ω resistor in a Pi (π) configuration. In addition, ceramic capacitors are used for all supply bypassing. When I asked Audio Control why they used an unregulated supply, they told me they believe regulation is unnecessary due to the op amps' excellent common mode rejection. While it is true the unit hasn't a trace of 60Hz hum at its outputs, elimination of hum is not the real issue. Walt Jung and others have shown the importance of low-impedance supply regulation and its effect on the dynamic performance of audio circuits. I remain convinced of the audible benefits of well-regulated power supplies.

The company has an obvious contradiction in design philosophies. On one hand, Audio Control has provided gold-plated connectors, presumably believing contacts do affect the sound. On the other hand, they see no need to regulate the power supply, and use op amps which are not up to the performance level of the better devices available today. I believe contacts affect the sound, but in my experience, using gold-plated connectors is like applying "icing on the cake;" it is something you add as the finishing touch,

**PHOTO 2: Rear panel, showing three groups of RCA interconnect jacks. From left to right are crossover inputs and outputs, new tape monitor loop and equalizer/analyzer connections to the preamp's tape monitor loop.**

when the circuitry and power supply are as close to ideal as possible.

The Richter Scale is extremely well-constructed and the layout is clean and uncluttered. The front and rear panels are shown in *Photos 1* and *2*. An inside top view is shown in *Photo 3*.

The owner's manual is skimpy and short on essential information. The instructions and diagrams that show you how to connect the device to your system are clear and concise, but the analysis and equalization procedure lacks instruction for setting the woofer to satellite level (the Richter Scale contains a low-frequency output level control for this, but the manual gives you no clue as to how to achieve the correct level). In my opinion this is a serious omission, since without correct woofer-to-satellite balance, your system will never benefit from correct woofer equalization. I have developed a procedure for doing this, however, and will explain it later.

## Bench Tests

My review unit exhibited a noise problem in the right channel high output (an excessive amount of high frequency "hiss"). I telephoned Audio Control and was told the fault was probably in the right channel's high output buffer op amp. Though they promptly sent me a replacement 4560 chip, it did not solve the problem. Further investigation (without any schematic) revealed a noisy section of the 4136 op amp in the right channel's crossover. Since I had some 4136 op amps in my shop, and did not wish to delay this review any further, I replaced the bad 4136 with one of my own, an Exar, the same brand used by Audio Control. Performance was therefore not degraded by lack of a factory-supplied part.

With the noise problem solved, I made some measurements. The specs included in the owner's manual are, for the sake of comparison, shown in *Fig. 1*. At 1kHz, I measured clipping at 8.5V RMS output. The Richter Scale, when driven into hard clipping (8.9V) inverts the waveform's polarity and produces the pulse-type distortion shown in *Fig. 2*.

This polarity inversion and pulse distortion is identical to that produced by the Phoenix Systems Parametric Equalizer, which I reviewed in *Audio Amateur*, 2/86. At 20kHz the left channel clipped at 7.6V and the right at 8.2V. I observed the same distortion and polarity reversal at this frequency. In his reply to my review of the Phoenix Parametric Equalizer, designer John Roberts noted this is a full supply pulse which occurs in the presence of a hard clipped rail-to-rail signal.

S/N ratio referenced to 7V RMS output (which is specified as the maximum output level) measured 106dB in the left channel and 104.5dB in the right. This is slightly worse than Audio Control's published specifications. The low-frequency outputs showed a level imbalance. The left chan-

**RICHTER SCALE SERIES III SPECIFICATIONS**

| | |
|---|---|
| Total Harmonic Distortion | 0.005% |
| Frequency Response | 18–100kHz |
| Input Impedance | 100kΩ |
| Output Impedance | 150Ω |
| Signal to noise Ratio (rated full output) | -120dB |
| Meter Range | -20dB to +6dB |
| Maximum Output Level | 7V RMS |
| Crossover Slope | 24dB/octave |
| Crossover Frequency | Programmable* |
| Subsonic Filter | 18dB/octave @ 20Hz |
| Size | 17" x 2.5" x 8.25" |
| Warranty | 5 years |
| Country of Origin | USA |

*While the crossover comes pre-set at 90Hz, it can be programmed for almost any frequency from 20–20kHz.

**FIGURE 1: Manufacturer's specifications.**

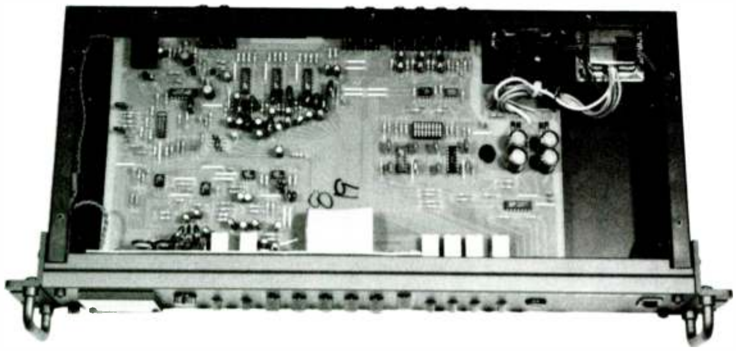

**PHOTO 3: Inside top view showing the large circuit board which contains most of the circuitry.**

nel low output was 1dB below the right at 20Hz and 90Hz, undoubtedly due to mistracking between the low-frequency output level control's two sections.

I measured total harmonic distortion at an input level of 1V. At 1kHz and 20kHz, the distortion was below the residual of my test set-up (0.015%). At 20Hz, the distortion was also below the test set-up residual level (0.03%). Both channels produced the same results and, in all cases I measured, the distortion products consisted of only second and third harmonics plus noise. I found no high-order distortion products, which is most desirable. I have no reason to doubt the manufacturer's specifications and my own measurements reveal no cause for concern.

Slew rate measured 2.2V/$\mu$Sec, based on a 10kHz square wave I used at the unit's maximum output. According to Walt Jung's criteria, with which I wholeheartedly agree, this is an inadequate figure. Readers unfamiliar with slew rate and its effects on sound should read Walt's series in *TAA* 1-4/77. These articles have become classics and are required reading for anyone serious about audio design. Frequency response measurements showed 70kHz and 100kHz to be 1dB and 2.5dB down, respectively. On the low end, the 1dB down point was 3Hz, and I traced the slew rate limitation to the 4560 buffer chips. I find it is strange that Audio Control chose a 100kHz bandwidth combined with low slew rate op amps.

The mislabeled infrasonic filter was down 3dB at 20Hz and rolled off at 18dB/octave. Audio Control specifies the rolloff rate, but they do not specify the frequency at which the filter takes effect. nor do they give any information in their manual on the rumble reducer characteristics. I measured the rumble reducer's filtering characteristics and have plotted the results in *Fig. 3*. It is a rather severe filter with a 3dB down point of 170Hz. A maximum reduction of 8.25dB occurs between 80 and 90Hz, tapering off to a 7dB reduction at 20Hz.

I found it interesting that the rumble reducer is contained in the Richter Scale's equalizer section while the infrasonic filter is contained in the crossover's low-frequency section. When I first attempted to measure its effect, I had difficulty observing any effect from the rumble reducer. Normally, when I test both channels of stereo gear, I parallel my signal generator to both inputs with a "Y" adapter. I discovered, however, that the

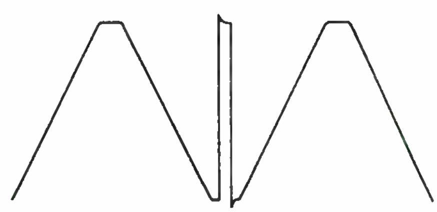

**FIGURE 2: A pulse-type distortion plus a polarity reversal occurs when the crossover is driven into hard clipping.**

rumble reducer has no effect with the left and right equalizer inputs paralleled. Without a schematic, I am at a loss to explain why this happens. It is extremely unlikely that this would cause problems during normal operation, but I thought it was worth noting.

## Installation/Setup

Installing the Richter Scale is simple and straightforward. Your preamp's main outputs feed the crossover's inputs, and the crossover's low and high outputs feed your woofer and satellite amplifier's inputs. The unit's equalizer/analyzer section connects into your preamp's tape monitor loop, so you must actuate your tape monitor button in order to operate the equalizer. Audio Control supplies a duplicate tape monitor loop so you can continue to operate your tape deck.

I found the setup procedure left much to be desired. The steps for setting up the equalizer are fine, but, as I mentioned previously, you are not given even a hint on how to adjust the woofer-to-satellite balance. In fact, the low-frequency level control isn't even mentioned in the setup instructions. This is a serious omission.

Through experimentation, I have developed a procedure for setting the woofer/satellite balance that works consistently well. You must make this adjustment *before* you adjust the equalizer (be sure the equalizer controls are set flat). Make sure the microphone is on a 3′ stand centered between the loudspeakers and only two to three feet back. You do not want to place the microphone in the listening area, since the woofer/satellite balance is not used to compensate for room acoustics. Only one setting is correct, but you can achieve it in two ways.

## Procedure A

Set the warble tone generator to 250Hz and adjust your system level so the meter reads 0dB. Now, move the warble tone generator setting to the crossover frequency (in this case, 90Hz) and adjust the low frequency output level control so the meter also reads 0dB. This will normally require you to rotate the low-frequency output level to get a gain increase (i.e., clockwise).

## Procedure B

Set the warble tone generator to the crossover frequency (in this case, 90Hz). Mute the low output (button next to the level control) and adjust your system level so the meter reads −6dB. Reactivate the low

*Continued on page 54*

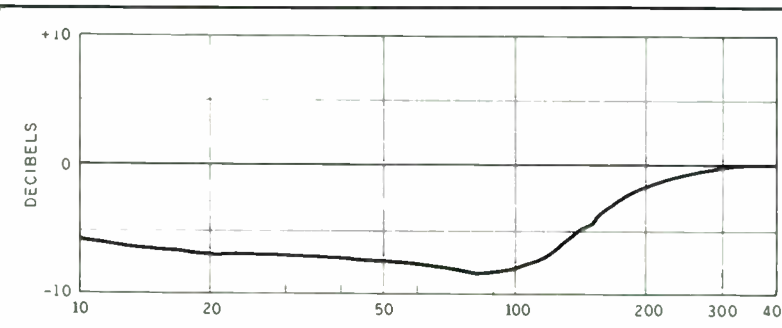


**FIGURE 3: Response characteristic of the rumble reducer.**

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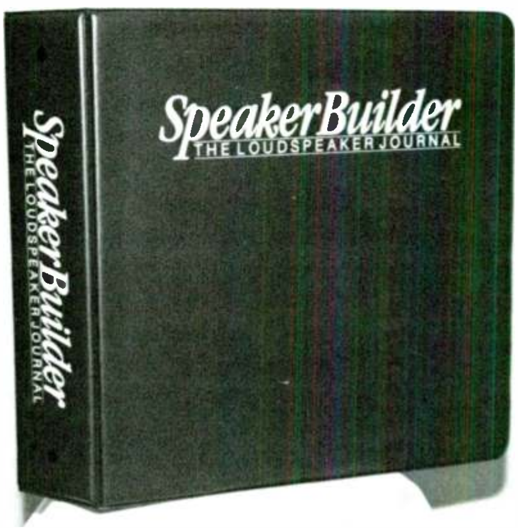


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## Kit Report

*Richter Scale continued from page 52*

output by pressing the mute button again (it will now be in its outermost position). Adjust the low-frequency level control so the level increases 6dB, or a reading of 0dB.

I recommend trying both procedures. When you find the one microphone position that yields the same result with both procedures, you have achieved the correct setting. I hope Audio Control will include this information in their manual, provided they credit this source. I think it is worth noting that Procedure "B" seems to work well even with the microphone placed in the listening area, while Procedure "A" most certainly does not.

If you try Procedure "A" with the mike in the listening area, you will wind up with a greatly reduced woofer level. In view of all this, I recommend you position the mike close to the loudspeakers (as described above). Before I developed this procedure, I set the woofer level "by ear," yet setting the woofer level using the above method produced a virtually identical result. My procedure, therefore, should produce musically accurate results and be technically valid.

Once you find the correct woofer/satellite level, you can turn to page 5 in the manual to learn how to set the equalizer. If you are using a 90Hz crossover frequency, I suggest you set the 125Hz control while the microphone is still close to the loudspeakers. Then, move the microphone to the listening position, set the warble tone generator to 90Hz, and adjust the system level so the meter reads 0dB. Set the warble tone generator to 63Hz, and adjust the 63Hz control to achieve *half* the needed boost or cut. Repeat the procedure for 45, 31.5 and 22.5Hz. Since the equalizer's controls interact, you will probably need to repeat the procedure. Always go back to the crossover frequency, remembering that you set the control(s) above the crossover frequency (in this case, 125Hz) with the up-close microphone position. This is my own suggestion, however, and is not mentioned in the manual.

Audio Control gives appropriate warnings about setting the 31.5 and 22.5Hz controls with smaller woofers. If you don't heed them, you could destroy small woofers by feeding them more low-frequency power than they can safely handle. The response should be within ±1dB down to 22.5Hz. If your woofers cannot safely handle the extremely low frequencies, you can only achieve flat response down to 45Hz.

I conducted my initial tests using the woofers in the latest version of my TL-10 transmission line (TL) loudspeakers (*SB* 1,2/82) as subwoofers and my TL-5 loudspeakers as satellites. My woofer amplifiers are a pair of modified Southwest Technical Products Tiger .01s, while I used my Pass A-40 amplifier to power the TL-5 satellites (*TAA* 4/78). This superb, pure Class-A amplifier normally powers my tweeters in my tri-amplified system.

When I first set up the equalizer, my TL-10 woofers measured flat ±1dB, without any equalization, from 90 down to 22.5Hz. I thought the analyzer must be extremely accurate since it showed my own woofer design was nearly perfect. All kidding aside, I realized I would need a pair of woofers with more modest capabilities to evaluate the equalizer/analyzer. Therefore I chose a small 8″ acoustic suspension woofer system I had built.

With the Richter Scale it was easy to equalize my small 8″ woofers; once you have a procedure for dealing with both the woofer/satellite balance and the equalization, the process is relatively simple. After working with the Richter Scale's equalizer/analyzer for a few days, I have concluded that it is the easiest to use and most effective low frequency room equalization system I have ever encountered. The easy use, stability, accuracy and warble tone system repeatability beat any ⅓-octave pink noise system I have encountered.

### The Sound

With the Richter Scale, my 8″ woofer system improved dramatically. The small acoustic suspension woofer actually delivered some of the weight and impact you normally associate with much larger woofer systems. It minimizes woofer and room response irregularities in the upper bass region to give a more open and natural quality to the music.

One thing the Richter Scale (or any other bass equalization device, for that matter) cannot do is eliminate colorations inherent in your enclosure design. If you admire the open un-boxed sound of a TL system as much as I do, understand that equalizing an 8″ acoustic suspension system won't turn it into my TL-10. Though it will restore the missing fundamentals, it will still sound like an acoustic suspension loudspeaker, albeit a larger one.

The Linkwitz-Riley crossover produces a stable radiation pattern, and makes relative subwoofer and satellite positioning far less critical than would be the case if the company had used an odd order 18dB/octave characteristic. I applaud Audio Control's choice of crossover characteristic.

As for the sound, I have two criticisms. First, the upper midrange and high frequencies have a slight edginess, which is most apparent on massed orchestral strings. Second, when compared to my own Linkwitz-Riley crossover, the Richter Scale has less dynamic impact. I hope Audio Control will consider using more up-to-date op amps in their next version of the Richter Scale. I would also like to see a regulated power supply and polypropylene bypasses on all electrolytic coupling capacitors. An unregulated power supply is inexcusable in a piece of audiophile gear.

If you purchase the Richter Scale, you can modify it (voiding the warranty) in accordance with my suggestions. The fly in the ointment, however, is the 4136 quad op amps' non-standard pin configuration. I'm sure the folks at Audio Control must realize *Speaker Builder* and *Audio Amateur* readers are unlikely to leave any piece of equipment in stock form if they think there is room for improvement. Any manufacturer selling products to *SB* or *TAA* readers should include a schematic diagram with their manual. We are not appliance operators!

In addition to my above mentioned suggestions, I wish Audio Control would consider two possibilities. First, I would like to see the Richter Scale equalizer/analyzer section offered as a separate unit. I'm sure many *SB* readers who do not need the crossover section would find this extremely attractive. Second, I would like the manufacturer to consider marketing both the Richter Scale and my suggested equalizer/analyzer in kit form. Even if the kits were not of the "ground-up" variety, such as those offered by Heath (this possibility scares many manufacturers), a Hafler-type approach, with factory assembled circuit boards, would suffice. I think this would greatly enhance Audio Control's product appeal to *SB* and *TAA* readers.

To summarize, despite my reservations, I recommend the Richter Scale and hope Audio Control will consider my suggestions. I must repeat, in my assessment the equalizer/analyzer is simply the best method I've seen for woofer/room equalization.

Gary A. Galo
*Contributing Editor*
*Photographs by the author*

*Manufacturer's Comments:*

*We'd like to add just a few comments to Mr. Galo's review. We consider the Richter Scale's crossover not just a subwoofer type. We agree fully with his comments on the advantages of multi-amping and are about to publish a technical paper on the subject. While Mr. Galo's guess about the microphone is near the mark, the unit we use is actually a Matsushita WM063T.* [Panasonic is one of the trade names of Matsushita—Ed.] *Although Mr. Galo is properly critical of the original 4136 op amp, we believe the current production of this chip has improved dramatically in terms of noise, slew rate and consistency since 1977. While Mr. Galo points out that the TL-074 is not a pin-compatible replacement for the 4136, the TL-075 is.*

*Our rumble-reduction circuitry sums the left and right channels in the low bass frequencies to cancel lateral out-of-phase rumble. Mr. Galo is incorrect in assuming the infrasonic filter is contained in the crossover circuitry. It is, in fact, in the equalizer section. The crossover is completely independent.*

*We like Mr. Galo's suggestion made under Procedure A. In fact, we will include it in fugure issues of the manual, and we will be happy to provide a schematic of the Richter Scale to any* Audio Amateur *or* Speaker Builder *reader requesting it.*

*Mr. Galo replies:*

*I agree that the Richter Scale is not just a subwoofer type. I point out in my review that Audio Control gives a formula for setting the crossover to frequencies above 250Hz. Audio Control is correct that the TL-075 is a pin-compatible replacement for the 4136, and I recommend its use. I am pleased the manufacturer approves of Procedure A for setting the satellite-to-woofer balance. I still think it would be worthwhile to check the setting with Procedure B. If both settings yield the same results, the setting is correct. Although I guessed the location of infrasonic filter incorrectly, I am sure that, if the manual had included a schematic, the location would have been evident.*

# SB *Mailbox*

## FAULTY FORMULAS

I made an error in the formulas for calculating the effects of series and parallel resistors in my *SB* 1/87 article "An Introduction to Frequency Response and LMP, Part I." On page 24, the formula for calculating $V_i$ (relative to *Fig. 9*) is only valid when you are not using $R_p$. Otherwise, the correct formula is:

$$V_i = \frac{R_p \times R_d}{R_p \times R_d + R_s \times (R_p + R_d)}$$

If, on the other hand, you use the arrangement illustrated below, the formulas become:

$$V_i = \frac{R_d}{R_d + R_s}$$

$$S_i = S_d + 20 \times (\log V_i) \quad \text{(logarithm base 10)}$$

$$R_i = \frac{R_p \times (R_d + R_s)}{R_p + R_d + R_s}$$

Ralph Gonzalez
Philadelphia, PA 19143

## ESL WONDER SPRAY

I have been working on electrostatic speakers for five years. Ideas from Roger Sanders and others have made constructing the speakers fairly simple. I did, however, become increasingly less fond of rubbing graphite and finally decided that there had to be a better way. EMI/RFI spray solves the problem.

Stretch your Mylar just tight enough to remove the major wrinkles and glue it to a stator. Mask off (with cardboard) the area to which you will glue the second stator and spray on an even coat of RFI shield. The spray dries quickly. You can then heat-shrink the Mylar (a few passes under the element of a stove set to "preheat" works quite well). (A good hairdryer is also effective.—Ed.)

Bill Maxwell
Portage, WI 53901

## GRILLE FABRIC

In response to Mr. Rauen's request for a source of knit fabric for grilles (*SB* 1/87), I recommend the Minnesota Fabrics retail chain in the midwest area. Successful fabric buying, however, depends more on your knowing what to buy than where you shop.

Look for polyester fabric, not too shiny (avoid acetate), and stretch a corner of it. It should stretch 2–3" per foot of material along both axes. Try breathing through it to assess air flow (you don't have to do this if light passes easily through the fabric).

Attaching the the fabric to the frame is easy if you follow these steps:

1) Attach the fabric loosely in the center of each edge of the frame. I use a stapler, but tacks or hot-melt glue should also work, depending on the shape and material of the frame.

2) Pull the corners of the fabric over the corners of the frame and attach them. Don't worry about wrinkles.

3) Working from each corner out, stretch the fabric as needed to pull out

## SB *Mailbox*

creases and bunching. Work gradually toward the midpoint of each frame edge.

4) Pull the fabric as needed to even tension, adding fasteners between the ones you have already put in.

Be aware of the "grain" of the fabric when cutting or attaching it. Most double-knit fabric has a different texture (nap) on each side.

Matthew Honnert
Carol Stream, IL 60188

## REALISM FACTORS

I look forward to each issue of *SB*. I have been interested in building speaker systems since high school where I had access to the signal generators, VTVMs and other equipment which enabled experimentation. Now, without instruments, I rely on kit manufacturers and driver vendors to supply data.

My most recent effort is a pair of Gold Sound Is. This two-way system is based on a generic 1″ ferrocooled tweeter and 6″ mid/woofer. Using their data for a ducted port, I optimized the dimensions of the box based on various concepts I found in *SB*. The outcome is very satisfying, except for the limited bass range. I plan to remedy this with a subwoofer.

I have become interested in tinkering with ways of enhancing "realism," imaging stability, depth, and other parameters. Taking cues from commercial manufacturers I plan to re-do my GS Is by adding a second "outside" mid/woofer which would be supplied with information from the opposite channel in reverse phase, as a means of reducing "crosstalk" to the listener position.

I would appreciate any information readers might have regarding the general topic of "realism" and specifically any experiences in arrangements of the kind I describe. Specifically, would a time delay circuit be important in obtaining a cancelling effect?

Nick Reiber
Westmont IL 60559

## GONDOR GONE

In *SB* 2/86, p. 6, it says, "Next time...we'll do plans for *Gondor* the sub—bass woofer...." As you know, the article did not appear. What happened? I am disappointed.

You see, for quite some time I have been searching for the ultimate subwoofer and I had hopes this might be the one. In fact, my desire for a superior quality and quantity of bass reproduction, along with a dissatisfaction with all commercially available models, was the main reason that I subscribed to *SB*.

Should I expect to see the woofer articles in the next issue? If not, could you suggest plans for another suitable subwoofer? I already have the *Audio Amateur* back issues which feature plans for the 24″ Hartley, but I need a design that will accurately respond even lower with more maximum acoustic output. Perhaps something with the percussive bass ability of, say, a Klipschorn, but with a much lower cutoff frequency.

H. Ray Mills, Jr.
Salisbury NC 28144

*In searching for a translator for the proposed* Gondor *article I discovered that Elrad (the German magazine that published the article) had been having all sorts of trouble with the design and it was not only failing with alarming regularity but was not performing as advertised. For this reason we decided against publishing the article.*

*David Ruether's* TAA *design will reproduce anything on a recording between 20 and 40Hz accurately. Anything below 20Hz gets dangerous to you physically and if you drive Ruether's device with a larger amplifier you can break dishes, knock pictures off the wall and crack windows.*

*You will notice that we are running subwoofers in the 1987 series of* SB *which I hope may be of interest.—Ed.*

## BRAVO BALLARD

Bob Ballard's "In Praise of Active Crossovers" (*SB* 4/86) is a nice encapsulation of the issue and should be studied by all audiophiles.

Going the biamplification route was the single most impressive improvement I made in my previous system. The present one was designed predicated wholly on the use of a three-way active crossover, and the driver array and signal routing would make little practical sense without the active nature of the crossover system.

Those of us who were watching carefully saw an interesting "Mailbox" debate between Mr. Ballard and Joe D'Appolito on active configurations subsequent to Ballard's article on a fully parallel phase-corrected design appearing in *SB* 3/82 and 4/82. Without getting deeply mired in this issue, I will just say I see both sides, but I finally elected to build my three-way on a classical parallel/series topology without any phase shifting compensation circuitry.

My crossovers are B4 Linkwitz-Riley, the first at 72Hz, and the second at 1100Hz. The Gauss 18″ woofers are each driven at 350W, the two pair of EV Force 10s driven separately at 100W per driver, and the Heil AMTs driven (when I complete them) from 20W tube amps.

To those who are not averse to etching their own circuit boards, I submit that the two op amp cascade filter is a truly ridiculously simple device, and the modern TO220 IC regulators make really good low-impedance power supply construction cheap and easy. While the economic obstacle of multiple amplifiers remains, the advantages to be gained are overwhelming.

Let me also address the economic issue. I subscribe to *Stereophile*, [Phone(US) 800-435-0715, Illinois 800-892-0753; Outside US: 505-982-2366] and I get evidence on a regular basis that audiophiles are seriously considering spending $5,000 for a power amplifier, because it will perform impeccably at high, mid, and low frequencies simultaneously. High powered equipment is being designed and built as wideband without compromise. Little thought seems to be given to the fact that, regardless of the bandpass of the amp, the ear detects the degradations of sound caused by the passive components in crossovers.

I believe a tremendous revolution in "high end" design might come to pass if amplifiers could be built for specific frequency ranges and power levels. Those of us who have experience with active crossovers have already developed preferences for our applications and buy amps at wonderfully low prices for these. Consider what distortion and other specs can mean relative to the narrower bandwidth and power levels in active systems. This starts one wondering also about exotic speaker wires as well, one of whose advertised advantages is superior full-spectrum transmission.

I fully agree with Bob Ballard. Let there be no inductors, whether air or iron core. Let there be no caps, high grade or otherwise. Let there be no Zobels, no attenuator pads, no passive EQ. Let there be only a lot of copper between the amp and the driver. It is much too simple and beneficial to do what is required with BIFET op amps to take any other method seriously.

When I use a passive crossover, it will be in the rear "ambient" systems. It may be the technically best passive that could be made for the arrays, but it could not supplant the active system used for the mains.

Paul W. Graham
Independence MO 64050

## MEASURING T/S PARAMETERS

Composite driver parameters received a lot of attention in *SB*'s 4/85 issue as well as how $V_b$ can be reduced through a reduction in $V_{AS}$. The question left unanswered for me was just exactly how the T/S parameters are measured. If I understand correctly, both drivers are wired in parallel and T/S parameters are measured normally. This would yield a $Q_{TS}$ approximately of a single driver, with only half the $R_E$, however.

But how do you measure the $V_{AS}$ with both drivers connected and take the literal value? Do you measure the $V_{AS}$ of a single driver and divide it in half or do you average it between both drivers and halve it? The *Mailbox* letters were confusing in that one letter mentions that $V_{AS}$ is halved while another letter states $V_{AS}$ doubles.

I own a pair of Magnepan SMGs and was considering upgrading the dividing networks from their original first-order 6dB/octave passive networks to a higher order network in hopes of a possible improvement in performance. I would like your suggestions as to whether a second-order 12dB/octave network, a third-order 18dB/octave network or perhaps, no change at all would work best with this speaker.

I use a Carver Sonic Hologram and am wondering, from the varying radiation patterns of various order networks, whether a particular network order would work best with the imaging capabilities of the Carver. I plan to bi-amp the SMGs with a pair of subwoofers using an 18dB/octave active network. Should I keep the cross-

*Continued on page 60*

# SB Mailbox

over slopes consistent throughout a multi-way system?

What is the best voltage drive level for measuring T/S parameters? One reader (*SB* 4/85) mentioned that to get predictable measurements of his Eclipse W1032R drivers he had to raise the drive level from the recommended 100mV to .5V. If you have the equations, or better yet, a BASIC source code listing that would enable me to plot the vertical polar response patterns of various order networks, I sure would appreciate them

I thank Mr. Bullock and *SB* for a wonderful publication. I look forward to receiving each issue.

David F. Del Zotto
Seattle WA 98133

*Contributing Editor Bullock replies:*

*Two identical drivers can be connected either in parallel or series, and used so that either one (isobarik) or both radiate directly to the outside of the box. Regardless of the four possible configurations chosen,* $f_s$, $Q_{ES}$ *and* $Q_{MS}$ *for the equivalent driver are the same as for one of the drivers. The equivalent driver* $R_E$ *doubles if the drivers are in series and halves if they are in parallel. In the isobarik configuration the equivalent driver* $V_{AS}$ *halves, otherwise it doubles.*

*If you connect and mount the drivers as you plan to use them, you should be able to measure these values using the usual procedures. Slight errors occur because both drivers are not "exactly" the same.*

*Hard and fast advice on crossover selection for a particular system is not possible because preference plays a large role. I tend to prefer higher order networks because they help minimize driver interaction. Even-order networks can help to give a more uniform vertical response pattern, although the inter-driver phase difference can upset this advantage. Higher-order passive networks usually "sound" more detailed to me.*

*The biggest improvement I have noted from changing crossovers occurs when I replace passive networks with active ones, regardless of the order. I have used both even- and odd-order active networks and they do sound different, but neither one sounds better than the other.*

*I have no idea what a sonic hologram does. I suspect that you should be able to choose a crossover network independently of this device.*

*According to Small, the T/S parameters should be measured at a low enough level so the test signal is an undistorted sinusoid. Half a volt seems a safe level to me. I have never had a problem with this, but I can see where it could happen because the parameter values depend on the drive level.*

*I accept my measurements on the basis of whether they are reproducible and I have had difficult cases. The biggest problem for me seems to be ambient noise. More than once I have wound up measuring a system outdoors at 4 a.m. on a still morning. What's worse is that I have to do it on two different occasions to check reproducibility.*

*I enclose a listing (*Fig. 1*) for a program that computes radiation pattern for a two-way system with an all-pass crossover. It doesn't take driver directivity or driver magnitude response into account, but it seems to have some relevance.*

```
100 ! Rad_pat
110 ! Radiation patterns for 2-way all-pass crossover networks.
120 ! Data entry
130 INPUT prompt "Crossover order is                               ":o1
140 INPUT prompt "Crossover frequency is                           ":f1
150 INPUT prompt "Pattern frequency is                             ":f2
160 INPUT prompt "Horizontal driver separation in inches is        ":h1
170 INPUT prompt "Vertical driver separation in inches is          ":v
180 IF o1=2 or o1=4 then GOTO 200
190 INPUT prompt "Tweeter polarity observed(1) or reversed(-1)     ":p1
200 INPUT prompt "Tweeter/woofer phase difference in degrees is    ":a9
210 LET w=f2/f1
220 ON o1 gosub 250,300,350,400
230 GOTO 440
240 ! First-order network
250 LET a1=p1*3.14159/2
260 LET L=1/sqr(w*w+1)
270 LET H= w/sqr(w*w+1)
280 RETURN
290 ! Second-order network
300 LET a1=0
310 LET L=1/(w*w+1)
320 LET H=w*w/(w*w+1)
330 RETURN
340 ! Third-order network
350 LET a1=-p1*3.14159/2
360 LET L=1/sqr(w*w*w*w*w*w+1)
370 LET H=w*w*w/sqr(w*w*w*w*w*w+1)
380 RETURN
390 ! Fourth-order network
400 LET a1=0
410 LET L=1/(w*w*w*w+1)
420 LET H=w*w*w*w/sqr(w*w*w*w+1)
430 RETURN
440 LET a2=a9*3.14159/180
450 LET a8=2*3.14159*f2/13544
460 PRINT "ANGLE","REL SPL"
470 FOR t=-70 to 70
480     LET t1=3.14159*t/180
490     LET a3=a8*(h1+v*tan(t1))
500     LET x=a1+a2+a3
510     LET p2=L*L+H*H+2*L*H*cos(x)
520     LET p=10*log(p2)/log(10)
540     PRINT t,p
550 NEXT t
560 END
```

**FIGURE 1: Editor Bullock's radiation pattern program code.**

## SPIKE POINTERS

Frustrated by the high prices and low quality of spiking kits for floor-standing speakers, I devised the following inexpensive alternative. Fasten four ½" or longer T-nuts to the base of each speaker by drilling holes just a little smaller than the T-nuts, coating the nuts with slow-curing epoxy, and tapping them in with a hammer. The epoxy is especially important with short T-nuts, which do not always stay in by themselves.

For each T-nut, grind a short length of threaded rod or decapitated bolt to a long, gradual point. Put a regular nut on each spike to lock it. Install these threaded spikes when the glue is set, being careful to adjust them so the speakers sit evenly and solidly on the floor. If your T-nuts are long enough to break through the speaker cabinets, cap them to avoid air leaks (you might be able to use short scraps of the threaded rod or short bolts). I recommend ¼–½" diameter threaded rods with medium or fine thread.

A rod or bolt threaded on the inside *and* outside that you could bolt to the speakers would be better than the T-nuts, but I have not found such a fitting.

I also want to share an experience I had with a friend's B&W DM-220s, which sounded overly boomy. When we opened the cabinets, we found they had no bracing at all. We cut pieces of scrap wood to fit cross-wise in all directions, and glued them to the inside panels, spaced about 6" apart. After the glue dried, we put the original fiber wool back in and reassembled the speakers. The difference in sound was tremendous.

C. J. Poulos
Hartford, CT 06106

*Dick Marsh of Livermore, CA 94550, has informed us that HILTI Fastening Systems, PO Box 454500, Tulsa, OK 74145, (918) 627-9711, manufactures threaded studs which, with their washers removed, make good speaker spikes.—Ed.*

## TINIER TICK TRICKS

In his "Passive Crossover Networks: Part III" in *SB* 3/85, Bullock states that, with his active crossover system, "The inevitable pops and ticks from records are not nearly as objectionable because even though they are still present, the duration of each one is so much shorter." Why is this so?

Steve Mallon
Chicago, IL 60634

*Contributing Editor Bullock replies:*

*I attribute the reduced duration of pops and ticks in my active crossovers to better transient response. My active networks contain generally higher quality components, which should help transient response. On the other hand, the crossovers are usually of higher order, which should hurt transient response.*

*My comment was based on subjective evidence—I may not be accurately interpreting what I hear. The active networks still sound better to me. Colloms gives ten reasons in* High Performance Loudspeakers *why this should be so. I encourage you to try active crossovers for yourself when you get the chance. I suspect you will hear the difference.*

## FIGHTING FERROFLUID

I am having trouble equalizing the impedance of my Dynaudio D21-AF tweeter. The parameters I measure do not correspond to the the spec sheet. I have never had trouble measuring Q parameters of other drivers I own, and am sure that my test equipment is working properly.

Art Katoozian
Los Angeles, CA 90048

*Contributing Editor Bullock replies:*

*I believe the problem you are fighting is Ferrofluid. It makes measuring parameters correctly almost impossible, especially for tweeters. If I were you, I would forget about trying to equalize the driver. Just keep the crossover point high enough to stay away from tweeter resonance (at least 32kHz). Use the average impedance around the crossover requency as the filter-design impedance.*

*According to Dynaudio's spec sheet, the impedance only varies between 6 and 8Ω in this region. That is 7Ω to within a 15% tolerance, which is quite respectable (much better than the equalized 3Ω value. It should not have any noticeable effect on response.*

*I have never done a sensitivity study of the impedance equalizer, but I suspect 20% tolerance components will work fine. Most people only use the formulas for a ballpark figure and tweak the circuit to get impedence as flat as possible.*

*A driver's resonance peak is never as sharp*

*Continued on page 64*

**au·dio·phobe / od-ē-ō-fōb / n** 1: An audio hobbyist with an all-consuming fear that he does not possess this month's "IN" equipment, the most expensive equipment or the least available equipment. 2: Someone who uses music as a medium by which to evaluate equipment, usually manifested by an extreme dread of having to listen to music without talking or getting up to check something in the system. 3: Someone who would freely consider spending twice the value of his record collection on a power amplifier. 4: Someone who loves to talk about audio but never expresses an opinion of his own, all discussion being based on what he has read, not on what he has heard.

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## SIXTH ORDER VENTED BASS

*continued from page 16*

ture in my workshop was around 70°. I did the right channel a few days later in the middle of a heat wave with the ambient temperature above 90°. (Nobody ever said I wasn't dedicated.) I was not able to come to an acceptable alignment or correlate the results between channels. Then, I realized many of the constants used in calculating box size and tuning are temperature-dependent.

I ran the right channel again a few days later when the weather had cooled, and obtained results which mostly agree with those for the left channel. To be on the safe side, I rechecked the left side, but they agreed quite closely with the results I obtained a week earlier. The moral is, for best performance, do your measuring at a temperature that will likely correspond to the one in your listening environment, and don't make any changes on a hot or cold day.

I hope this article encourages some of you to proceed with your dream system. I will be happy to correspond with anyone wishing to share experiences with this or a similar system. I sincerely believe a sixth-order vented system is a most viable design.

*Editor's Note: To correspond with the author, please state any questions clearly, leave room for his response, include a self-addressed, stamped envelope, and send it to him in care of the magazine.*

**ABOUT THE AUTHOR**

*John Levreault is the computer industry marketing manager for the Component Group of Analog Devices in Wilmington, Massachusetts, and has been involved in analog design engineering for 15 years, most recently specializing in high-speed data conversion. He has been constructing his own audio equipment for more than ten years and has been designing speakers for the last eight. He is an avid acoustic guitar player and also enjoys amateur radio.*

## SB *Mailbox*

*Continued from page 61*

*when Ferrofluid is used. One of the reasons Ferrofluid is used is to damp the peak.*

*If I am designing a box to fit a tabulated response, I usually stop worrying about ∝ when it is within 20% of the tabulated value. I then try to get h within 10%. I have not followed this procedure, though, since I wrote BOXRESPONSE (SB 1/84). Now I use a tabulated alignment only for initial design values. I then measure the parameters, feed them into BOXRESPONSE, and see what the response looks like. If it is close to what I want (which is not always flat), I consider the alignment complete. I rarely need to make further adjustments, and those I do need to make are invariably just changing the vent or sealing a leak.*

## DANBY DATA DESIRED

In the late 1940s, I visited a small shop in Philadelphia named Danby Radio Corporation. Danby had developed a corner horn called the Danby Jupiter, which had quite a reputation. It was hand-crafted and sold at a reasonable price. It had phenomena bass response.

The horn was driven, as I recall, by a Stentorian 12" woofer. Danby once said that one of the larger high-fidelity manufacturers had approached him with a bid to license the design, but the offer was not acceptable. The Danby horn apparently died, but my recollection of its superb sound did not. Even Klipsch, in one of his later "historical" papers, paid tribute to the Danby design.

Has anybody else seen or heard of the Danby Jupiter Corner Horn? I will appreciate any information on the design.

L. Paul Monahan
Reston, VA 22091

## CARELESS ENCLOSURE

I am amazed at how little time, money, and interest goes into developing loudspeakers and enclosures. I often open $1000 speakers and find $100 worth of parts. As a manufacturer of many kinds of loudspeakers (transmission line is our specialty), I have done a great deal of experimenting with different enclosure designs and materials. Many big-name systems can be improved significantly with $25 worth of changes in enclosures and components.

Many dedicated speaker makers produce beautiful systems, but they cannot afford multi-page ad campaigns or tantalizing dealer mark-ups.

Kirk Neal
Mercerville, NJ 08619

## HORN DILEMMA

I have plans for two Klipsch-style bass horns. One is dated 1945 and has no model identification (prints are reported to have escaped from Bell Labs about that time). The other is a Speakerlab SK.

The opening in the first speaker's mounting panel through which it drives the throat is a section of a circle bounded by two parallel chords 3" from the center. The circle radius is 5.5", and I calculate the opening to be 62.56 sq. in. In the front of the panel the throat is a total of 66 sq. in.

The SK horn's opening is 39 sq. in. and its throat is 75 sq. in. Why is the SK horn's throat so much larger than its opening?

I am also wondering how a driver can be more efficient attached to a horn than to a conventional baffle. How does this fit with the law of conservation of energy?

I am building Mr. Edgar's midrange horn (*SB* 1/86) using the Polydax 6" driver. The text suggests a ½" gap, but *Fig. 24* graphs response with 1" gap. Which is correct?

Paul Tanner
Ft. Oglethorpe, GA 30742

*Contributing Editor Edgar replies:*

*The 1945 Klipschorn plans are correct; the SK horn plans have the wrong throat opening (recently pointed out to me by SB reader Dave Wharran). Basically, as you discovered, the 39 sq. in. throat opening of the SK horn plans should be 78 sq. in. to match the two sections that lead up and down from the throat. It has always been a well-established horn principle that you can divide a horn into multiple sections as long as the total cross-sectional area follows the exponential area expansion rule. Obviously, the designer of the SK horn was ignorant of that fact.*

*I would build the older Klipschorn. Its 62½ sq. in. throat is closer to the optimum throat sizes associated with the 15" drivers suitable for horns. Either the JBL 2220H (56 sq. in.) or the EVM 15L (61 sq. in.) should work well. The modified SK horn throat of 78 sq. in. is too large for most 15" horn drivers (with the exception of the 71 sq. in. JBL E-130.*

*Another reason to choose the Klipsch design is that it has Klipsch's "rubbber throat," which allows you to annul the throat reactance at the flare cutoff to achieve optimum bass response. The term "rubber throat" refers to the way Klipsch stretched the initial throat section to be longer than the flare rate specifies. By adjusting the sealed back chamber's volume so the horn-driver-speaker-system resonance was equal to the flare frequency, Klipsch was able to achieve good bass response below the flare frequency. His critics wondered if he had worked some "black magic."*

*A horn is an efficient transducer because it acts as an acoustical transformer that couples the low impedance of the air load to the high impedance a driver likes to work into. The end result is that the horn system's radiation resistance is dominated by a real resistance with a smaller reactive term, and many real acoustic watts radiate into space.*

*In contrast, a direct radiator is "mass loaded" by the mass of the cone and voice coil (a very reactive load). The air-load portion of the radiation resistance is very small compared to the reactive component. Consequently, less than 1% of an amp's power ends up radiating as real acoustic power. Most of the power goes into heating the voice coil and the air inside the enclosure. Some reactive power is reflected back to the amp, where some of it produces additional heat in the output stage.*

*In an ideal horn system, efficiency can approach 50% when throat resistance equals voice-coil resistance. Typical efficiency for the more realistic case of a finite horn with a reactive component in the radiation resistance (throat reactance near flare cutoff) can be 20%.*

*Nature never gives anything without taking something back. One thing you give up in return for high efficiency in a horn system is bandwidth. A 15" woofer in a horn will be lucky to reach 300Hz before the cone's mass chokes response. A 15" woofer in a sealed box can easily achieve 1kHz. Another drawback to horns is size. Efficient bass horns are large. If you are willing to trade efficiency for size, you can easily design a sealed-box woofer system with one-tenth the volume of a Klipschorn.*

*The prime advantage of horn systems today is their low distortion and high dynamic range. Relatively low-power amps can drive them. A well-designed horn's overall sound has an effortless quality (a direct-radiator system may have a strained quality at large SPLs).*

*To answer your final question, there was a typo in Fig. 24's caption. The gap should be ½", not 1".*

## ISOBARIK ODYSSEY

In the Mailbox section of *SB* 3/86 Paul Graham and John Cockroft discuss my statement (*SB* 4/85, p.52) that isobariks can have poor midrange. In hopes that information about the shortcomings of my isobarik systems will be helpful to them and others, here is the story of my isobarik adventures.

My first (and probably best) isobarik system was a large subwoofer using KEF B139s in a 3 cu. ft. box with about 2 cu.

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## SB *Mailbox*

*Continued from page 65*

ft. behind the rear woofer and the rest taken up by the isobarik chamber. I used roughly the same volume proportion in all my isobariks. I mounted all my inner woofers on internal baffles rather than n tunnels. The first subwoofer went very deep, but the voice coil bottomed rather easily and the sound wasn't as tight as I wanted. It had, however, the best bass I ever got out of my B139s.

I was impressed enough to try again, and built a small two-way isobarik using the little KEF B110s. The rear woofer's cone was about 6" behind the front woofer's. I mounted them back-to-back in parallel. The bass was very good (it went considerably deeper than the same woofer did in a transmission line and was just as tight), but the woofer still bottomed easily. The midrange was very poor. A crude frequency-response curve revealed no large peaks or dips, but the clarity was like that of a cheap transistor radio. I decided the inner woofer was garbling the midrange, and tried adjusting the damping materials with no success. I removed the inner woofer, which helped midrange significantly. I finally put the drivers back in their old transmission-line enclosures and threw the isobarik boxes away.

A year later I realized I could probably have saved the isobariks by inserting a 100Hz low-pass filter in series with the inner woofer, the output of which was an asset only in the bass and a liability in the midrange.

In my third isobarik, I tried to use the principle to build a compact subwoofer with the B139s. The result sounded tubby. I decided that lower woofers would come closer to the sound I was looking for. I bought a pair of Dynaudio 21W54s.

I originally put the 21W54s in large (2.2 cu. ft.) sealed boxes to serve as the bottom end of a Strathearn system, but soon decided I wanted deeper bass in smaller boxes. I bought two more 21W54s and built 1.2 cu. ft. isobarik enclosures. I mounted the woofers sort of back-to-back, with the inner woofer below the outer one, allowing their magnets to overlap and requiring less volume in the chamber. I wired a 10mH inductor in series with the inner woofer in one of the boxes and temporarily crossed the woofer systems to 1" dome tweeters to experiment. The midrange was far better with the 10mH inductor in place than without it, but as I supected, the big sealed boxes had the best midrange. When crossed over to the Strathearns at 300Hz, the midrange problem of the isobariks was not significant (although the normal rise in the 21W54s'

## SB Mailbox

still was). I never bottomed these woofers, but they didn't go nearly as deep as my first isobarik subwoofers did.

Mr. Cockroft's systems could be free of my garbled-midrange problem because his woofers are closer together, because of superior internal geometry, or because his cones are more acoustically opaque. His designs have been much more successful than mine in the midrange, so I encourage you to stick to his recommendations if you intend to use your isobariks into the midrange.

I have found that isobarik loading does more than halve $V_{as}$. The Dynaudio Isobariks I just described replaced a single-woofer sealed system roughly twice their size. The current theory predicts that they'd have very similar bass response, but they didn't. The single-woofer system's response sloped gently down from 110Hz, was ±3dB at 60Hz, and ±16dB at 30Hz. The isobarik had a mild hump around 80Hz (roughly the resonance frequency of the inner woofer), was ±3dB at 40Hz, and ±14dB at 20Hz. I think you can expect deeper bass from isobarik loading than the current theory predicts.

Duke LeJeune
New Orleans, LA 70118

### BRICK ENCLOSURES

I want to build brick enclosures for loudspeakers on either side of my new fireplace. I imagine masonry enclosures are exceptional. Does anyone have information on this kind of system?

Daniel A. Shea,
Wichita, KS 7214

*Editor Dell replies:*

*The ony references to brick enclosures for loudspeakers I know of appeared in G. A. Briggs'* Loudspeakers, *third edition, p.84, (Wharfedale, 1949), which is now out of print.*

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# Classified Advertising

**CROSSOVERS, ELECTRONICS AND** passive second, third, and fourth-order. Example: stereo electronic fourth-order assembled, $49.95; mono three-way LC third-order, $18.70. INDUCTORS: $2XSQRT L (Mhy). Also 40W/channel amp kit, $59.95; all parts except transformer, heatsinks, and cabinet. Amp assembled complete in cabinet, $99.95. ***BRUSSEAU ENGINEERING,*** 561A Vine St., Glendale, CA 91204, (818) 500-1528. T2/87

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## FOR SALE

Hiphonic MC-A5 moving-coil cartridge brought from Japan—used three hours. Not compatible with my system. Sounds great. $350-400 value, will sell for $175. Dave Staub, RD #3, Box 448, Dallastown, PA 17313.

Focal 15N 700, $250; 7N303, $60; T120FC, $25; F600 24dB passive crossover, $80; JBL 2235H (Aquaplas doped), $200; 112A (8″ midbass), $100; 2105H, $50; Dynaudio 21W54, $75; D52, $40; D21AF, $25; Peerless PMT51, $20; KJ20MDR, $10; TP165F, $20; Audax MHD21B37R 8″ Bextrene, $45; (all the above sold in pairs). Original BOSE 901 system, $200 plus freight; Revox A77 MK IV ¼-track with accessories, $350. V.E. Nichols, 3043 Las Vegas Trail, Ft. Worth, TX 76116, (817) 244-5763.



Strathearn ribbons, two pairs with transformers, $350; Teac TN400 direct-drive turntable module, armless, $100; Corol H-100 horn tweeters, 30W @ 110dB SPL/W, great pro-sound, $100 each; two ITT dome midranges, $50 pair. John, (617) 851-4209, 6-10 EST.

TL072 op amps, brand new, never used, received direct from TI and Motorola. Great for modifying CD players (smoother and more detailed midrange). $5/pair. Harold, (301) 661-1202.



Dynaudio D28AFs, $55/pair; Audax TX 2025 RSN, $55/pair; Becker 910 A100 woofers, $58/pair; Eico ST40 integrated tube amp, $65; ST97 FM stereo tube tuner, $25; *Audio Amateur* 1973, 1975, $8/year ppd. Steve Fritz, (805) 736-0259.

Two each Electrovoice crossovers, X36-3500Hz, X8-800Hz. Two Electrovoice T35 horn tweeters, also Shure 688 4-mic, one line stereo mixer. Will trade all for pair of good bookshelf speakers or make offer. Louis Dyer, 1584 W Holland, Fresno, CA 93705, (209) 221-8592 after 10 a.m.

Pair of Galo TL-10s with Madisound woofers, passive crossover mid to tweeters. Now wired for triamp. UGLY! Can't ship. $600. DeCoursey three-way for above, $150; NAD 3150 int amp, $250; NAD 2150 power amp, $165. Sell all for $1,000. Don Antle, Box 2092, Vernon, TX 76384, (817) 552-2021.

Acoustat six (pair), very good condition, $2250; Acoustat TNT-200 amp, excellent, $600; Audionics Space & Image Composer, very good, new R/C, $400; dbx 3BX-R remote control, new, $100; JVC HM-200E binaural headphone-mikes, excellent, $80; Signet MK12T transformers, very good, $75. William Sommerwerck, Apt. 217, 133 W Locust, Mechanicsburg, PA 17055.

Futterman H3AA 1979, NYAL upgraded 1981, $1600/pair; Audible Illusions mini mite I preamp, $225; Polk Audio L.F.14, make offer. Phil Keller, 129 Ishi Circle, Sacramento, CA 95833, (916) 922-5369 evenings.

Carver C4000 preamp, excellent condition, updated version of Sonic Hologram circuit. Best offer over $450. Shipped prepaid in original carton. Allen Keller, 1670 Ritter Dr., Muskegon, MI 49441, (616) 780-3296.

Pair 18″ Electrovoice EVM18B Series II bass drivers in original sealed cartons. List $340 each, first $400 takes both. C. Orme, 1027 Valley Rd., Colorado Springs, CO 80904.

Hafler 101 preamp, $175; Pioneer 40W integrated amp, $100; Vector Research equalizer, $80; Sanyo turntable with cartridge, $60; Kenwood tape deck, $100; Cherrywood subwoofer and satellite speaker system, $200. All with original boxes. Package special, except Hafler $499. Charles Loengard, 20 West 86 St., NY, NY 10024, (212) 496-6827.

## WANTED

Ribbons for Decca ribbon tweeters. Kim Tasler, RR 1, Churdan, IA 50050, (712) 685-3166.

BGW 250B amp, past *Stereophile* issues, Transcriptors Skeleton or Saturn table, moving-coil cartridge suitable for Transcriptors Vestigal arm, any literature on ESLs. Dave, 924 Hulton Rd., Oakmont, PA 15139, (412) 828-9049.

Acoustat panels, speakers, tube servo-amps; Pass A40 power amp; parts or amps; advice or modifications on driving electrostatic speakers. Dennis Boyle, Sulphur Springs, TX 75482/400, (214) 885-0966.

Empire 598 Type II turntable, Allied or similar catalogs, equipment directories before 1963. Steve Fritz, (805) 736-0259.

Dynaco MK VI vacuum tube amps for parts. Complete or incomplete kits. Need output transformer and choke. Will pay top dollar. Also, Dynaco C-100, energy storage (cap) banks for ST-416 amp. Will pay top dollar. Jeff Goldman, (215) 628-2000 during working hours.

Service manual for KLH-TNE 1000 Impulse Noise Reduction Unit. Steve Mallon, 3419 N. Nottingham, Chicago, IL 60634.

Electrovoice Aristocrat speakers in good working condition. Also interested in artciles, owners' manuals and other information about the Aristocrat series of speakers. Jeofrey S. Bean, 31 Kendall Lane, Framingham, MA 01701, (617) 872-0519.

# CLUBS

**Space in this section is available to audio clubs and societies everywhere free of charge to aid the work of the organization. Copy must be provided by a designated officer of the club or society who will be responsible for keeping it current. Send notices to Audio Clubs in care of the magazine.**

**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.



**AUDIOPHILES IN CENTRAL PENNSYLVANIA** (also eastern Pennsylvania and Delaware): Interested in forming a serious audio organization? Contact Steve Gray, 625F Willow St., Highspire, PA 17034 or phone (717) 939-4815.

**SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY** (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, AB listening tests, equipment clinics, recording studio visits, annual picnic and audio fun. The club journal is *LC, The SMWTMS Network*. Corresponding member's subscription available. Call (313) 477-6502 (days) or write David Carlstrom, SMWTMS, PO Box 1464, Berkley, MI 48072-0464.

**THE BOSTON AUDIO SOCIETY INVITES YOU** to join and receive the monthly B.A.S. SPEAKER with reviews, debates, scientific analyses, and summaries of lectures by major engineers. Read about Apogee, Nytal, Conrad-Johnson, dbx digital, Snell, music criticism and other topics. Rates on request. PO Box 7, Boston, MA 02215.

**PACIFIC NORTHWEST AUDIO SOCIETY** (PAS) consists of 50 audio enthusiasts meeting monthly, second Wednesdays, 7:30 to 9:30 p.m. at 4545 Island Crest Way, Mercer Island, Washington. Be our guest, write Box 435, Mercer Island, WA 98040 or call Bob McDonald, (206) 232-8130.

**THE ATLANTA AUDIO SOCIETY** started in October 1983 and has regular meetings on the third Sunday of each month as well as special programs with leaders in the industry, such as Mr. William Conrad of Conrad-Johnson and Mr. William Johnson of Audio Research. We are currently looking for additional members in the Southeast. all members receive the minutes of each meeting and program, as well as other relevant announcements and correspondence. For full information and membership packet, write Atlanta Audio Society, PO Box 92130, Atlanta, GA 30314, or call Howard Royal in Newnan, GA, (404) 253-6419.

**HI-FI CLUB OF CAPE TOWN**, South Africa issues monthly newsletter for members and subscribers. Get a different approach to understanding audio, send two IRCs for next newsletter to PO Box 18262, Wynberg 7824 South Africa.

**WASHINGTON AREA AUDIO SOCIETY** (N. VA., MD and DC) 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 contact: Horace J. Vignale, 13514 Bentley Circle, Lake Ridge, VA 22192-4316.

**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.

**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 share with other members equipment at every meeting. Plans for the future are to invite audio luminaries to lecture, and to occassionally incorporate and include "live" music. 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) 251-7044 or write Dan Fraser, VAS, Box 4265, Vancouver, BC, Canada V6B 3Z7. We would like to be on your mailing list.

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

## *Advertising Index*

# ETON IS COMING!

## 4-200/25-hex

### WOOFER-MIRANGE 4" HEXACONE-DIAPHRAGM

*The 4-200/25-HEX is one of the most sophisticated 4" woofer/midranges on the market. By using the excellent HEXACONE-diaphragm the performance is only comparable with electrostatic and electromagnetic drivers. We recommend this driver for satellite or mini boxes as well as for midrange applications in larger combinations of high-end speaker-systems.*

| | | |
|---|---|---|
| Nominal-Diameter | (mm) | 120 |
| Frequency-Range | (fo) | 60-4000 Hz |
| Impedance | (Z) | 8 Ohms |
| DC-Impedance | (RE) | 5,4 Ohms |
| Resonance | (fs) | 54,2 Hz |
| Total Q | (QTS) | 0,18 |
| Electrical Q | (QES) | 0,19 |
| Mechanical Q | (QMS) | 6,65 |
| Sensitivity | (dB/w/m) | 92 |

## 7-380/32-hex

### WOOFER 7" HEXACONE-DIAPHRAGM

*The 7-380/32-HEX is one of the most sophisticated 7" woofer on the market. By using the excellent HEXACONE-diagphragm the performance is only comparable with electrostatic or electromagnetic drivers. The frequency and step-function assure the 7-380/32-HEX to be uncoloured and well suited for high-end 2 way bass reflex systems.*

| | | |
|---|---|---|
| Nominal-Diameter | (mm) | 185 |
| Frequency-Range | (fo) | 40-3500 Hz |
| Impedance | (Z) | 8 Ohms |
| DC-Impedance | (RE) | 6,8 Ohms |
| Resonance | (fs) | 37 Hz |
| Total Q | (QTS) | 0,27 |
| Electrical Q | (QES) | 0,3 |
| Mechanical Q | (QMS) | 2,64 |
| Sensitivity | (dB/w/m) | 91 |

## 8-480/32-hex

### WOOFER 8,5" HEXACONE-DIAPHRAGM

*The 8-480/32-HEX is one of the most sophisticated 8,5" woofer on the market. By using the excellent HEXACONE-diaphragm the performance is only comparable with electrostatic and electromagnetic drivers. The frequency response and step-function assure the 8-480/32-HEX to be extreme uncoloured and well suited for high-end 2 or 3 way bass reflex systems.*

| | | |
|---|---|---|
| Nominal-Diameter | (mm) | 220 |
| Frequency-Range | (fo) | 30-3000 Hz |
| Impedance | (Z) | 8 Ohms |
| DC-Impedance | (RE) | 6,5 Ohms |
| Resonance | (fs) | 30 Hz |
| Total Q | (QTS) | 0,33 |
| Electrical Q | (QES) | 0,38 |
| Mechanical Q | (QMS) | 3,08 |
| Sensitivity | (dB/w/m) | 91 |

ETON 300 HEX KIT

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