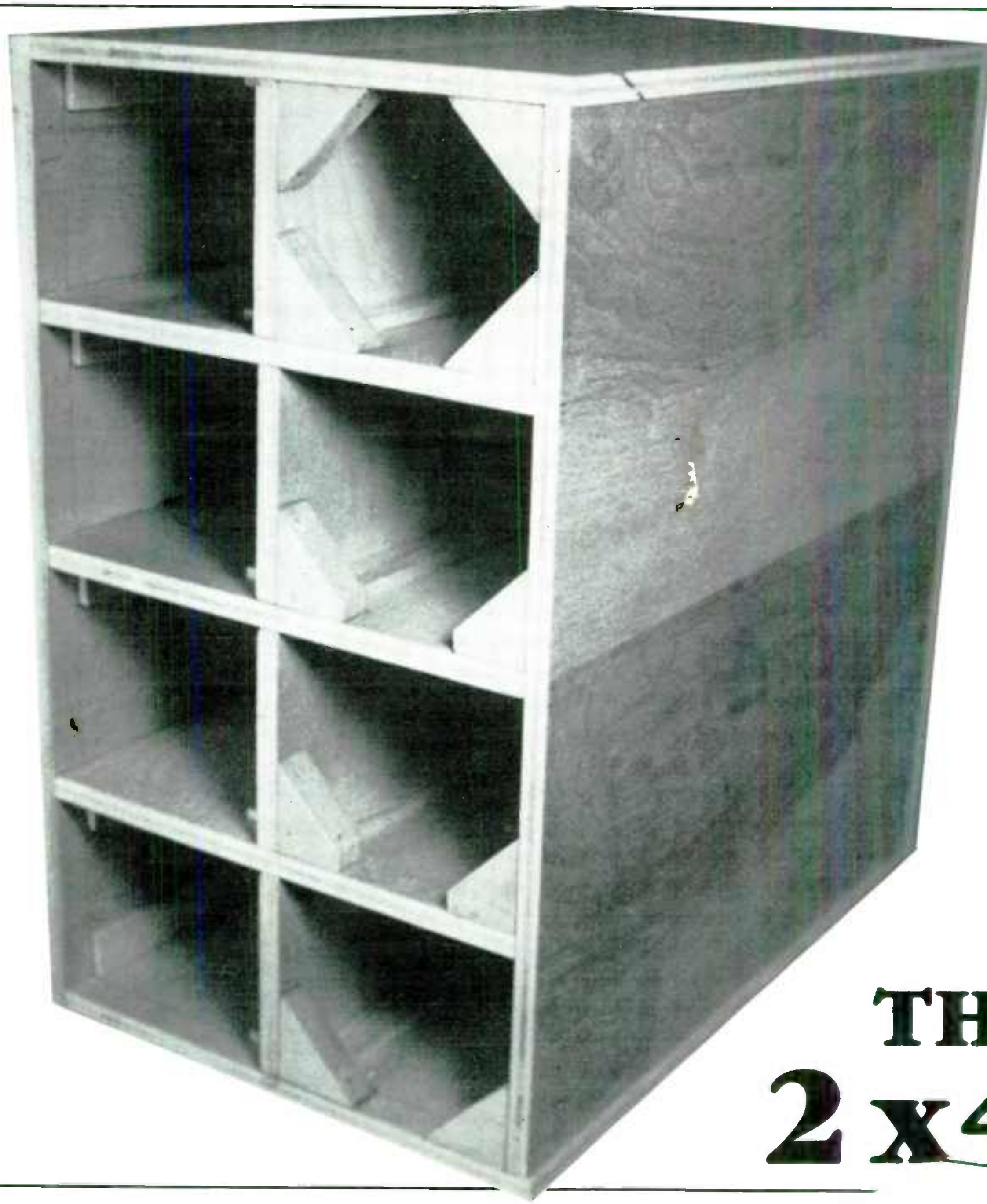
TWO/86

Aureays (Bach)

U.S. $4.00
CANADA $5.00

SPEAKER BUILDER



THE 2x4

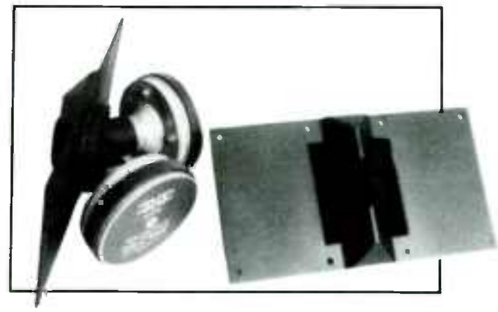
Good News

DCL COMPANY developed the X-Vector high frequency wave guide for high frequency sound dispersion from industry standard drivers. The unit's dual threaded throats utilize two high frequency drivers to provide a 140 degree wide by 60 degree high dispersion pattern through its specially engineered cross-vector wave guides and phasing devices. From its output lens the X-Vector delivers a controlled soundfield.

The integral 6 by 11-inch baffle surfaces make mounting easy. Its 1-inch throat opening allows the performance/power capabilities of typical compression drivers to be realized from a relatively small, easily installed and economical configuration. The 2kHz to 22kHz frequency range requires a crossover frequency at 3kHz. X-Vector is made from structural aluminum and high-density compounds, finished in a black enamel with a protective acoustic foam over the mouth.

Ask DCL Company, 10822 SW. 188th St., Miami, FL 33157 for more information.

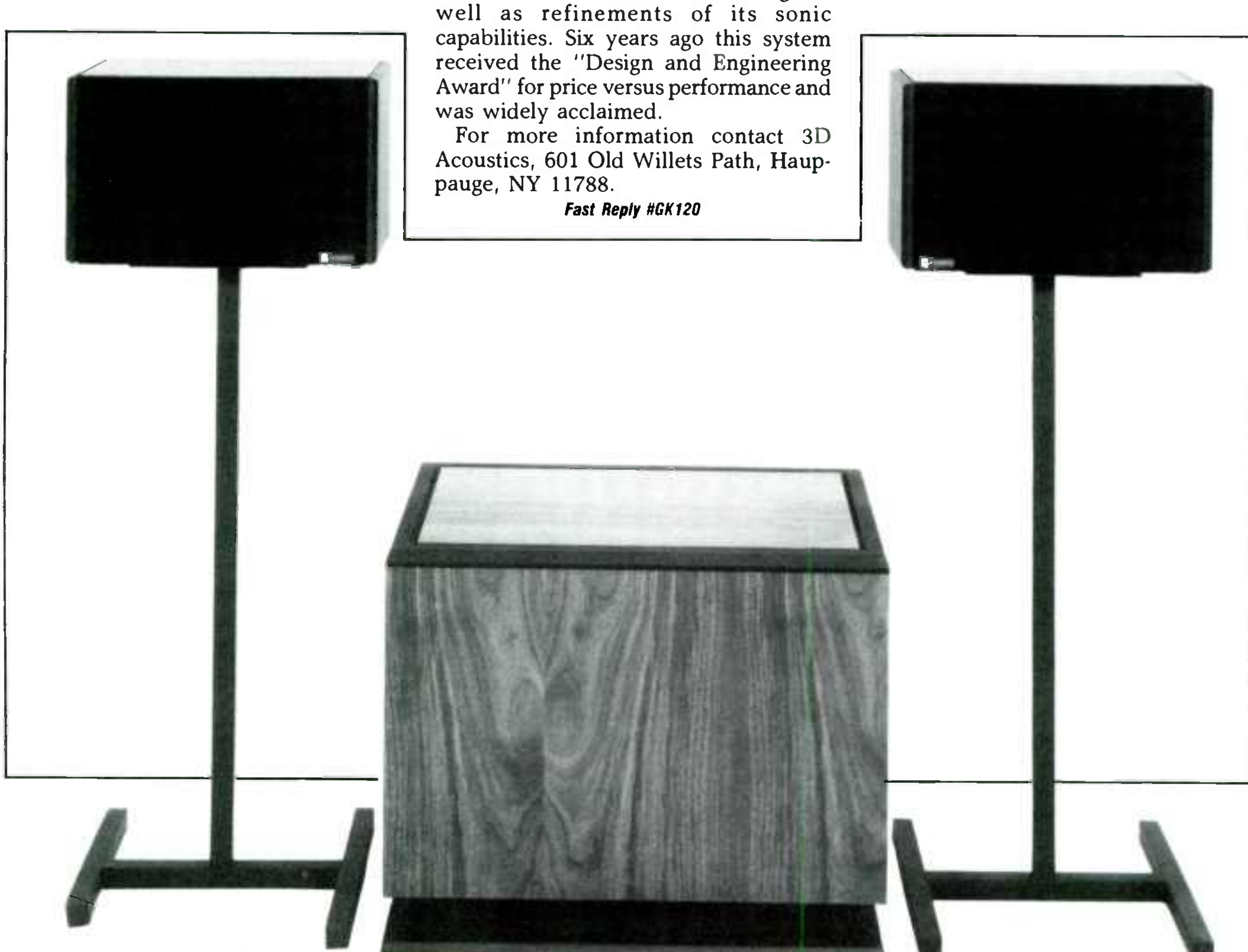
Fast Reply #GK138



3D ACOUSTICS re-introduces its Three Piece Loudspeaker System, the Model 3D610, with a new black and walnut design as well as refinements of its sonic capabilities. Six years ago this system received the "Design and Engineering Award" for price versus performance and was widely acclaimed.

For more information contact 3D Acoustics, 601 Old Willets Path, Hauppauge, NY 11788.

Fast Reply #GK120



BOSTON ACOUSTICS' A40 Series II two-way acoustic suspension loudspeaker system is an advanced version of their original A40. While it is the same size, the Series II incorporates new drivers, which the manufacturer claims produces extended bass response, greater sonic accuracy, lower distortion and higher power handling. The enclosure is available in rosewood or black vinyl.

BA's new 6-inch bass/midrange has a polymer diaphragm formulated to minimize cone breakup and insure smooth response in the Series II. The new ¾-inch high frequency driver is a proprietary design with a contoured faceplate, flush mounted in the baffle to eliminate diffraction. Its voice-coil is ferrofluid cooled for the wide dynamic range of compact disks. The unit features a frequency response of 65Hz–20kHz with power handling at 40W.

For more information write Boston Acoustics, 247 Lynnfield St., Peabody, MA 01960.

Fast Reply #GK336

dbx's Soundfield® One loudspeaker system, Model 1A, with their Controller can be used with other speaker brands and operate in any room. The Controller provides unusual tone-shaping, filters and rumble suppression for all loudspeakers. The Soundfield 1A system incorporates the same technology and achieves the same properties of the earlier Model One.

dbx claims their Soundfield Imaging® effects a significant improvement in the spatial perspective of music reproduction. With the three-dimensional soundfield and accurate tonal balance a listener need not sit midway between two speakers to experience proper tonal balance and satisfactory stereo imaging.

The Controller provides over all system equalization by dealing with room acoustic problems and compensating for various sonic recording deficiencies. It is an outboard line-level analog signal processor that connects to the tape monitor, EPL (external processor loop), or preamplifier output jacks of a stereo system. It incorporates high and low frequency compensation, ambience control, tape monitor and bypass controls, and features switching to compensate for against-the-wall loudspeaker placement.

For further information write dbx, PO Box 100C, Newton, MA 02195.

Fast Reply #GK583

HIFISOUND is now the owner of Strathearn Audio Limited, and the manufacturer of the Strathearn, now known as the Stratec SLC II system. The Strathearn midrange and high-frequency driver is an electrodynamic transducer consisting of a thin, low-mass transparent polyester diaphragm stressed across a low-resonance support frame. The conductive elements confront force field magnet planes giving a highly uniform balanced field in the conductor plane. Unlike electrostatic units, no separate power supply is required and the non-reactive matching transformer is suitable for coupling it to conventional amplifiers.

For further informaton contact Hifisound LSV, Saerbeck & Morava, Judefelderstrasse 35, 4400 Munster, W. Germany.

Fast Reply #GK121

The FG 1001 is **TECH ART's** new 1MHz function generator circuit board, modularly designed to fit into any system, covers a frequency span of 0.01Hz–1.0MHz in six selectable ranges, each continuously variable in a 100:1 ratio. Sine, triangle and square waves are selectable and variable to 10V peak-to-peak with an adjustable offset of ±5V DC. Trimmers allow user-adjustment of sine and triangle symmetry, sine and triangle distortion, and maximum triangle and sine wave output levels.

The 6 by 4½-inch circuit board plugs into a 22/44 pin-edge card connector and is compatible with industry standard card cages. Switches and controls connect via 16-pin dip sockets. The FG 1001 facilitates making audio oscillators, bench-top test equipment, communication devices and laser entertainment systems.

For more information write Technological Artisans, 53 W. 72nd St. (3G), New York, NY 10024.

Fast Reply #GK134

ZAPCO, Zeff Advanced Products Company, introduces two new amplifiers to its line of automobile audio products and announces its contract with Ford Motor Company to design and manufacture two new mobile audio products.

ZAPCO's M-80 80W monaural amplifier/electronic crossover features 0.03% worst case distortion, voltage protection and an integral 100Hz electronic crossover. It is designed to drive the subwoofer(s) in a bi-amplified system.

The S-80 is a 40W per channel stereo amplifier featuring reverse voltage protection, 0.03% THD and has a 10A power requirement.

For more information contact ZAPCO, 2549 Yosemite Blvd., Ste. F, Modesto, CA 95354.

Fast Reply #GK124

SPEAKER BUILDER
MAGAZINE

Advertising Representative
Chris Smith—InterMarketing Associates
12 West St., Suite 20
Keene, NH 03431 Phone: (603) 352-1725

Editorial and Circulation Offices
Post Office Box 494
Peterborough, New Hampshire 03458

All subscriptions are for the whole year. Each subscription begins with the first issue of the year and ends with the last issue of the year. A sample issue costs $4 in the US, $5 in Canada.

Subscription rates in the United States and possessions: one year (four issues) $12, two years (eight issues) $20.

To subscribe, renew or change address in all areas outside the UK write to Circulation Department, PO Box 494, Peterborough, NH 03458. For subscriptions, renewals or changes of address in the UK write to J.L. Lovegrove, Leazings, Leafield, OX8 5PG England. For gift subscriptions please include gift recipient's name and your own, with remittance. A gift card will be sent.

A Note To Contributors

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

Speaker Builder Magazine (US ISSN 0199-7920) is published four times a year at $12 per year; $20 for two years, by Edward T. Dell, Jr. at 5 Old Jaffrey Rd., Peterborough, NH 03458 USA. Second class postage paid at Peterborough, NH.

POSTMASTER: If undeliverable send PS form 3579 to SPEAKER BUILDER PO Box 494, Peterborough, NH 03458.

AUDIO CONCEPTS, no stranger to *SB*'s pages, has just issued their new 1986 catalog. First-time builders and professionals alike will find a wide range of loudspeaker kits, drivers, crossovers and accessories to update and improve your audio systems.

Since starting in a basement almost ten years ago, Audio Concepts has become known for their fast service, low prices and a helpful and an informed willingness to help customers, as well as to take suggestions about improving their kits.

If you haven't received your catalog yet write Audio Concepts, 1631 Caledonia St., La Crosse, WI 54603, or call (608) 781-2110.

Fast Reply #GK45

Rick Chinn who works for **AUDIO CONTROL** has written a nice comprehensive overview of active crossover networks presumably as a background to their new Richter Scale Series III, which is a two channel subwoofer crossover and equalizer with a measuring microphone to analyze and set bass response.

Any speaker builder who would like a copy of "Crossover Networks from A to Linkwitz-Riley" should send a #10 envelope with his or her address and 42 cents U.S. postage attached to Audio Control, Dept. SB., 6520 212th Southwest, Lynwood, WA 98046.

Dr. Richard Small, a leading figure in the development of loudspeaker system theory and of Thiele/Small parameter fame, has taken the newly created position as Head of Research for UK loudspeaker manufacturer **KEF ELECTRONICS** of Maidstone, Kent, England.

He is leaving his position as Senior Lecturer at the University of Sydney, Australia, to be responsible for developing and coordinating advanced research projects, in line with KEF's long term commitment to the improvement of transducers, associated electronics, enclosures and measuring techniques.

About This Issue

Tom Cox likes transmission lines a lot. Using small drivers (5") in multiples of four, he manages to pack a lot of range and response into a two cubic foot package (p.9).

So you wish to make a notch filter and you've done the calculations but the result is odd? **David Weems** made the same discovery. Theory disparities can be worked out though, as you will find in *Notch Filters* starting on page 24.

Len Hupp likes, as many do, the small BBC-designed LS3/5A monitor. Len has removed some of the small classic's limitations as Len details the changes for you on pp. 30–31.

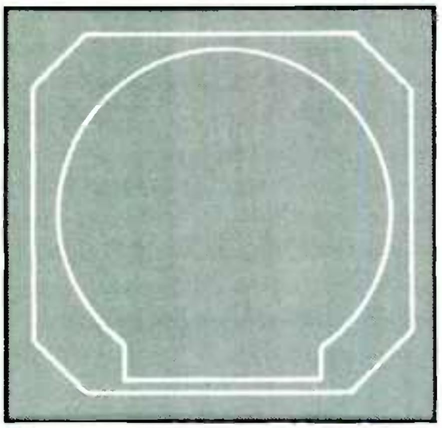
Gary Galo, ever alert to new driver developments, has put seven cone and done midrange units suitable for the transmission line through his test routine and reports on them beginning on page 32.

You will find the work of readers **Ralph Gonzalez** and **David Meraner** gracing our *Tools, Tips & Techniques* pages (40–41) and **Bob LeBeck's** clever adaptation of the late **Bert Webb's** transmission line on pp. 42–43.

Next time **John Cockroft** builds a *Mini-Dancer*, we'll do plans for *Gondor* the sub-bass woofer and **G. R. Koonce** will unveil a program for checking component specifications in passive crossovers, just to touch the high spots.

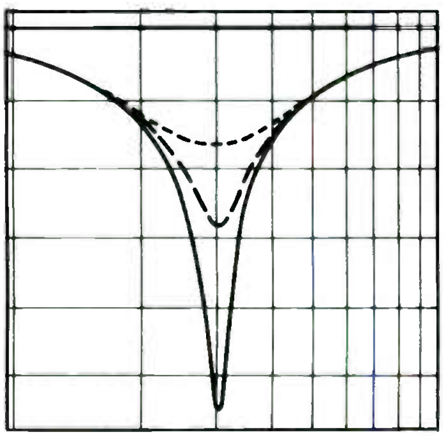
SPEAKER BUILDER

VOLUME 7 NUMBER 2 MAY 1986



11

FEATURES



26

DEPARTMENTS

14

Editorial

Bags, Burgers and Cheese

The credits following Garrison Keillor's PBS Prairie Home Companion show now always remind me that in a few quarters the idea of mass production is not a winner. In the past, one supporter seemed so unlikely, I wondered what they were doing helping to fund a Public Radio program about that nostalgia-packed mythical town of Lake Wobegon which seems permanently stuck in a 50 year old time warp. The announcer always says, "Partial funding for this program is provided by Coach Leatherware of New York City.

Well I wonder no more. It seems that a Manhattan couple has been directing a small garment district factory which makes one of the world's most sought-after handbags. Among those who know a good bag when they see one, the Coach is reportedly the best. It is functional, high quality, and the company can't keep up with the demand, even with 200 employees.

Miles and Lillian Cahn owned Coach Leatherware for the last 24 years. Two years ago, planning to retire, the Cahns bought 300 acres in Gallatin, NY two hours from the city. Six months ago, bored with retirement, they bought 50 French Alpine goats, hired two world class experts on goat cheese and started a farm. The milk goes into a delicacy called chevre. It looks as though the Cahn's cheese making will be even more successful than their leather bags—weekly sellout revenues were above $3,000 a week at last report. Experts are already saying their product rivals the famous French one.

One of the Boston TV stations carried a small feature story last week about a butcher who, having run a successful meat market for 30 years, retired to Florida. Bored with that after a few years, he returned to his home town and rented a small store, turning it into a hamburger shop. Twenty years later, at 80, he is still running the business nine hours a day, six days a week. When the reporter asked a row of customers perched on stools why they come to his hamburger stand, each of them stopped munching just long enough to say a few words. One man's answer summed up all their sentiments: "It's the best hamburger there is," he said.

My purpose here is not to denigrate the miracles of mass production or even of mass marketing. Such efforts have improved the quality of what many people can afford and raised the nutrional level of many. But if you want something really good, it cannot be mass-produced. And that applies to handbags, goat cheese, hamburgers and to loudspeakers. The lines of relationship between the person whose understanding and care are vital to the product cannot be stretched for any great distance. Most of those firms producing the drivers speaker builders are using are relatively small. Their production lines are diminutive.

Readers of this magazine who are enjoying this gourmet feast of loudspeaker design and construction know what that close intimacy feels like and how satisfying it can be. The pleasure of that hand-to-hand contact with this technology and our collective success with it is making us a progressively better informed group. And we are making better loudspeaker systems. The distance between the caring, knowledgeable mind and the finished system is very short.

If you are, as yet, only an armchair participant, I strongly encourage you to take yourself in hand and build a new design, even if it is that pair of simple extension speakers you have wanted for your bedroom all these years. Once you take that first step, you will doubtless move firmly and without hesitation to the next one, convinced that the really ultimate system you have always dreamed of is not impossible after all.

This is a fine time for rediscovering individual craftsmanship. The tools are better, the materials are better, and the way in which we are pooling our experience in this publication is making the design information better. If you have been at all dubious about your ability to make a better system for yourself, be doubtful no longer. It's time to join the fun.—E.T.D.

2 x 4 TRANSMISSION LINE

BY THOMAS E. COX

Driver manufacturers have responded to the demand for better automobile audio systems with a variety of high quality, wide range and relatively low cost drivers in smaller sizes that offer fresh potential for home speaker designers.

In the 2 x 4 design, *Photo 1*, I have combined two columns (one of drivers, and one of transmission line ports) with four transmission lines (one for each woofer). Using four-inch drivers with their high, free air, resonant frequency, enables the entire system to be packaged in a two cubic foot enclosure.

DESIGN GOALS. My design goals for the 2 x 4 were:

- transmission line woofer loading
- cylindrical sound radiation pattern
- restricted vertical sound distribution combined with uniform horizontal distribution
- clean, useful output from 50–5kHz without crossover networks
- compact size
- and no exotic cabinet making skills required.

DESIGN CHOICES. *Transmission line length.* To design the transmission line I constructed a "one time" experimental line as described in my *SB* 4/85 article, p. 9. I used data derived from the experimental line to determine an optimum line length for the 2 x 4. However, if manufacturer's lot variations for the driver resonant frequencies are too wide, the line length should be modified. I'll cover this in more detail in the construction section.

Cylindrical sound radiation pattern. The normal radiation pattern of a vertical column array of drivers is a cylinder with its axis through the drivers. The 2 x 4 also has a second

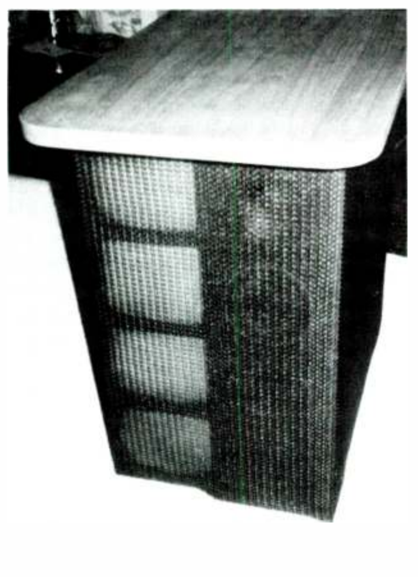


PHOTO 1: The completed 2 x 4, compact and versatile.

column array of transmission line ports, which radiate low frequencies passing through the wool/polyester low pass filter.

Restricted vertical/uniform horizontal sound distribution. The design goal of restricting vertical radiation was based on Queen's article in the *AES Loudspeakers.*[1] Queen describes experiments that show clarity and image enhancements by restricting speaker sound radiation to the floor or ceiling, combined with uniform horizontal distribution at high frequencies.

Schroeder[2] noted the same general effect when he found a high correlation between high ratios of horizontal/vertical reflection intensity and listener preference for specific concert halls. Vertical radiation is normally restricted by closely spaced drivers mounted in a column. This is due to phase differences between drivers when arranged as a line source. The details of this vertical restriction are explained in Crowhurst's book.[3] Since the 2 x 4 drivers are arranged in a closely spaced vertical column, sound radiation to the floor and ceiling is restricted.

Klepper and Steele's article in *AES Loudspeakers,*[4] shows that line sources (column arrays) have non-uniform horizontal sound distribution patterns when all drivers in the line source are identical.

The article describes improvement in the uniformity of horizontal distribution, by reducing the effective length of the line source (column) with increasing frequency. This is accomplished in the 2 x 4 by using coaxial tweeters in the middle drivers only. Therefore as input frequency increases, the effective height of the column decreases, from the full four driver height, to only the two middle tweeters.

50–5kHz without crossovers. The design goal of 50–5kHz, without crossovers, is based on my belief that effective midrange coverage, of all musical instruments' *fundamental* frequencies, results in more clarity and definition. With the exception of pipe organs, that means a frequency coverage of roughly 30–5kHz.

While the 2 x 4 does not reach the low end target, it does cover the critical 100–500Hz region as well as the usual midrange (500–5kHz) without crossovers. *Figure 1* shows how I approached the low frequency response goal. Free air resonant frequency for one of the European Loudspeakers of America 4502s was 93Hz. When I mounted the ELA woofer in its individual transmission line, the low frequency impedance peak dropped to 74Hz; and in tandem with the other three woofers in a column array, the

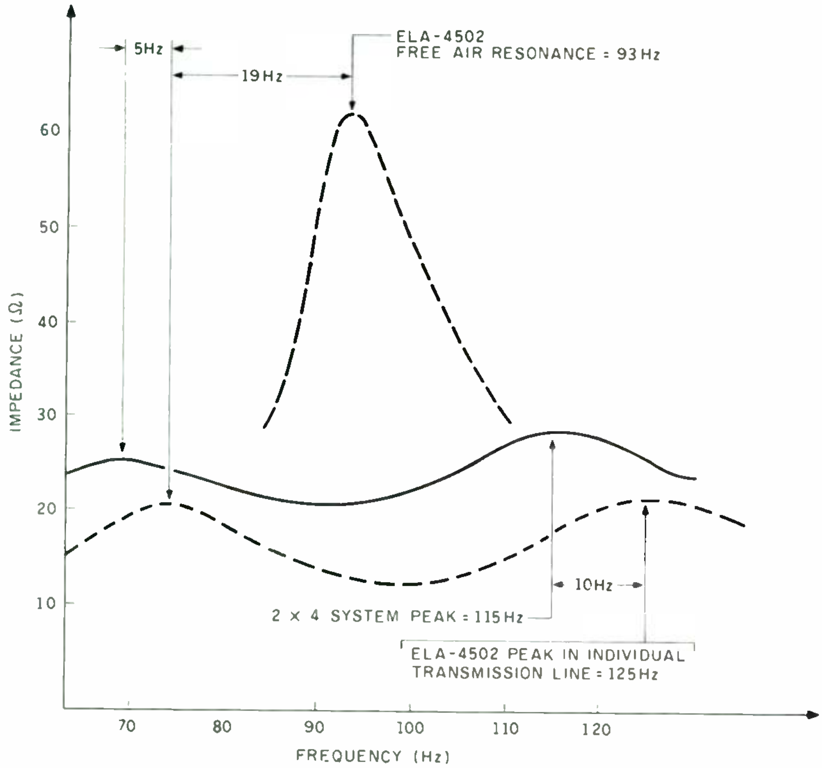


FIGURE 1: Single driver and system impedance/frequency graphs. This shows how the low frequency response goal was approached.

system peak dropped further to 69Hz. By this point we are within hearing distance of the 50Hz low frequency goal.

One rule of thumb related to high frequency beaming of cone type drivers says: tweeter crossover should occur before cone diameter equals a full wave length. For the 2 x 4 woofers, with an effective cone diameter of 3¼ inches, this works out to approximately 4kHz, which is close to the design goal of 5kHz.

Compact size. Since the 2 x 4 was planned as a present to one of my daughters, who rearranges the furniture in her room at unpredictable intervals, size was a major consideration.

Ease of construction. The final design goal called for a small challenge to my limited cabinet making tools and experience. Surfaces requiring a furniture finish were minimized, with most edges either square or at 45 degree angles.

CONSTRUCTION. *Figures 2* and *3* show the basic construction details in front, side and top views. All horizontal dimensions were based on a driver resonant frequency range of 80–90Hz. I have received drivers, of the type used in the 2 x 4, from different manufacturing lots with measured free air resonant frequencies that ranged from 78–106Hz. If the average value of the four drivers is 95Hz or greater, I would suggest reducing the appropriate horizontal dimensions by one inch (shortening the transmission line length by two inches).

The enclosure is made from ½-inch birch surfaced, hardwood plywood, but high density particle board can be substituted if you prefer. The cutting guide, *Fig. 4,* shows the layout for the 2 x 4 enclosures. With the 2 x 4's small dimensions, and cross braced construction, ½-inch plywood should provide sufficient internal damping, and external stiffness to minimize spurious low frequency sound generation due to panel vibration. Rap your knuckles on the outside of the completed enclosure and you should hear a dull, well damped thud.

Notice that *Fig. 2* shows only the left-hand unit in a stereo pair. The right-hand unit has the same dimensions, but the driver panel and the panel mounting blocks are on the left side. Looking at a stereo pair from the front, driver columns are on the inside (closest to each other), and the line ports are on the outside.

A helpful construction tip: label two edges of all the pieces used in the enclosure (i.e., front, inside, top, bottom etc.), and you will make fewer assembly mistakes.

CORE UNIT CONSTRUCTION. Begin with the horizontal and side panels. Draw pencil lines on the vertical side panels to locate the ten, ⅞ by ¼ by 14 ⅛-inch support mouldings for the horizontal panels. Pencil layouts are useful in each assembly operation to locate fasteners and gluing areas. One end of each support moulding should be located ¾ of an inch from the front edge of the enclosure. This provides a stop for driver panel mounts and the foam speaker grille inserts at each line port.

Cut the support mouldings to size, and fasten them to the side panels with ⅝-inch brads and wood glue. When gluing be sure to clean off excess glue where later assembly requires square internal corners. (*On outside surfaces let the glue beads harden, and then chip off, to prevent glue from getting into the wood's pores if you intend to use a natural finish on the birch.—Ed*)

Next cut 12 pieces of ⅞ by ¼-inch moulding to 10¾ inch lengths. This moulding will attach the center vertical panels to the top and bottom of the three internal, horizontal panels. After the first moulding in each pair is nailed and glued, use a center vertical panel as a guide to determine proper spacing between the mouldings.

Place an index card between the vertical panel and the installed moulding (the index card allows about one one-hundredth of an inch tolerance in the slot width to make final assembly easier). Now tightly fasten the second moulding next to the vertical panel with ⅝-inch brads and glue. Notice also that the vertical center panels are only 4¹⁄₁₆ inches high to allow ¹⁄₁₆ inch tolerance for easier assembly.

After the support mouldings have been fastened, assemble the side panels to the horizontal panels with 1-inch brads and glue. This step requires some speed and confidence in assembly. Partially driven brads, plus pencil line layouts, help define glue areas and panel alignment. The brads are not intended to provide much structural strength, so be sure to liberally glue support mouldings, side panels and the horizontal panel edges.

To avoid a "leaning tower" effect place the core unit upright on a flat surface, before the glue hardens, and check both front and sides for verticality with a large square. Place a piece of sturdy plywood on top of the core

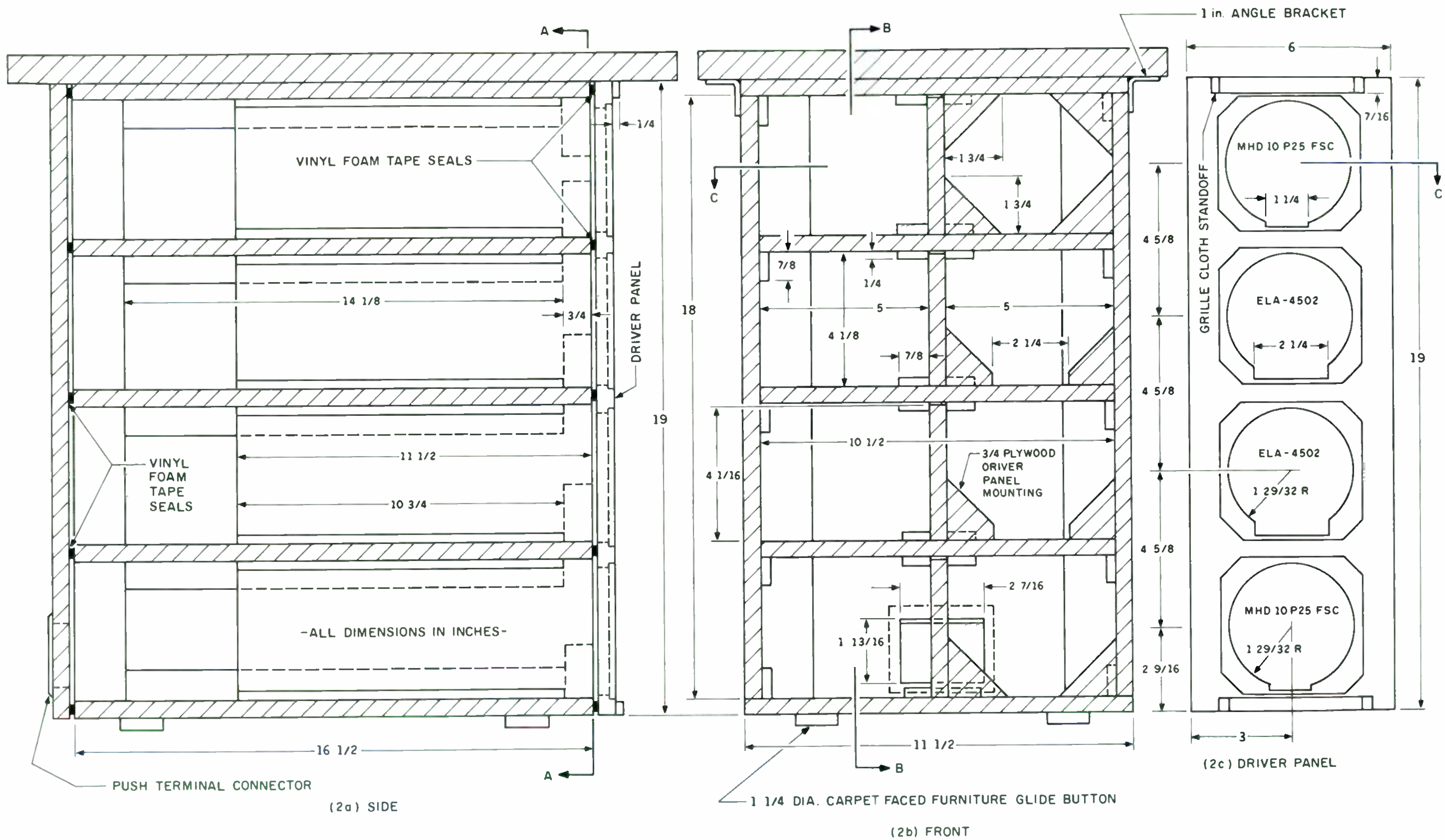


FIGURE 2: Basic construction details: *a*, side view, with push terminal connector; *b*, front view shows driver panel mounting; c shows driver panel with 1 $^{29}/_{32}$ inch radius cutouts.

unit, and some weight on top of that, to insure that the core assembly's front and sides will remain squared while the glue hardens.

Corner reflectors are made from 1⅝ by 1⅝-inch clear white pine stock (called Baluster by lumberyards). A long diagonal saw cut is tricky, so I bought the corner reflector material at a local millwork firm and they made the cut for me. This produces two lengths of triangular cross-section material.

Each 4¹⁄₁₆-inch corner reflector is fastened with two 1¼-inch # 8-wood screws, wood shims (I used tongue depressors) and wood glue to provide more structural stiffness. The partially completed core assembly is shown in *Photo 2*.

Cut four more ⅞ by ¼ by 10¾-inch center panel support mouldings and fasten them to the inside surfaces of the top and bottom panels. Assemble the top and bottom panels to the core unit with 1-inch brads and glue. Spread glue in the bottom slot for one of the vertical center panels. Add glue to the bottom edge of a vertical panel, and slide it into position using the ¹⁄₁₆ inch tolerance to keep as much glue in the slot as possible. Clean excess glue from the corners where driver panel mounts will be assembled later.

Attach the bottom edge of all vertical center panels, and let the glue dry overnight. Then turn the core unit upside down and glue the top edges of the vertical panels, using the slot tolerance to force glue between the panels and mouldings. Seal the gaps left between the vertical center panel's top edges and the horizontal panel with wooden shims and glue (only the exposed front ¾ inches need be shimmed).

DRIVER PANEL MOUNTINGS. Next add the driver panel mountings (*Fig. 2b* and *Photo 3*). These triangular pieces are cut from a 1¾ by ¾ by 18-inch strip of plywood (solid lumber could be used) by alternating 45 degree and square cuts. Four driver panel mounting blocks are cut down to 1⅜ inch on one side, to allow a 2¼ inch opening for the ELA 4502 driver terminals.

Mount the triangular blocks with 1-inch brads and glue against the ends of the horizontal support mouldings (where available). Remember, these driver mounting blocks will be on the right-hand side for one enclosure, and on the left-hand side for the other mirror image enclosure. Drive the brads partially into the mounting blocks at an angle toward the front of the enclosure, so they can be driven home easily when glued in place.

Fasten a 19-inch piece of ½-inch, half-round moulding to the front edge of the port opening side panel, to provide a rounded corner for the grille cloth covering. The core unit is now complete up to the stage shown in *Photo 3.*

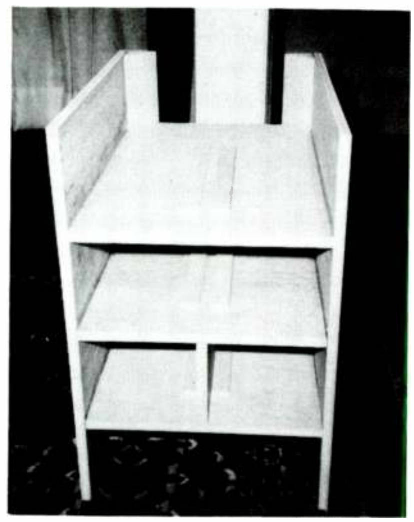


PHOTO 2: The partially completed core.

DRIVER PANELS. The driver panels are also constructed from ½-inch birch plywood (*Fig. 4*). The vertical outside edges are quarter rounded for smoothness when stretching the grille cloth, and some diffraction reduction (*Fig. 3*), with a ⅜-inch rounding router bit. Lay out the center of each driver hole and cut 1²⁹⁄₃₂-inch radius holes (*Fig. 2c*).

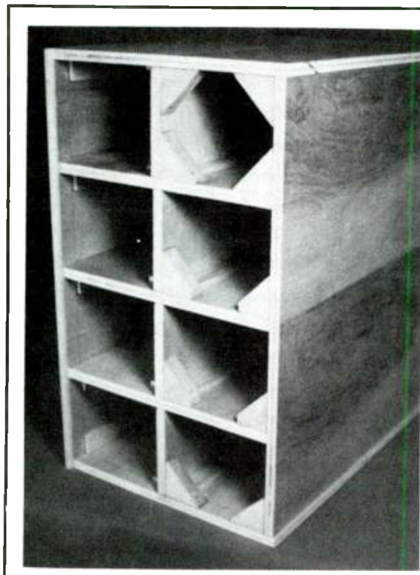


PHOTO 3: Core unit with triangular driver panel mountings.

After the holes have been cut, and notches made for driver connection terminal clearance, place the rear of the driver through its hole until the mounting flange is seated against the driver panel.

Make sure that mounting flanges cover and seal the terminal notches by at least ³⁄₃₂ of an inch. Trace the driver flange outline in pencil on the driver panel. Rout these patterns to a depth of ³⁄₁₆ of an inch so the driver flange will fit flush into the routed recess. Do this for each driver.

Grille cloth standoffs, which provide clearance between grille cloth and drivers, are made from a 4-inch piece of ⅞ by ¼-inch moulding. Split this piece down the middle with a pen knife to form two ⁷⁄₁₆ by ¼ by 4-inch strips. The standoff ends are also rounded to form a smooth curve for the grille cloth. Fasten the standoffs to the driver panel with ⅝-inch brads and glue.

DRIVER MOUNTING. The top Audax driver is mounted to the driver panel mounts with four 1-inch # 6-pan head, self threading screws, which extend through clearance holes in the driver panel. The other drivers are fastened to the driver panel with two ⅜ inch # 6-pan head, self threading screws in the top driver mounting holes and two 1-inch # 6 screws, through clearance holes into the driver panel mounts.

To provide a secure seal for the bottom three drivers, and an attachment for the top driver, run a bead of black silicone gasket material in the routed panel areas. Temporarily fasten the drivers to the panel using only 6/32 x 1-inch machine screws and nuts, and ⅜-inch #6-self threading screws. Do not fasten too tightly. The object is to pull up the fasteners so the gasket material is uniformly compressed, but not squeezed out.

The gasket material between the driver flange and the panel helps dampen driver frame vibrations before they reach the driver panel. A view of the assembled panel is shown in *Photo 4*.

DRIVER PANEL WIRING. *Figure 5* is the driver wiring schematic, and *Photo 4* shows a view of the completed wiring. Postpone final woofer to woofer wiring until transmission line matching (to be described later) is completed. Mount solder lug terminal strips and small, screw mounted, wir-

Continued on page 14

MADISOUND SPEAKER COMPONENTS

8982 TABLE BLUFF ROAD
BOX 4283
MADISON, WISCONSIN 53711
PHONE (608) 831-3433

MADISOUND is now stocking PERFECT LAY WINDING audio inductors from Solen Engineering. These are audio grade inductors using 14 gauge wire with the following specifications:

* Perfect Lay Hexagonal Winding. Winding Space Factor: 86.7%
* Oxygen Content: Less than 200 parts per million on surface.
* Insulation: 130 degree centigrade Single coating Nylon-Polyurethane.
* Computer Optimized Coil Dimension.
* Encapsulation: Varnish dip coating with 4 nylon ties.
* No Saturation distortion: Test voltage 1500 VAC.
* No Hysteresis distortion. Test voltage 1500 VAC.
* Inductance tolerance: within 1% of value listed.
* Conductivity: Better than 101.5% of National Electrical and Manufacturing Association (NEMA) standard sample.
* Wire Diameter: .064 inches; 1.63 mm. High Purity Annealed Copper.

PERFECT LAY WINDING AIR CORE INDUCTORS

Inductance	Resistance	Size		Price	Inductance	Resistance	Size		Price
L-Mh	DC ohms	Height	Diameter	Each	L Mh	DC ohms	Height	Diameter	Each
.22	.08	.56 in.	2.25 in.	$3.70	1.52	.29	.88 in.	3.5 in.	$11.70
.33	.10	.63	2.5	4.60	1.82	.30	.88	3.5	12.40
.47	.13	.63	2.5	5.70	2.0	.31	.88	3.5	13.00
.56	.15	.63	2.5	6.40	2.22	.33	.88	3.5	13.80
.62	.16	.63	2.5	6.80	2.47	.36	.88	3.5	14.60
.68	.17	.75	3.0	7.10	2.75	.39	.88	3.5	15.30
.75	.18	.75	3.0	7.40	3.0	.42	.88	3.5	16.00
.82	.19	.75	3.0	7.80	3.3	.45	1.0	4.0	16.80
.91	.20	.75	3.0	8.10	3.7	.49	1.0	4.0	17.50
1.0	.21	.75	3.0	8.50	4.0	.50	1.0	4.0	18.30
1.1	.23	.75	3.0	9.00	4.5	.56	1.0	4.0	19.60
1.22	.26	.75	3.0	9.80	5.0	.59	1.0	4.0	22.00
1.47	.28	.75	3.0	11.00	5.5	.63	1.0	4.0	23.40

Values between sizes listed are also available. Add 10% to cost of value larger than your requirement.

Madisound also stocks audio standard inductors as well as the popular SIDEWINDER inductors

FAST REPLY #GK20

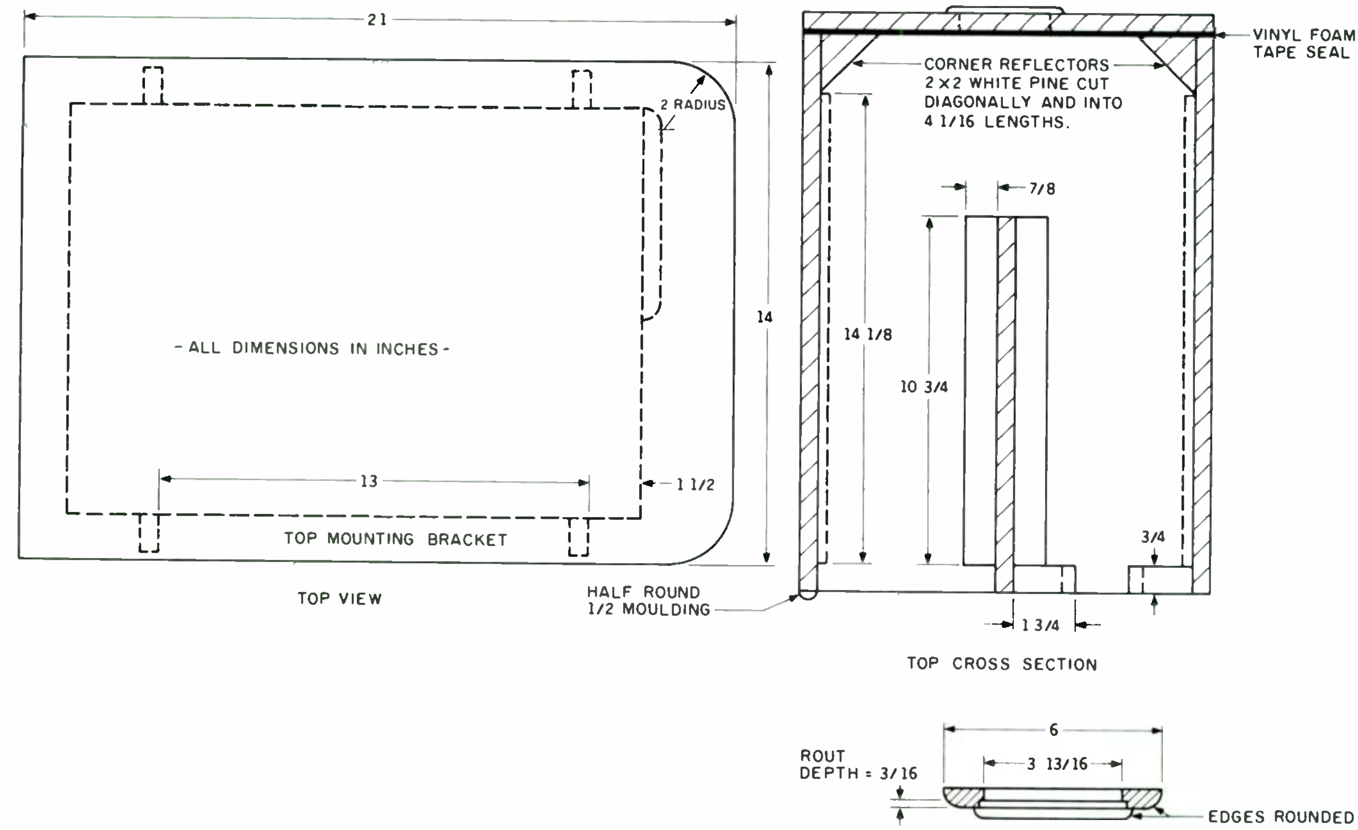


FIGURE 3: Basic construction details from top view and cross sections.

Continued from page 12

ing cable clamps (available from industrial electronics parts suppliers), at appropriate locations to provide support and termination points for wires and components.

For the driver panel use # 20-gauge stranded wire, and to connect the push terminal connector on the rear panel use #16-gauge stranded wire. Draw a pencil layout of the driver panel mounts on the back of the driver panel (*Photo 4*). Use this layout to make sure to locate the terminal strips so neither the strips nor the capacitor, nor resistor, will be obstructed by the panel mounts when assembling the driver panel to the core unit.

Manufacturer's specifications show the Audax drivers as 6dB less sensitive than the ELA 4502s. To balance these differences the series parallel connection, shown in *Fig. 5*, provides roughly twice the current to the less sensitive Audax units.

To provide more equal driver currents to both woofers and tweeters of the ELA 4502s, a 7.5Ω, 10W resistor has been placed in series with the parallel connected tweeters. The 15μF capacitor serves as a low frequency blocking capacitor to limit low frequency current to the tweeters.

The negative leads of both the tweeter and the woofer are combined in a single terminal on the ELA 4502.

REAR PANEL ASSEMBLY. Install the speaker push terminal connector in the rear panel as shown in *Fig. 2a*. Use silicone gasket material to seal the terminal connector flange to the rear panel. Tighten the mounting screws, and let the silicone gasket material ooze around the edges. When the silicone material has fully hardened, the surplus material can be cleanly and easily trimmed with a razor blade.

Install closed cell, vinyl foam, pressure sensitive tape (sold as weather-stripping) to all rear core panel edges (*Fig. 2a*. Fasten the rear panel to each corner reflector with eight 1 ¼-inch, # 8-flat head steel screws, located to avoid interference with the side panel screws. Tighten the screws alternately so the vinyl foam seal is uniformly compressed to roughly $^1/_8$ inch.

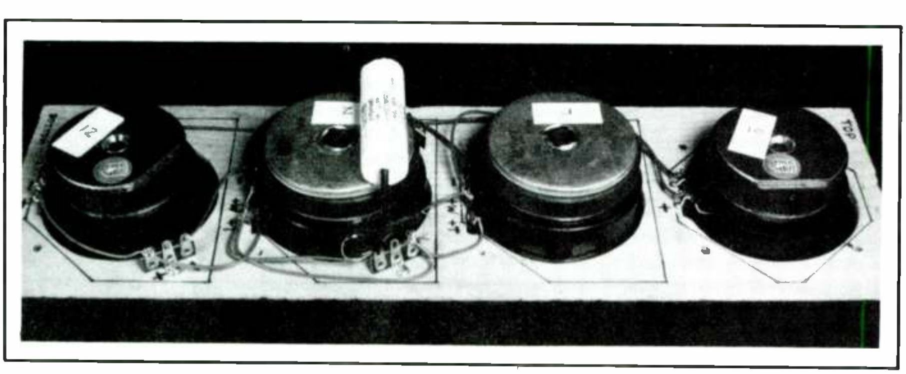


PHOTO 4: Assembled driver panel components with gasket material between flange and panel to help dampen driver frame vibration. Note the pencil lines to avoid final assembly parts obstruction.

PAINTING. Paint all external surfaces of the completed core unit and driver panel with two coats of flat

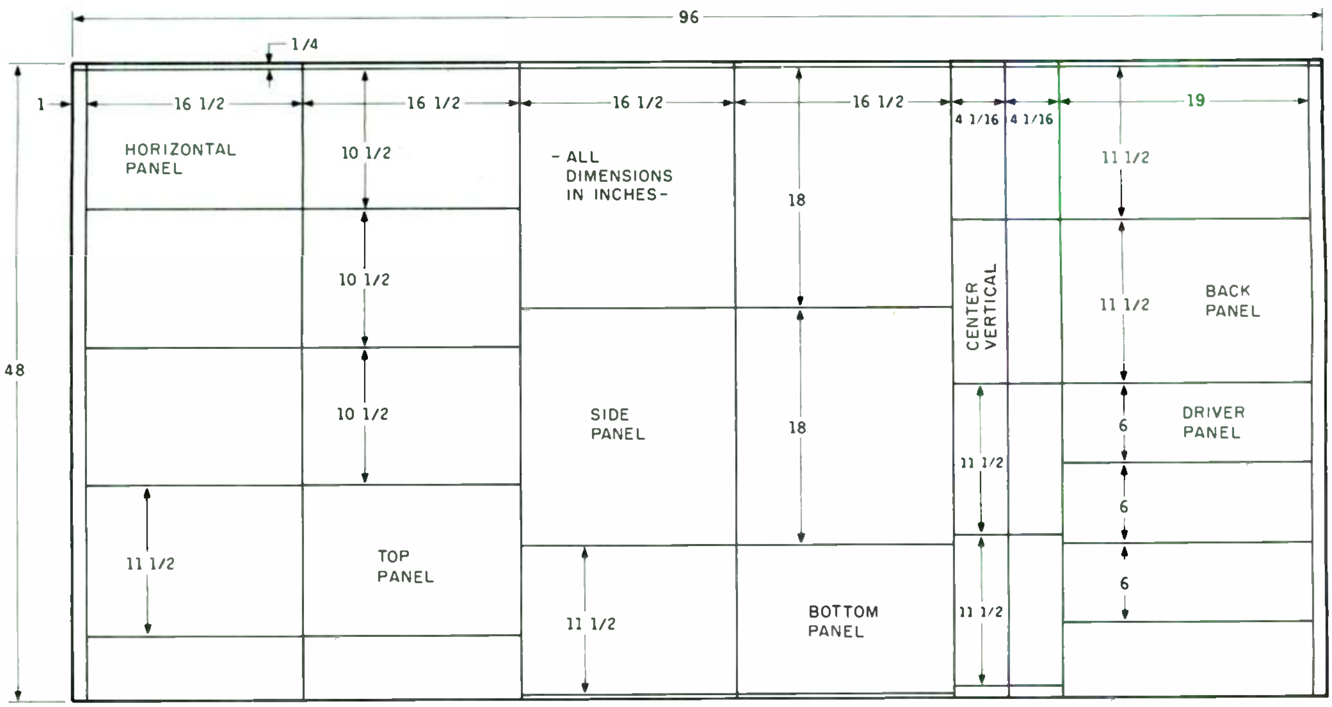


FIGURE 4: Cutting guide for the 2 x 4s. I used one 4 by 8 foot sheet of Birch faced plywood.

black paint. Extend the paint an inch or two into the port ends of the transmission lines. This makes the ports and drivers invisible behind the grille cloth.

DRIVER PANEL SEALING. On each opening in the core unit mount the same type of vinyl foam tape used for the back panel (*Fig. 2a*). This sealant should be placed carefully so that each driver affects only its own transmission line. The vinyl tape also seals the wiring between drivers when the driver panel is tightened.

DRIVER TESTING. *Figure 6* shows the test setup I used for driver matching, and the final tuning of the transmission lines. Although the 2 x 4 system could be built without this equipment, I strongly recommend some means of impedance measurement for optimum performance.

The signal generator should provide sine wave output over a frequency range of at least 40-200Hz. An electronic multimeter, with flat frequency response over the range of frequencies to be covered in the test, is the preferred AC voltmeter reading.

Since the drivers are relatively low cost units, I purchased five of each type for closer resonant frequency matching. All ten were broken-in overnight, using an all night music station as the signal source.

After break-in each driver was labeled, and suspended in air with string through the driver mounting holes. Each driver was connected to the test setup in *Fig. 6*, and signal generator frequency varied from 50-150Hz. The drivers (woofer sections only for ELA 4502) were labeled with the frequency at which the AC voltmeter showed its highest reading.

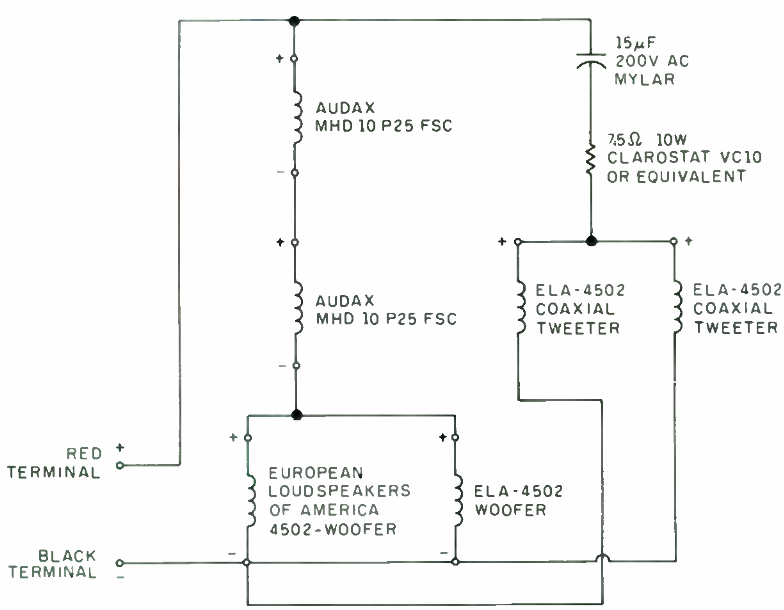


FIGURE 5: Driver wiring schematic. The series parallel connection provides roughly twice the current to the less sensitive Audax drivers.

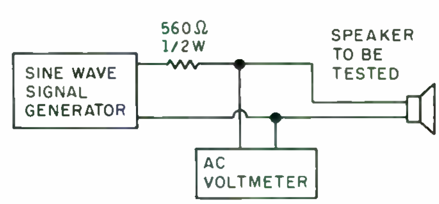


FIGURE 6: My impedance peak test setup is strongly recommended for optimum performance.

This is the free air, resonant frequency for that driver.

When all the drivers had been tested I selected sets of two Audax, and two ELA 4502s, which had free air resonant frequencies as closely matched as possible.

LINE STUFFING PROCEDURES. One and one-half ounces of long-hair wool is required for the driver half of one transmission line. The long-hair wool must be combed out to a light fluffy texture, and sprayed with a mothproofing solution before being used as stuffing. *Photo 5* is a rear view of a stuffed transmission line.

I removed the panel of the core unit and placed handfuls of wool as uniformly as possible, from the driver panel mounts, to the rear of the driver half of the transmission line. The other half, which ends in the line port, was stuffed with Poly-Fil brand, Extra-Loft batting, available at sewing supply stores from 45-inch rolls. I cut the large piece into 7 by 45 inch strips to get the initial test density of 8 ounces per cubic foot. I used 1¾ by 11 ¼-inch strips, which adjust density by 20 percent.

Black foam speaker grille material, forced fitted into the last ¾ inch of the transmission lines, provides an acoustically transparent but visually opaque port opening (*Photo 6*). By using wool for the first half, you gain additional scattering and absorption as the rear directed high frequency sound waves go through the wool/polyester interface.

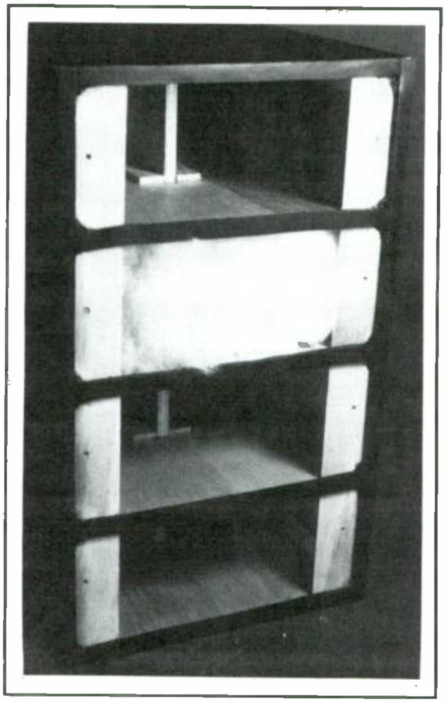


PHOTO 5: Rear view of core unit with one stuffed transmission line.

MATCHING THE LINES. An option available to the speaker builder, which is not available to the purchaser of an assembled speaker, is the chance to fine tune the speaker system. In the rush to meet a gift deadline, my first 2 x 4 unit was assembled without tuning individual transmission lines. The second unit with individual tuning performed so much better, I later disassembled the first unit and tuned each line individually.

Tuning involves connecting only one driver at a time to its transmission line, with all the other drivers in the column disconnected. Therefore I did not wire the drivers to each other until tuning was completed. I used long alligator clip leads for temporary connections between a driver and the impedance/peak test equipment.

With one of the drivers connected to the test equipment, the driver panel was assembled to the core unit to seal that test driver to its transmission line. Then the input signal frequency was varied from 50–200Hz. The impedance peaks for that driver/transmission line combination were noted. I tested each of the other three driver/transmission line combinations separately using the same method.

The lowest frequency peaks for both Audax and ELA units were quite close, but the higher peaks were separated by about 10Hz.

Line tuning involved removing a small portion of the polyester material from some of the lines until the higher peaks in all lines matched as closely as possible. I made all changes to the polyester side without making any changes to the wool side. I obtained the best results by removing the polyester material to achieve matching lines, rather than increasing the original 8 ounces per cubic foot density.

When all tuning was completed I soldered the final wiring between drivers. The vinyl foam seal between the driver panel and the core unit had to be replaced. The sealing and resealing, around the temporary test leads used for tuning, left permanent depressions in the seals. I soldered the leads from the push terminal on the rear panel to the driver panel wiring and the driver panel fastened to the core unit, with the seals compressed to approximately $^1/_8$ inch.

PHOTO 6: The completed enclosure before final finishing.

EXTERIOR FINISHING. Acoustically the 2 x 4 was now complete, as shown in *Photo 6*, but I wanted some finishing touches to make it as pleasing to the eye as it was to the ear. One possibility I considered was using the core units as the end supports of a shelved wall unit. However, since portability was a design goal, I decided to finish them as shown in *Photo 1*.

First I wrapped a piece of black plastic grille cloth around the front and sides. The grille cloth extended slightly over the top, bottom and back of the core unit. I fastened it to the top, front edge with a staple gun. The exposed staples on the top edge are concealed by the finished top. Then I pulled it taut over the face, and stapled along the bottom.

I cut notches in the corners so the grille cloth would extend tightly along the sides of the core unit. I notched the back corners, and stapled the remaining grille cloth material to the rear panel.

I mounted furniture guides, faced with carpeting, on the bottom of the core unit to provide isolation and protection from my floor.

For the finished top I used cherry ¼-inch veneer plywood, glued to another sheet of plywood, for a total thickness of $^7/_8$ inch. I covered the top's edges with 1-inch cherry veneer strips, which can be glued with an ordinary electric iron and trimmed with a razor knife. I finished the veneered surfaces with a penetrating oil as described in

Continued on page 18

Continued from page 16

Allen's excellent book on wood finishing.[5]

SPEAKER PLACEMENT. Column speakers are far less sensitive to room placement, and are often recom-

Continued on page 58

MATERIALS LIST
FOR ONE ENCLOSURE

Wood Materials

- 2 Side panels 18 x 16½*
- 3 Horizontal panels 10½ x 16½
- 1 Top panel 11½ x 16½
- 1 Bottom panel 11½ x 16½
- 4 Center vertical panels 4¹⁄₁₆ x 11½
- 1 Back panel 19 x 11½
- 1 Driver panel 19 x 6

The above pieces were cut from one half of a 4 x 8-foot sheet of ½-inch Birch plywood (See Cutting Guide*)*.

- 10 Horizontal panel support strips ⅞ x ¼ x 14⅛ *(lattice moulding)*.
- 16 Center vertical panel support strips ⅞ x ¼ x 10¾
- 8 Corner reflectors 4¹⁄₁₆ x 1⅝ *(right triangle)*
- 10 Driver panel mounts ¾ x 1¾ *(right triangle)*

Other Materials

- 1 8 oz. dispenser of Tite Bond Wood Glue
- 1 Small box of 1 x 17-gauge galvanized brads
- 24 1¼ x #8-flat head steel wood screws
- 10 1 x #6-pan head self threading steel screws
- 9 ⅜ x #6-pan head self-threading steel screws
- 1 Container of black silicone gasket material
- 1 Package (17ft) closed cell vinyl foam self-sticking tape ⅜ x ¼
- 1 Small package ¼ x ⅜ 3 solder lug terminal strips (H.H. Smith Inc. #864)
- 1 Color coded 1¾ x 2⅜ rectangular flush-mount push type speaker wire terminal (Madisound)
- 1 Package black sculptured foam speaker grille (Radio Shack #30FM)
- 1 8 oz. can flat black enamel
- 6 Ounces long-haired wool (J. Ebbert)
- 6 Ounces Poly-Fil Extra-Loft batting
- 1 Spray can mothproofing
- 4 Carpet bottom furniture guides

* *All dimensions in inches.*

Drivers

- 2 Audax MHD10P25FSC 4Ω nominal impedance
- 2 European Loudspeakers of America Model 4502 (Drivers available from Madisound)

Electrical Components

- 1 15µf 200V AC mylar capacitor
- 1 7.5Ω 10W resistor (Clarostat VC-10 or equivalent)

A Structured Tuning Method

Here is a structured method for tuning an array of individual transmission lines (TLs), which provides excellent matching with a minimum of "false starts."

I stuff the driver half of each TL with 1½-ounces of long haired wool. I stacked 1½-ounces of polyester batting (Poly Fil Extra Loft®), cut into 5- by 16-inch strips, in the port half of the top TL, and inserted the port end piece of acoustically transparent foam to complete stuffing the line.

With all the polyester strips in place, I measure the driver impedance peaks for that line/driver combination. Then I remove the strips one at a time, and measure the frequency changes in the impedance peak. The same process is repeated for each of the four TL/driver combinations in the array. *Table 1* shows the measurements for one Audax/TL combination and one ELA 4502/TL combination in hertz with the number of strips used. You can determine the number of strips for the best match by comparing tables for each line in the array.

Table 2 shows the impedance peak frequencies for all four TLs before matching (six polyester strips for each TL's port half) and after matching (four strips for each Audax driven TL, and two strips for each ELA 4502). After I adjust the number of polyester strips, I match the high frequency peaks and the low frequency peaks for both the Audax set and the ELA set. In all cases, matching is accomplished by reducing the number of polyester strips. □

TABLE 1

	AUDAX		ELA	
Strips	**Low**	**High**	**Low**	**High**
6	63	110	79	106
5	60	110	77	106
4	60	115	77	109
3	59	115	77	114
2	60	120	73	116
1	60	121	73	116
0	61	122	73	122

TABLE 2

	Before		After	
	Low	**High**	**Low**	**High**
Top Audax	63	111	59	115
2nd ELA 4502	79	106	73	115
3rd ELA 4502	none	106	72	115
Bottom Audax	75	114	59	115

FREE VOLUME SUBWOOFER SYSTEMS

BY THOMAS L. CLARKE

The emphasis in many professional audio installations today is on high-level music reproduction with substantial bass content. The new digital program sources provide program material with considerable energy in the lowest octave (20–40Hz). The usual approach to reproduce these sounds uses a specially designed subwoofer, crossed over to conventional drivers.

For reliable high-level sound reproduction the subwoofer should be efficient enough to reduce power requirements and voice-coil heating. If the subwoofer can be made as efficient as the other speakers in the system, you can use passive crossovers to avoid the expense of an electronic crossover and separate amplifier. The price for efficient low-frequency reproduction, according to the laws of physics, is size.

Richard Small established a relation between efficiency, enclosure volume and cutoff frequency, which says that two enclosures of the same type and efficiency have linear sizes proportional to the wavelength they reproduce. *Figure 1* shows Small's relation, and illustrates the proportions between wavelength and enclosure size. The truth in Small's relation is obvious for horn-type enclosures, but holds for any type of enclosure. Thus, an enclosure designed to reproduce 20Hz must be twice the size and have eight times the volume of one designed to reproduce 40Hz.

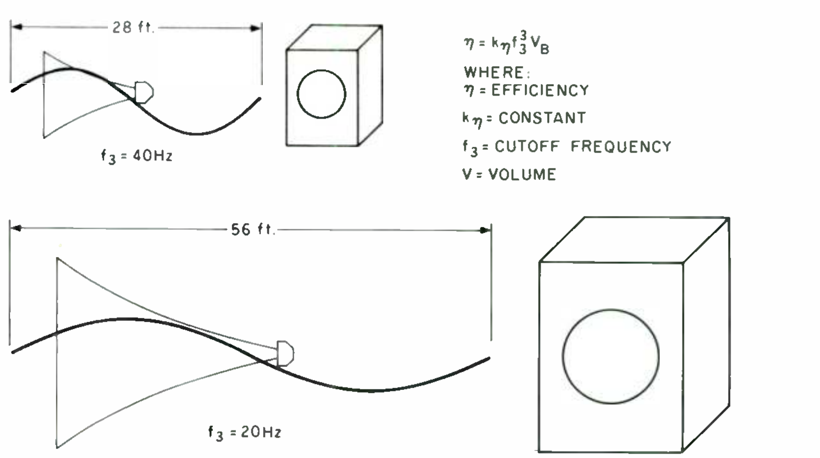


FIGURE 1: An illustration of Small's relation between enclosure size, cutoff frequency and efficiency.

The Altec Model 8182 system is a good example of the tradeoff. It is a moderately efficient (92.7dB 1W SPL at 1 meter) vented box design, with a cutoff frequency of 23Hz. It requires a substantial 24 cubic foot internal volume to achieve this performance. The maximum SPL from the 8182's 18-inch driver is about 112dB in a 3k cubic foot room. Larger rooms require multiple systems of probably unacceptable size.

FREE VOLUME. While the laws of physics cannot be broken, they can be circumvented. One way is to use existing free spaces, such as a closet, as part of the enclosure volume. A low-resonance frequency driver mounted in the closet wall is the simplest way to use such volume, but the resulting "infinite baffle" does not load the driver well acoustically, which results in poor power handling. A better technique is based on a type of enclosure called an augmented passive radiator (APR) (U.S. Patent #4076097).

Figure 2 shows an APR enclosure made from a vented box or an equivalent conventional passive-radiator system. The passive-radiator cone is enlarged into a donut shape, and a diaphragm placed across the hole of the donut is baffled by another chamber. If the passive-radiator cone is transformed further it leads to a convenient form where the APR system uses two cones joined apex-to-apex. The box volume and system cutoff frequency can be kept constant in this progression by going to a higher compliance driver. The other driver characteristics remain unchanged.

The donut hole can be baffled by any convenient volume. A rigid enclosure is not required because the system's response does not depend on

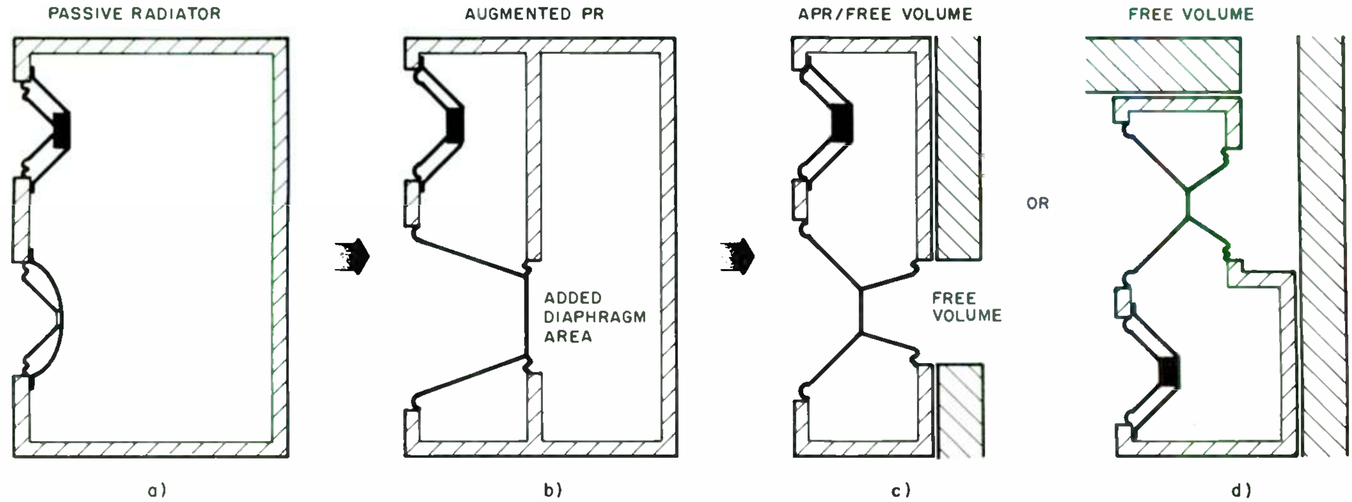


FIGURE 2: The progression from conventional passive-radiator system to an APR/FV system. My living room fireplace opening serves as the perfect free volume mounting place.

stored energy in the compressed air. A large portion of the enclosure's volume can be freely hidden. In a vented box, this technique has the advantage of heavier acoustical loading to boost power handling over an infinite baffle. The free volume should, of course, be rigid enough to avoid problems with rattles and noise.

CREATIVE ENCLOSURES. *Figure 2(c)* is my APR free volume design, as it mounts into my living room fireplace opening. If you use this design, or any other vented to the outside, you can avoid pipe organ or Helmholtz resonator effects by closing the flue or vent. The APR installation is limited only by the installer's imagination, and can be vented to attic or basement spaces by adapting your idea to the designs in *Fig. 2*. Systems can even be mounted in windows to achieve a true "infinite baffle" by using the entire world as a baffle.

These enclosures are constructed with conventional techniques. The passive radiator cones can be mounted apex-to-apex as *Fig. 2* shows, or in deeper enclosures you can use a connecting tube to join the cones. The individual APR cones can be glued directly to the enclosure to eliminate the need for metal baskets on the passive radiators. For the smaller cone in a passive-radiator, a rolled-edge gives the best combination of linear compliance and suspension linearity.

If properly sized cones are unavailable, you can easily build rolled-edge cones by spreading a flexible material (such as silicone rubber) over a half-round mold, coated with plastic food-wrap.[1] Mine, as shown in the figures, is a 22-inch cone with a 15-inch accordion edge cone. For cloth edge cones, apply sealant to avoid a leak through the cloth.

The APR/free volume designs are always custom. You could build an enclosure with about half the interior volume of a vented box for the particular driver. The two cones mount apex-to-apex; the smaller is the same size as the driver and the larger is the next size greater (for example 12 and 15-inch cones for a 12-inch driver). Weight is then added or subtracted from the cones to achieve the desired response.

CHOOSING DRIVERS. A better alternative for an optimal system would use the driver's Thiele/Small parameters together with the design examples in *Table 1*. If you use half the volume of a vented box it can lead to a poor design for some drivers. The three Thiele/Small parameters are driver resonance frequency, f_S, driver damping factor Q_{TS} and the acoustic volume equivalent to the driver suspension compliance, V_{AS}.

These three quantities completely characterize a driver, so that two drivers with the same parameters and enclosures will perform identically. The drivers may have entirely different sizes, power handling capability or other specifications, but fortunately many manufacturers and suppliers provide the parameters.

Table 1 gives the designs for four drivers of varying Q_{TS}. The driver parameters are followed by the box volume, V_B (in cubic feet), the cutoff frequency f_3 (in hertz) and in grams for properly tuning the passive radiator mass. Driver sensitivities, in dB/1W input at one meter, are also included.

Below the APR designs are the corresponding parameters for a vented box enclosure matched to the same driver. The designs use Chebyshev or equi-ripple alignments.[2] These designs assume that the combination of a small cone equal to the driver size is used together with a larger cone of the next larger size. If non-standard cones are used, the larger should be 30 percent greater than the smaller one.

The first design uses a 10-inch commercial driver with low efficiency. For the APR/FV system to achieve a 23.6Hz cutoff frequency you will need a 1.5 cubic foot enclosure, whereas a vented box requires nearly five cubic feet. A system with two of these drivers in the same enclosure would be very similar to the one shown in the figures.

The second APR/FV system uses a larger 12-inch commercial driver to reach a 22.3Hz cutoff with an internal volume of 8.66 cubic feet. A comparable vented box only goes to 24Hz and requires 18 cubic feet, and the sensitivity rises to 95.2 with the larger volume.

The third system uses a 15-inch professional driver to reach the very low 19Hz cutoff. The sensitivity is a moderate 92.8, while the volume is only 9.75 cubic feet. A vented box will require 16 cubic feet to reach only 23.5Hz. This system is best with the lowest octave reproduced with absolute fidelity.

The last system uses a large 18-inch professional driver to achieve high ef-

TABLE 1

APR / FV — 1.3 DIAMETER RATIO

Model	Q_{TS}	V_{AS} (ft³)	f_S (Hz)	V_b (ft³)	f_3 (Hz)	Rip. (dB)	f_B (Hz)	PR Mass (gm)	Sens. dB 1W/1M
Pyle W10C300F	**0.49**	**2.5**	**31**	**1.5** [0.59]	**23.6** [0.76]	**1.4**	**27.9** [0.90]	**95** [2.5]	**87.9**
10" dia.				*4.67*	*22.2*	*0.1*	*25.1*		
Pyle W12C200F	**0.40**	**20.5**	**25**	**8.66** [0.42]	**22.3** [0.89]	**0.9**	**26.3** [1.05]	**51** [2.6]	**95.2**
12" dia.				*18*	*24.8*	*0*	*24.9*		
Altec 411-8A	**0.35**	**29.3**	**19**	**9.75** [0.33]	**19.0** [1.0]	**0.7**	**22.3** [1.18]	**178** [2.7]	**92.8**
15" dia.				*16*	*23.5*	*0*	*21.3*		
JBL 2245H	**0.27**	**29.0**	**20**	**5.9** [0.20]	**25.6** [1.28]	**0.4**	**29.8** [1.48]	**348** [2.7]	**95.1**
18" dia.				*7.54*	*35.2*	*0*	*28.5*		

APR/FV = Bold Face; *Vented Box = Italic.*

NOTE: In the table above figures for APR/FV designs are in BOLD type; the vented box design values are in *Italic type*. The values in brackets may be used to adapt the system to other drivers.

ficiency and power handling. The volume is 5.9 cubic feet and the cutoff is a moderate 25.6Hz. The comparison vented box is slightly larger at 7.5 cubic feet, but has a much higher cutoff of 35.2Hz. This system should be used to achieve high SPLs. It clearly shows the advantages of an APR/FV enclosure over the Altec Model 8182 system.

These systems can be adapted to other drivers by using the numbers in brackets, below the system specifications, in *Table 1*. Choose the example with Q_{TS} nearest the target driver. The system's volume is found by multiplying equivalent driver volume, V_{AS}, by the number below V_b for that example. Similarly, you find f_3 and f_B by multiplying the target driver's resonance frequency by the corresponding bracketed numbers.

The passive radiator mass is found by multiplying the driver's moving mass by the bracketed number below the mass. This mass value assumes that the small cone is the same size as the driver; if not, multiply the mass by the ratio of the cone diameters to the fourth power. You can make the best use of an APR/FV system when you must reproduce low frequencies at high levels.

More complete details of how to use this concept together with basic computer programs and licensing information are available from the author. Write, Thomas L. Clarke, 5814 SW. 81 St., Miami, FL 33143.

REFERENCES

1. Clarke, T.L., "Build a Superthruster Loudspeaker," *CQ* (Volume 20), February 1979, pp. 101–104.
2. Clarke, T.L., "Augmented Passive-Radiator Loudspeaker Systems," Parts I and II, *JAES* (Volume 29), June and July/August 1981, pp. 511–516.

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.

NOTCH FILTERS

BY DAVID B. WEEMS

Have you ever found that notch filters did not work the way theory predicted? Well, that's not surprising. Notch filter formulas ignore some practical aspects of filter behavior, and filter discussions ignore areas of background information. As you probably know, a notch filter is a frequency discriminating circuit consisting of an inductor, a capacitor and usually a resistor. The inductive reactance equals the capacitive reactance at predictable frequencies to form a tank circuit.

If this tank circuit is designed to resonate at a speaker's response peak, it will offer a high impedance to current flow at that frequency and reduce loudspeaker output. The shunt resistor controls a filter's impedance and Q. Filters can be used as sharply tuned notch filters or as broadband response shapers.

I recently discovered how useful notch filters can be. After I helped a friend install a small double-chamber reflex, two-way system (6½-inch woofer and 2-inch tweeter), we noticed some flaws. The stereo image was unstable, with more than just a suggestion of shrillness on some program material. We began experimenting with various crossover networks and then with filters. The filters, as it turned out, made the biggest difference. But, first we had to find the problem.

PEAK TESTING. The methods described here can be used for crossover experiments and detecting peaks. To test your drivers, install them in their enclosures and run leads out for each one. You can then alter the crossover wiring or insert filters, while you monitor performance changes without disturbing the drivers.

It is nice if you can test with equalizers and analyzers, but we tracked down our problems with just an audio generator, an amplifier and a sound level meter.

We used a Heathkit IG-18 (now renumbered IG-5818) audio generator with a Radio Shack 40-3019 sound level meter (SLM: no longer available). We connected the audio generator to an old power amplifier, which fed the speaker through the crossover network. Note that the final crossover network should be in the system, and you must use an amplifier to couple the audio generator to the loudspeaker.

I cannot overstate the importance of maintaining a constant acoustical environment while testing your speakers with an SLM. Outdoor testing is the preferred way to test, but often it is not practical due to the weather, traffic noise or other influences. You can learn a lot from indoor experiments by maintaining absolute control over room acoustics. For example, if you are working alone, you will need to make changes in the filter wiring while observing the SLM. You may find that body movements of no more than an inch will change your SLM reading. One slight shift in your position, or an object in the soundfield, will give inconsistent test data. You can keep the soundfield clear by having an assistant make the wiring changes while you read the SLM.

Our rough frequency response tests indicated some uncontrolled peaks. We disconnected the driver to study the woofer and tweeter performance separately. We replaced the disconnected driver with dummy resistors to maintain the crossover network's proper termination, and we found a significant peak beyond the passband on each tested woofer. One woofer's response peak occurred at 3.2kHz, the other's at 3.7kHz. When we checked the tweeters, one peaked at a little over 5kHz and the other at just under 6kHz. Once we brought the problem into focus we saw it was the filter design.

CHOOSING VALUES. When you compare inductance and capacitance values in various notch filters, you are likely to be puzzled by the design choices. After all, the range is almost infinite as long as the capacitive reactance equals inductive reactance at peak frequency. You may understand filter action better by going back to basic engineering textbooks, but you may find you have as many questions as you had before.

Some books only deal with theoretical conditions, where each component is perfect. Those that treat practical problems usually discuss the kind plaguing high frequency circuits, rather than audio bands.

The SRA *Loudspeaker Design Cookbook*[1] gives a formula to calculate the values of filter components. It is:

$$C = \frac{1}{15.2f}$$

where (f) is the frequency of the peak. After finding the value for C, you can substitute it into the following formula to find L:

$$L = \frac{0.025}{f^2C}$$

The value of C is farads, and L is henries.

The LDC formula produces a filter with an LC ratio near 5.8. Filters with low LC ratios, such as this, will theoretically have a narrower bandwidth than a filter with higher LC ratios. As we shall see, bandwidth does not always follow theory. You may need a filter that can give greater attenuation for severe peaks. A rule of thumb for a higher impedance filter is:

$$C = \frac{1}{100f}$$

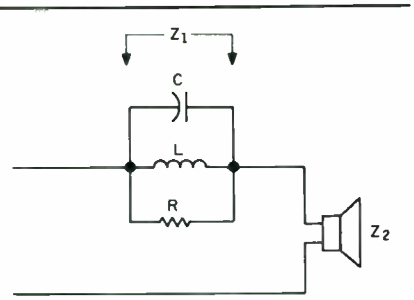
The LC ratio here is 250.

Finally, this intermediate formula produces an LC ratio of 40:

$$C = \frac{1}{40f}$$

The formula you choose will depend on the peak's amplitude and bandwidth. By altering the LC ratio you can change the theoretical Q of your filter. The real Q may be quite different.

FILTER Q. You will often see or hear the statement that the Q of a parallel resonant circuit is determined by the Q of the coil. The capacitor, it is said, has little or no effect on Q. These assertions are probably more valid for high frequency circuits than for the audio band. I checked their relevance to notch filters by running tests with single coil filters and matching them to capacitors with equal value. I chose a commercially available 1mH air-core coil inductor, wound with 18-gauge magnet wire on a plastic form, for each test (*Fig. 1*). The filter's impedance peaks at about 2.5kHz when the coil is wired into a parallel circuit with a 4μF capacitor.



Z_1 = Impedance of filter
Z_2 = Impedance of speaker

Voltage ratio (V_R) = $Z_2 + Z_2$

Cut (dB) = 20log V_R
or
Cut (dB) = 20log $\frac{Z_2}{Z_1} + Z_2$

FIGURE 1: The parallel resonant circuit used for notch filters.

I used a non-polarized (NP) electrolytic capacitor on the first test. The Q for that combination was about seven. Next, I substituted a couple of 2μF mylar capacitors wired in parallel to maintain the same total capacitance and the Q jumped to 14.5. Finally, I tried an old Western Electric capacitor gleaned from a piece of surplus telephone equipment in a salvage yard. Its value was apparently greater because it produced a lower resonance frequency and a Q of 14.7.

These tests suggest an irony. If you want a high Q circuit, theory insists on a low LC, requiring a large value for C. Good quality capacitors, with large values, can be expensive. If I use a 2.5kHz filter with an LC ratio of six, I will need a 26μF capacitor. Fortunately, we can usually get practical values of Q with NP electrolytic capacitors.

According to textbooks, you can predict the Q from the LC ratio. In this formula R is the shunt resistance:

$$Q = R\sqrt{C/L}$$

This formula has little practical use. For a better estimate of filter Q, make up the filter and measure its impedance at its resonant frequency. You can test the impedance of a filter in the same way you measure a driver's impedance, by the constant current method.[2] When you find a maximum impedance, you can calculate Q with this formula where Z is the impedance of the filter at resonance and f is the frequency of resonance:

$$Q = Z \times 2\pi fC$$

You can measure Q directly by recording the impedance, or maximum voltage across the filter at resonance. Then find the two frequencies, one below resonance f_1, and the other above resonance f_2, where the impedance (or voltage) falls to 0.707 that of the value at resonance. Then:

$$Q = \frac{f}{\Delta f}$$

where f is the resonance frequency and Δf is $f_1 - f_2$. You can check the accuracy of your work by this test:

$$f = \sqrt{f_1 f_2}$$

As you can see, measuring filter Q is not difficult. If we want to make our filter a mirror image of a speaker's response curve, it would be useful to be able to predict the Q with reasonable accuracy. This involves decisions about the value of R as well as the LC ratio.

R VALUES. It appears to be easy to calculate R for any desired Q according to textbook formulas. The calculations are easy, but getting a filter with the right Q requires bending the formulas a bit. Here is one formula:

$$R = \frac{X_L f}{\Delta f}$$

or

$$R = X_L Q$$

I made up a filter with a low LC, designed to resonate at 3kHz to test this formula. The formula for an LC ratio of six showed that a 21.9μF capacitor would be right. I planned to use a 22μF NP electrolytic capacitor, but when I measured it the true value turned out to be 21.4μF. I would need an air-core coil with an inductance of 0.13mH to match. I then made a series of tests, which are reported in *Table 1*.

TABLE 1

THEORETICAL Q's vs. MEASURED Q's

R	**Theoretical Q**	**Measured Q: NP Elect. Caps.**	**Measured Q: Mylar Caps.**
1.0	0.42	0.48	0.47
1.5	0.63	0.63	0.63
2.5	1.05	0.95	0.95
5	2.1	1.7	1.8
10	4.2	2.6	3.0
15	6.2	2.9	3.9
22	9.1	3.5	4.8
30	12.5	3.7	5.5
100	41.5	4.3	8.1
510	212	5.2	10.3
∞	∞	5.4	12

LC ratio = 6. Components: C = 21.4μF; L = 0.13mH; f = 3kHz.

Check this hypothetical problem against *Table 1*. Suppose we have a driver with a peak at 3kHz, and a bandwidth peak of 800Hz. The bandwidth is defined as the −3dB bandwidth peak. The Q should be 3.75 (3k/800). For the low LC ratio filters X_L as well as X_C is about 2.4Ω. So, by formula:

$$R = 2.4\Omega \times 3.75 = 9\Omega$$

Table 1 shows the closest value is 10Ω. It appears that the measured Q with a 9Ω resistor would be somewhat below 2.6. We see farther down the column that a real filter with an electrolytic capacitor and a Q of 3.75, we

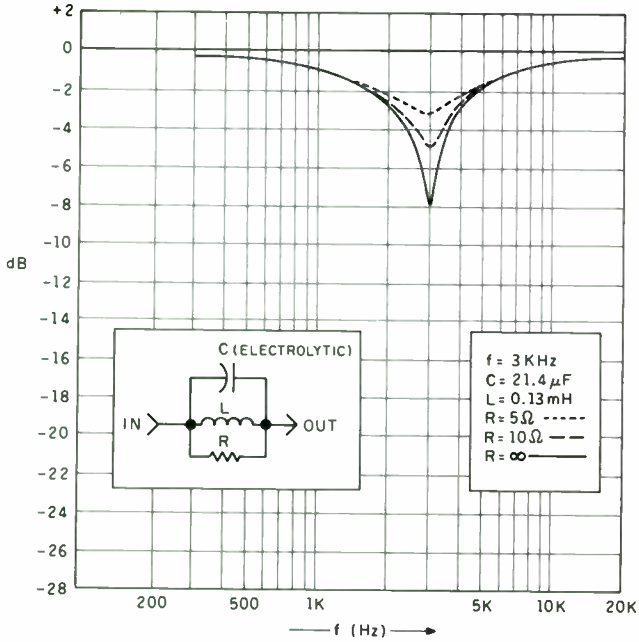


FIGURE 2: A notch filter response with an LC ratio of six, made with a non-polarized electrolytic capacitor.

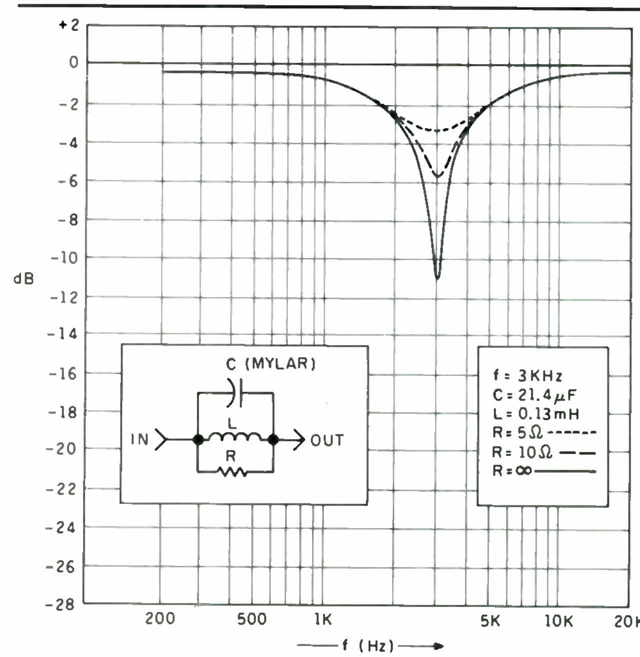


FIGURE 3: The response of the same filter in Fig. 2, but with a mylar capacitor.

would expect to use a parallel resistance of slightly more than 30Ω: more than three times the calculated value. This is hardly reassuring for anyone who relies on formulas alone. For broadband filters the formulas work fairly well. *Tables 2, 3* and *4* show how Q varies from the theoretical value with various LC ratios. *Figures 2, 3, 4* and *5* show the response curves of filters with various LC ratios and values of R.

PRACTICAL FILTERS. The tests reported in the tables and figures show that for filters with Qs greater than one, the bandwidth is wider than predicted by theory. Filters with either very high or very low LC ratios have limited maximum Qs. The LC ratio probably limits the Q for filters with very high LC ratios, such as the one with a ratio of 8600.

The limiting factor for low LC filters is likely to be large values of capacitance. The available Qs, even with electrolytic capacitors, may be higher than you need for most purposes. The ear's ability to detect deviations from a flat frequency response depends on the aberration's bandwidth and level. The narrower the bandwidth, the more prominent it must be to be heard.

For a practical approach to filter design, refer to *Table 5*. If you know the approximate bandwidth and level to be eliminated, you can easily choose an appropriate filter from the table. Here is an example of how you can tailor a filter to meet your needs.

Suppose you locate a 9dB peak at 3kHz that has a 1kHz bandwidth, or a Q of 3. *Table 5* shows us that with an LC ratio of 6 we can expect a cut of 6dB with a filter Q of 3. The cut would be about 10dB; much closer to the needed attenuation.

If you are finicky, you may want to tinker with filter components to get the precise degree of cut and bandwidth. The data presented in *Table 5* is based on tests with filters using air-core coils and NP electrolytic capacitors.

To estimate R, find the combination of LC ratio and Q that best fits your needs and then find the value of R in *Tables 1, 2* or *3*. If you make a filter with an LC ratio that is different from those listed, you can make a rough interpolation for R. Although *Table 5* is based on a single series of tests, such estimates will probably be more accurate than finding the values of R from a formula.

Once you settle on the LC combination, you can test the various resistors by finding the desired impedance at

TABLE 2

THEORETICAL Q's vs. MEASURED Q's

R	Theoretical Q	Measured Q
1.5	0.24	0.28
2.5	0.40	0.45
5	0.79	0.8
10	1.6	1.3
15	2.4	1.7
22	3.5	2.0
30	4.7	2.7
100	15.8	4.5
510	81	5.4
∞	∞	5.8

LC ratio = 40; Components: C = 8μF N.P. electrolytic; L = 0.32mH; f = 3125Hz.

TABLE 3

THEORETICAL Q's vs. MEASURED Q's

R	Theoretical Q	Measured Q NP Elect. Caps.	Measured Q Mylar Caps.
5	0.32	0.37	0.32
10	0.63	0.67	0.69
15	0.95	0.88	0.88
22	1.39	1.18	1.33
30	1.90	1.44	1.68
100	6.3	3.25	4.6
510	32.3	6	10.3
∞	∞	7	14.5

LC ratio = 250; Components: C = 4μF; L = 1mH; f = 2.5kHz.

TABLE 4

THEORETICAL Q's vs. MEASURED Q's

R	Theoretical Q	Measured Q
30	0.32	0.31
50	0.54	0.47
100	1.08	0.82
510	5.50	1.74
1000	10.78	2.07
∞	∞	2.73

LC ratio = 8600; Components: C = 0.5μF; L = 4.3mH iron core; f = 3.4kHz.

the resonance frequency. That will be:

$$Z = \frac{Q}{2\pi fC}$$

For filters that work outside the driver's passband, you can usually pick just about any LC combination and omit R. Just make sure the filter's action does not overlap the passband.

With all the filter design variables there is a good chance that you can throw caution to the wind and do an acceptable job using the components you have on hand. It's at least worth a try. For example, suppose you need a filter tuned to 1.5kHz with a bandwidth of 600Hz, or a Q of 2.5, and you have a coil with an inductance of 0.5mH that you would like to use. The matching capacitor would be:

$$C = \frac{0.025}{f^2L}$$

$$= \frac{0.025}{1500^2\ 0.0005}$$

$$= 0.0000222 \text{ or } 22.2\mu F$$

This filter's LC ratio is a relatively low 22.5.

If you check *Table 5* this filter may be just about right if the peak you want to kill is between about 5 and 8dB. If the peak is lower in amplitude, say 3dB, then depending on what Q you choose, your filter will either produce a dip at resonance or two shallow depressions. If the filter appears to be close enough, you would solve for the desired impedance:

$$Z = \frac{2.5}{2 \times 3.14 \times 1500 \times 0.0000222} = 11.9\Omega$$

With a target impedance of 11.9Ω you would wire in various resistors and check the impedance at 1.5kΩ, until you find the resistance that produces about a 11.9Ω filter impedance.

If you do not have the materials for coil making, or do not want to make them, you can buy ready-made coils. If you buy them ready-made, choose the value of the coil first and then find the matching value of capacitance. For example, one filter I tested for this article required a 0.13mH coil. I had trouble finding a commercial coil with 0.13mH inductance, but I found a 0.14mH coil. It worked almost as well as the "ideal" with a 20μF capacitor.

FILTER TESTING. When you have made your filter, wire it into the speaker circuit and temporarily short it with a test lead. With your audio generator, set the peak frequency and adjust the level so the output reads at a conveniently high value on your tripod-mounted SLM. I used +6dB on the SLM, the maximum mark on the scale. To check the degree of suppression, unclip the test lead and observe the change in the SLM reading. This test double checks your impedance measurements.

You should observe a couple of cautions when using notch filters. First, if you use one with a crossover network, make sure the filter does not reflect a high impedance to the crossover. You can avoid this by designing an impedance equalizer for the driver[3,4,5] and wiring it into the system between the crossover and the notch filter as shown in *Fig. 6*.

TABLE 5

Q-Adj. by R	Filter Response (dB) LC = 6	LC = 40	LC = 250
0.3	−0.77	−2	−4
0.5	−1.25	−3	−6
1	−2.4	−5	−10
2	−4	−8	−14
3	−6	−10	−17
4	−7	−12	−21
5	−8	−14	−22
7	—	—	−24

Typical response cut at resonance for notch filters with various LC ratios. These results can be expected for filters made with air core coils and non-polarized electrolytic capacitors.

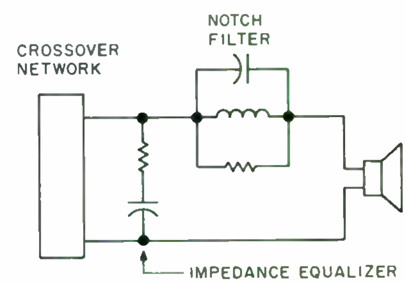


FIGURE 6: This speaker circuit combines an impedance equalizer with a notch filter.

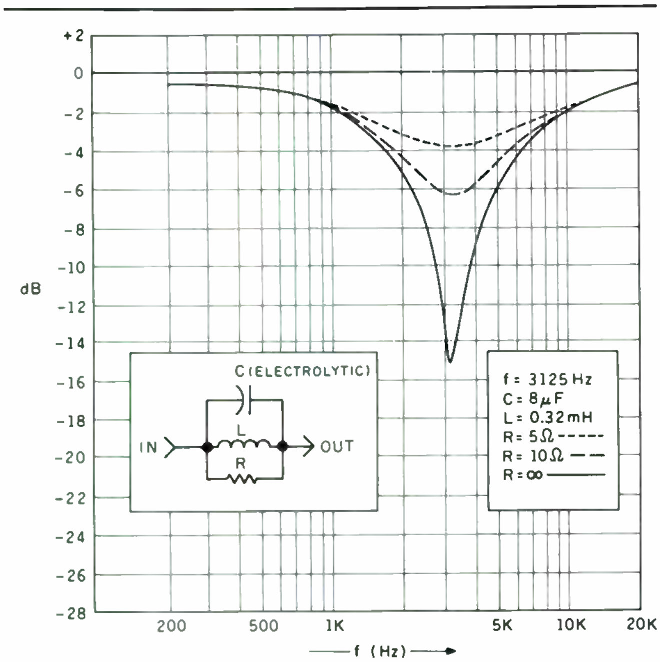


FIGURE 4: A notch filter response with an LC ratio of 40.

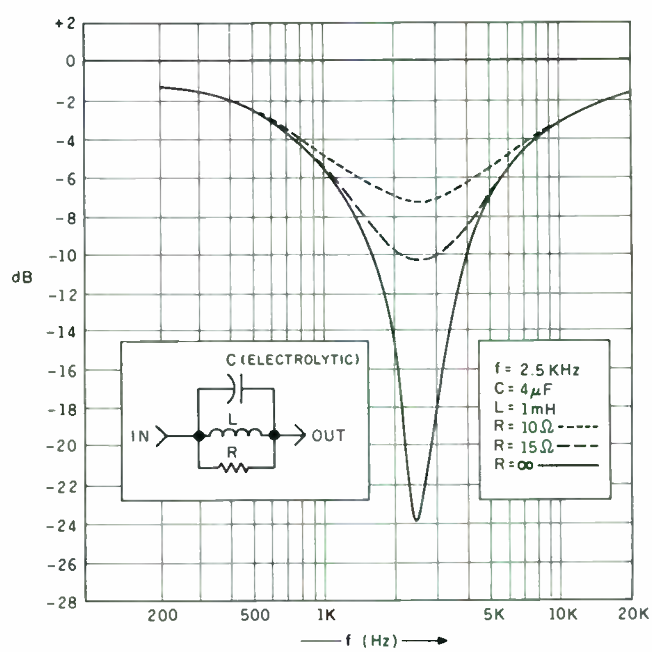


FIGURE 5: A notch filter response with an LC ratio of 250.

If you're making subjective evaluations of your notch filter, say A/B listening tests, keep the loudness the same with the filter in and out of the circuit. Broadband filters usually produce significant changes in sound level, and most listeners will vote against the hook-up with the reduced output.

OUR LISTENING TESTS. We made four filters for the two double-chamber reflex speakers, following the described procedures. We wired them in series with the drivers they were designed to correct. The improvement was striking, and much greater than we had expected. The surface noise on old records lost its bite, and highs were much more natural. The most obvious change was in the stereo image. Definition was also significantly improved. Above all, the loudspeakers seemed to invite extended listening without fatigue.

Everyone who heard the system, before and after we added the filters, concluded that removing the peak improved the quality more significantly than the tinkering we had done with the crossover and enclosure.

REFERENCES

1. Speaker Research Associates, *Loudspeaker Design Cookbook*, p. 31. *[Now out of print. A new edition is in preparation and will be announced soon. Use Fast Reply No. GK84 for notification when the new edition is ready.—Ed]*
2. Bullock, R. M., "How You Can Determine Design Parameters for Your Loudspeakers," *SB* 1/81, p. 12.
3. Harms, W., "Evaluating The Zobel," *SB* 4/82, p. 14.
4. Knittel, M. R., "Impedance Compensating Crossover," *SB* 1/83, p. 11.
5. Bullock, R. M., "Passive Crossover Networks," *SB* 1/85, p. 13.

KITS

from **OLD COLONY**

Post Office Box 243 • Peterborough, NH 03458-0243

ORDERING INFORMATION Prices, except as noted, are prepaid in the USA and insured. We prefer to ship via UPS, which requires a street address. If you cannot receive UPS delivery, please include an extra $2 for insured service via Parcel Post. We cannot accept responsibility for safety or delivery of uninsured Parcel Post shipments. Orders under $10 please add $2 service charge.

PRICES SUBJECT TO CHANGE WITHOUT NOTICE.
CHARGE CARD PHONE ORDER SERVICE M-F 9am—4pm EDT (603) 924-6371
MasterCard/Visa Cards Welcome. UPS next day and 2nd day air are available to some areas.

CROSSOVERS

KC-4A: ELECTRONIC CROSSOVER, KIT A. [2:72] Single channel, 2-way. All parts including C-4 board and LF351 IC's. Choose frequency of 60, 120, 240, 480, 1k, 2k, 5k or 10k. Each **$9.75**

KC-4B: ELECTRONIC CROSSOVER, KIT B. [2:72] Single channel, 3-way. All parts including C-4 board & LF351 IC's. Choose frequency of 60, 120, 240, 480, 1k, 2k, 5k or 10k. Each **$12.75**

● **KF-7: CROSSOVER FOR WEBB TLS.** [1:75] Passive 4-way x-over, in pairs, assembled. Components are included for both STC and Celestion tweeters. Made by Falcon of England. CLOSEOUT Pair **$50.00**

KK-6L: WALDRON TUBE CROSSOVER LOW PASS: Single channel, 18dB/octave, Butterworth, [3:79] includes 3-gang pot. Choose 1: 19-210; 43-465; 88-960; 190-2100; 430-4650; 880-9600; 1900-21,000 Hz. Each **$43.00**

KK-6H: WALDRON TUBE CROSSOVER HIGH PASS: Single channel, 18dB/octave, Butterworth, [3:79] includes 3-gang pot. Please specify 1 of the frequencies in KK-6L. No other can be supplied. Each **$45.00**

KK-6S: SWITCH OPTION. 6-pole, 5-pos. rotary switch, shorting, for up to 5 frequency choices per single channel. Each **$8.00**
ordered with 2 kits above, Each **$7.00**

KK-7: WALDRON TUBE CROSSOVER POWER SUPPLY. [3:79] Includes board, x-fmr, fuse, semiconductors, line cord, capacitors to power 4 tube x-over boards (8 tubes), 1 stereo biamped circuit. Each **$88.00**

SBK-A1: LINKWITZ CROSSOVER/FILTER. [SB 4:80] 3-way x-over/filter/delay. 24dB/octave at 100Hz and 1.5kHz and 12dB/octave below 30Hz, with delayed woofer turn-on. Use the Sulzer supply KL-4A with KL-4B or KL-4C.
Per channel **$64.00** Two channels **$120.00** SBK Board only **$14.00**

SBK-C1A: JUNG ELECTRONIC 2-WAY CROSSOVER. [SB 3:82] 30Hz filter with WJ-3 board & 4136 IC adapted as 1 channel x-over. Can be 6, 12 or 18dB/octave. Choose frequency of 60, 120, 250, 500, 1k, 2k, 5k or 10k. Each **$24.75**

SBK-C1B: THREE WAY, SINGLE CHANNEL CROSSOVER. [SB 3:82] Contains 2 each SBK-C1A. Choose high & low frequency. Each **$49.70**

SBK-C1C: TWO CHANNEL, COMMON BASS CROSSOVER. [SB 3:82] Contains 2 each SBK-C1A. Choose 1 frequency. Each **$49.70**

SBK-C2: BALLARD ACTIVE CROSSOVER. [SB 3,4:82] 3-way x-over with variable phase correction for precise alignment. Kit includes PC board ($5^3/_8 \times 9^1/_2$"), precision resistors, polystyrene & polypropylene caps. Requires ± 15V DC power supply—not included. Can use KL-4A with KL-4B or C. Two channel **$134.00**

FILTERS & SPEAKER SAVER

KF-6: 30Hz RUMBLE FILTER. [4:75] 2 channel universal filter card, 1% metal film resistors and 5% capacitors for operation as 18dB/octave; 30Hz, 0dB gain only. Each **$19.75**

KH-2A: SPEAKER SAVER. [3:77] Turn on/off protection & fast opto-coupler circuitry to prevent damage to your system. 4PDT relay & socket for 2 channels. Each **$40.00**

KH-2B: OUTPUT FAULT OPTION. Additional board mounted components for speaker protection in case of amplifier failure. Each **$6.75**

KH-2C: COMPLETE SPEAKER SAVER KIT. Includes KH-2A & KH-2B. Each **$40.00**

KL-5 WILLIAMSON BANDPASS FILTER. [2:80] 2 channel, plug-in board and all parts for 24dB/octave 20Hz-15kHz with precision cap/resistor pairs. TL075 IC's. Each **$31.00**

AIDS & TEST EQUIPMENT

KH-7: GLOECKLER PRECISION 101dB ATTENUATOR. [4:77] All switches, 1% metal film and 5% carbon film resistors to build prototype. Chassis, input/output jacks are not included. Each **$55.00**

KL-3C: INVERSE RIAA NETWORK COMPLETE. [1:80] 1 KL-3R and 1 KL-3H with 1% polystyrene capacitors. Alternate 600 ohm or 900 ohm R_2'/C_2' components for 2 channels. Each **$35.00**

KL-3R: INVERSE RIAA. [1:80] Resistor/capacitor package complete. Contains stereo R_2'/C_2' alternates. Each **25.00**

KL-3H: INVERSE RIAA HARDWARE. [1:80] Box, terminals, gold jacks, and all hardware in KL-3C. No resistors or caps. Each **$13.50**

KF-4: SINE-SQUARE AUDIO GENERATOR. [4:75] Morrey's MOD kit for Heath IG-18 (IG5218). 2 boards and parts to modify the unit to distortion levels of parts per million range. Each **$37.00**

KJ-6: CAPACITOR CHECKER. [4:78] All switches, IC's, resistors, 4½" D'Arsonval meter, x-fmr and PC board to measure capacitance, leakage and insulation. Each **$78.00**

KK-3: THE WARBLER OSCILLATOR. [1:79] Switches, IC's, x-fmr and PC board for checking room response and speaker performance w/o anechoic chamber. Each **$56.00**

KL-6: MASTEL TIMERLESS TONE BURST GENERATOR. [2:80] All parts with circuit board. No power supply. Each **$19.00**

KM-1: CARLSTROM-MULLER SORCERER'S APPRENTICE [2:81] 4 boards and all parts for construction of the first half of a swept function generator with power supply. No knobs or chassis. Each **$145.00**

KM-2: CARLSTROM-MULLER PAUL BUNYAN. [3:81] All parts except knobs, chassis, output connectors and wire. Includes 2 circuit boards and power supply. Each **$85.00**

KM-3: CARLSTROM-MULLER SORCERER'S APPRENTICE/PAUL BUNYAN [2:81, 3:81] All parts in KM-1 and KM-2. Each **$225.00**

SBK-D2 WITTENBREDER AUDIO PULSE GENERATOR. [SB 2:83] All parts, board, pots, power cord, switches and power supply included. Each **$70.00**

SBK-E4: MULLER PINK NOISE GENERATOR. [SB 4:84] All parts, board, 1% MF resistors, capacitors, IC's, and toggle switches included. No battery or enclosure. Each **$27.50**

SYSTEM ACCESSORIES

KH-8: MORREY SUPER BUFFER. [4:77] All parts, 1% metal film resistors, NE531 IC's, and PC board for 2 channel output buffer. Each **$16.00**

KJ-3: TV SOUND TAKEOFF. [2:78]. Circuit board, vol. control, coils, IC, co-ax cable (1 ft.) and all parts including power x-fmr. Each **$21.50**

SBK-D1: NEWCOMB PEAK POWER INDICATOR. [SB 1:83] All parts & board. No power supply required. Two for **$10.00** Each **$6.00**

SBK-E2: NEWCOMB NEW PEAK POWER INDICATOR. [SB 2:84] All parts & board, new multicolor bar graph display; red, green & yellow LED's for 1 channel. No power supply needed. Two for **$17.00** Each **$11.50**

KC-5: GLOECKLER 23 POSITION LEVEL CONTROL. [2:72] All metal film resistors, shorting rotary switch & 2 boards for a 2 channel, 2dB per step attenuator. Choose 10k or 250k ohms. Each **$36.75**

KR-1: GLOECKLER STEPUP MOVING COIL TRANSFORMER. [2:83] X-fmrs., Bud Box, gold connectors, & interconnect cable for stereo. Each **$335.00**

KL-2: WHITE DYNAMIC RANGE & CLIPPING INDICATOR. [1:80] 1 channel, including board, with 12 indicators for preamp or x-over output indicators. Requires ± 15V power supply @ 63 mils. Single channel. Each **$49.00**
Two channels. **$95.00** Four channels. **$180.00**

● **KS-7: SCOTCHCAL® PANEL KIT.** [2:84] One 10 × 12" sheet each of 4 types of pressure sensitive panel material (blk on aluminum, blk on transparent poly, blk on white poly, matte clear overlay), one pint of developer plus pads, and instructions. Requires a simple frame and a light source: ultraviolet, photofloods or the sun, plus your own press-on lettering materials. Postpaid. Each **$34.50**

● **CLOSEOUT: KITS NOT AVAILABLE AFTER PRESENT STOCK IS GONE.**

What's Included? Kits include all the parts needed to make a functioning circuit, such as circuit boards, semiconductors, resistors and capacitors. Power supplies are not included in most cases. Unlike kits by Heath, Dyna and others, the enclosure, face plate, knobs, hookup wire, line cord, patch cords and similar parts are not included. Step by step instructions usually are not included, but the articles in ***Audio Amateur*** and ***Speaker Builder*** are helpful guides. Article reprints are included with the kits. Our aim is to get you started with the basic parts—some of which are often difficult to find—and let you have the satisfaction and pride of finishing your unit in your own way.

BI–WIRING THE LS3/5A

BY LEN HUPP

The following modifications are an outgrowth of my experimentation with the technique of bi-wiring loudspeaker systems. Mr. Fulton of FMI (Fulton Music Industries) has stated that bi-wiring provides about 70 percent of the sonic improvements of bi-amplifying, while using only one stereo power amplifier.

BI-WIRING. Bi-wiring, as proposed by Mr. Fulton, is a simple procedure. It requires the inputs to the woofer and tweeter sections of the crossover be separated so each driver (and its associated crossover section) will have its own set of input connectors, accessible from inside or outside the enclosure. Separate two-conductor cables are then run from the power amplifier outputs to each of the drivers. This means each driver gets its own speaker cable from its own respective amplifier channel. Basically, this is all there is to bi-wiring, and I believe it has made sonic improvements to every system on which I have tried it.

The LS3/5A Mini-Monitor is one of the speaker systems that has greatly benefited from the bi-wiring technique. I removed the crossovers from the enclosures and mounted them in a stacked array in their own separate, shielded enclosure. I mounted separate terminals for the woofer and tweeter on the backs of the enclosures, with 18-gauge wire running from the drivers to their terminal connectors. The power amplifier was connected to the pair of crossover networks with short and heavy-gauge cables. I also ran heavy-gauge cables from each crossover output to the proper drivers.

IMPROVEMENTS. The system now sounds firmer and less boomy on the bottom end, cleaner and more transparent in the midrange and top end, has better depth and imaging and significantly less veiling from top to bottom. This dramatic improvement was accomplished without changing anything else in the chain of audio components.

Those of you who may be thinking that merely doubling up your speaker cables, instead of bi-wiring, will get the same results may like to know that it will not. If you want the improvements of the bi-wiring method, you must bi-wire the speaker systems. Bi-amping or tri-amping will give you even better results, but at a higher price.

The reason I tried bi-wiring originally was to improve the sound of my LS3/5A system. In its stock form the LS3/5A has its virtues, but with limitations. These limitations are due, as its designers know, strictly to cost constraints and application factors.

FLAWS & LIMITATIONS. The LS3/5A's BBC (British Broadcasting Corporation) designers intended it to be a small, consistent studio voice monitor. The humble LS3/5A successfully reached this goal. Even with its flaws and limitations, this Mini-Monitor manages to reproduce the important middle octaves of the audible frequency spectrum more accurately than speaker systems selling for many time its price.

Its flaws and limitations are not lessened just because it sounds better than many other speaker systems. Since the BBC determines its specifications, the three British licensed manufacturers cannot change or improve the units or they will certainly lose their licenses. However if you own a pair of LS3/5As, and are not afraid of voiding the warranty, you can dramatically improve their overall sound accuracy. It only requires a bit of time, patience and care.

The first modification is to bi-wire as previously described. The next step is to permanently remove the grille covers, or replace them with a thin reticulated foam material. The stock grilles cause as much as a 2dB loss in the mid- and high-end. The next step is to pry off the black, perforated metal tweeter grilles without cutting the T-27's tweeter leads or the domes. The perforated grilles increase the treble by increasing the air pressure between the dome and grille, and give the tweeter its slightly raspy sound. The treble is flat and smooth with the grilles removed, and the glitches and peaks are eliminated.

ENCLOSURES. The next and last step is to seal the inside of the enclosures with a layer of mortite, rope caulk or silicone, and stuff them with a 6½ by 11 by 6-inch piece of high-grade fiberglass insulation. This lowers the LS3/5A's resonant Q and eliminates part of the upper-bass hump. It will also extend the lower-bass limit a bit and improve tautness, impact and control in the bass region.

For the ambitious, it is a simple procedure to disconnect the crossover entirely and bi-amplify the LS3/5As, since the crossovers are now mounted in their own external enclosure. The woofer and tweeter sensitivities are fairly close, which makes it possible to use the same model amplifiers for both sections.

The LS3/5A enclosure, because it was designed as a voice monitor, is too small and weak for good bass. Its walls are only about a centimeter thick, and insufficiently dead. You may want to build a new enclosure of at least 2 to 2½-centimeters (¾–1-inch) thick fiber board to improve the bass.

If you bi-amplify and build new enclosures, the only original parts of the LS3/5As you will have left are the T-27 tweeters and the B-110 mid/bass drivers. It would be much cheaper to buy just the raw drivers and build or purchase the other necessary items. But if you want to improve your original system, without discarding any of the original components, you can still get dramatic results by bi-wiring, removing the crossovers, removing the enclosure and tweeter grilles, sealing the enclosure and adding damping material.

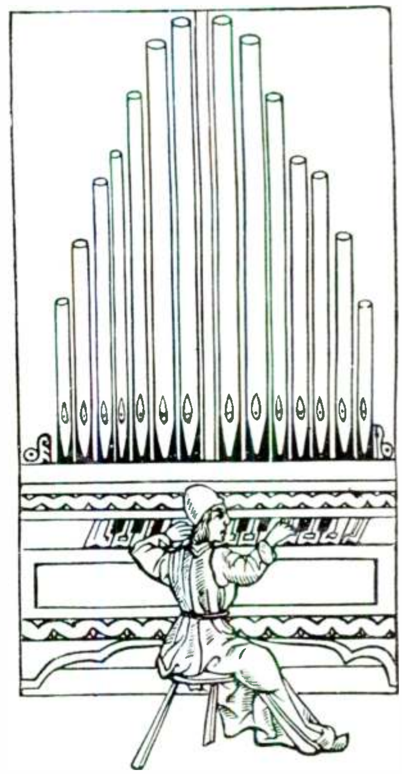
SOUND QUALITY. Your fully modified LS3/5As should show the following improvements: the high-end should be airier, less spitty and metallic; the midrange should sound more transparent; and the bass should be tauter, firmer, deeper, without boom and more controlled. Overall each section will be cleaner, smoother, more detailed and better defined. You will also notice a more three-dimensional stereo image, better instrumental and vocal positioning, and a sonic stage that has significantly better depth, height and width.

The improved LS3/5As have better top-to-bottom coherency, a more linear amplitude response and sound less dark and distant than their stock brethren. They also sound more open, and project into the listening room with better clarity and rendition of delicate "fleeting transients." Not surprisingly, you will hear lowered distortion throughout the frequency range and a reduced sense of dynamic range compression.

You may find yourself playing your improved LS3/5As much louder than before. The T-27 and D-110 have limited power handling and can be easily blown, so remember they are still small monitor loudspeakers intended for moderate listening levels in small rooms.

ABOUT THE AUTHOR

Len Hupp has a B.S. in Business Administration from the University of Missouri. He was reviewer/consultant for The Audiogram, *and wrote for and published five issues of* Audio Horizons. *In recent years he has been designing and marketing audio products and accessories under the trademark Music-Link.®*



SEVEN TRANSMISSION LINE MIDRANGE DRIVERS

BY GARY GALO
Contributing Editor

This second installment in my survey of loudspeaker components (see *SB* 2/85, p. 40 for my report on ten-inch woofers) is the result of a two-year search for an improved midrange driver for my transmission line loudspeakers (see *SB* 1/82 and 2/82). The midrange is normally defined as the area covering approximately 300Hz–5kHz. The actual range covered by a midrange driver will vary depending on the characteristics of the woofer and tweeter you choose to accompany it. The accuracy of midrange reproduction is crucial for a successful loudspeaker for two reasons. First, this is the area where the ear is most sensitive and second most of the musical information lies in the midrange region.

An accurate midrange driver will have a uniform response across its operating range, low distortion and low coloration. It must be able to reproduce the complex inner detail heard in live music. I am a believer in the potential of quality dynamic drivers. My ultimate goal is to achieve the clarity found in electrostatic loudspeakers, without the artificial "plastic" colorations inherent in many electrostatic designs.

Cones and Domes

Two choices face speaker builders when selecting a midrange driver: cone drivers and dome drivers. Both have strengths and weaknesses, and their advocates.

Cone drivers generally have low resonant frequencies, allowing crossover frequencies in the 100–400Hz range. This is desirable when using large (12-inch or greater) woofers. Normally, large woofers cannot operate linearly at midrange frequencies. However, cone drivers often exhibit a somewhat "harsh" sound in the upper midrange region (2–5kHz), and many have a rising response in the 1–3kHz region.

The best cone drivers are now manufactured with plastic cone materials, such as Bextrene or polypropylene, and usually feature a rubber or polyvinyl chloride (PVC) surround. Most cone midranges, including the four surveyed here, are not rear sealed and make sub-enclosures necessary. The advantage here is that you can choose an enclosure design according to your own philosophies. The common choices are sealed (acoustic suspension), pressure release (aperiodic) or transmission line designs. I prefer the transmission line (if you haven't noticed).

Dome drivers have higher resonant frequencies, and most cannot operate below 600–700Hz. The woofer, therefore, must cover higher frequencies than some may prefer, and virtually rules out using 12-inch (or larger) woofers in quality three-way systems. A well designed 10-inch woofer can usually operate up to 700Hz without serious problems. Even so, some may object to putting a crossover point in such a sensitive region.

Well designed domes are usually free of the upper midrange harshness exhibited by cone drivers. Two-inch units usually perform well up to 5kHz. Dome midrange drivers have better dispersion than cone drivers at higher frequencies. This dispersion is not, as is commonly believed, due to the shape of the dome, but rather due to the smaller diameter of the dome driver (most cone midranges

TABLE 1

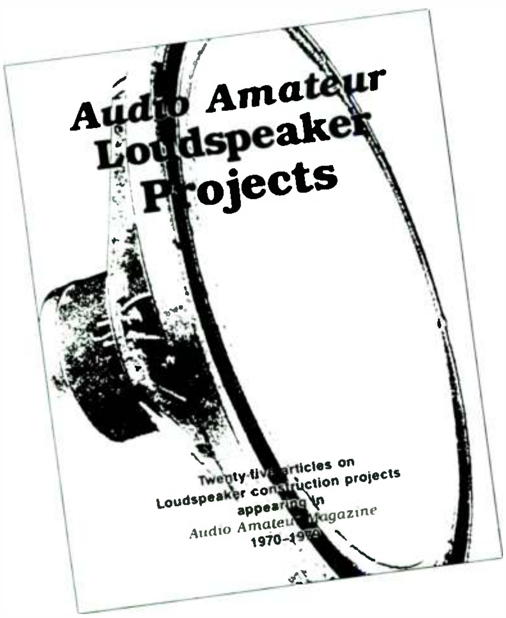
MANUFACTURER'S SPECIFICATIONS

	Nominal Imp. Ω	Resonant Freq. (F_S)	Dia. of C/D	Total Q (Q_{TS})	Sens. dB 1M/1W	Voice-coil Dia. (d)	Voice-coil length (h)	Usable Freq. Res.	Voice-coil layers	Voice-coil Ind. (Le)	Nominal Power	Notes	Price
Dynaudio D-76	8	220	3″ 75mm	NA	88	3″ 75mm	6mm	300-5kHz 6dB/oct.	2	0.2	180W		$50
Philips AD0211/SQ8	8	340	2″	NA	96	2″	3.3mm	550-5kHz 12dB/oct	NA	NA	50W >800Hz		$21
Philips AD02160/SQ4	4	360	2″	NA	NA	2″	NA	550-10kHz 12dB/oct	NA	NA	60W >800Hz		$29
Audax HD13B25H4C12	8	30	4 7/32″ 107mm	0.18	84.4	1″ 25mm	15/32″ 12mm	60-3kHz	4	NA	40W	*	$22
Audax HD13B25H2C12	8	35	″	0.29	84.2	″	″	60-3kHz	2	NA	30W	*	$50
KEF B-100	8	35	4⅛″ 110mm	0.31	96	1″ 26mm	NA	55-3kHz	NA	0.45	NA		$50
Madisound M5102	8	71	3″ 76.5mm	0.37	89.9	NA	NA	75-9kHz	2	0.23	50W	**	$15

Notes: *Usable response not specified by Audax. Data interpreted from manufacturer's actual response curve.
**Diameter (D) not specified by Madisound. Measured by author.

Audio Amateur Loudspeaker Projects

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



Contents

ORDER BLANK

NAME

STREET & NO.

CITY STATE ZIP

CREDIT CARD NO. EXPIRE

SIGNATURE

Send me ________ copies of *Audio Amateur Loudspeaker Projects* at $20.00 each, postpaid in the USA.

I enclose $ ________

☐ MC VISA ☐ ☐ Check

Add $3.00 for shipping outside USA. $7.00 for air service.

OLD COLONY BOOKS PO BOX 243, Peterborough, NH 03458 have 3–4½-inch diaphragms). Many dome drivers are also manufactured with their own sub-enclosure.

Some designers complain about coloration problems in dome units near the lower end of the operating region. I believe these problems can be overcome if the manufacturer has properly designed the sub-enclosure, and if the crossover frequency is high and the roll-off steep.

Choosing a cone or dome will depend on your requirements and philosophies. Many designs have surfaced, both good and bad, that use both types of drivers. The quality of the individual units and how they are used in a complete design is extremely important.

TABLE 2

MEASURED HARMONIC DISTORTION (1W)

	500Hz	1kHz	2kHz	4kHz
Dynaudio D-76	0.8	0.4	0.3	0.35
Philips AD0211/SQ8	1.25	0.85	0.7	1.2
Philips AD02160/SQ4	1.1	0.55	0.5	0.65
Audax HD13B25H4C12	0.95	0.4	0.45	0.4
Audax HD13B25H2C12	0.56	0.45	0.28	0.4
KEF B-110 Sample 1	0.9	1.0	0.4	0.34
Sample 2	1.0	1.9	1.0	NA
Madisound M1502	0.6	0.9	0.3	0.18

The Cone Drivers

Table 1 shows the manufacturer's specifications for the drivers I tested. The measured harmonic distortion is given in *Table 2.* I made measurements using an ElectroVoice CO-15P omni-directional condenser microphone, Heath/Morrey IG-18 generator, Heath IM-5238 AC voltmeter and a Heath IM-5258 harmonic distortion analyzer. The residual distortion of the generator and analyzer is 0.015 percent. The distortion and frequency response measurements were made in the near-field and, unless otherwise indicated, are free-field measurements (without enclosures for the cone units) and without baffle board mounting.

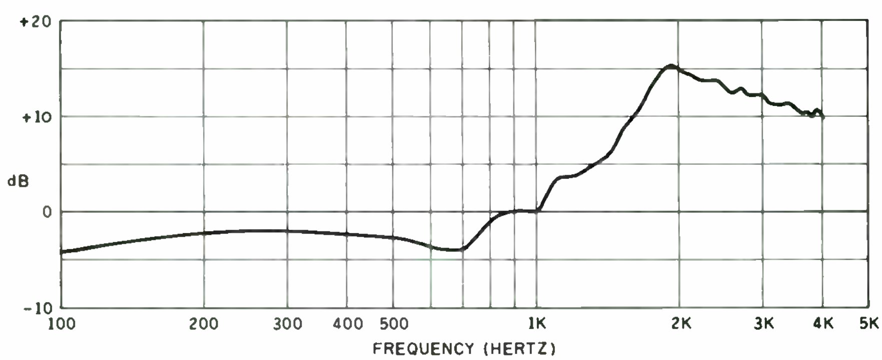


FIGURE 1: The free-field response of the Madisound M5102.

The frequency response of the Madisound M5102 driver is shown in *Fig. 1.* It is listed as a 5-inch driver, but the frame diameter is 4½-inches. It has a clear polypropylene cone and a treated foam surround. The M5102 had the most erratic frequency response of all the drivers I tested, with 15dB rise centered around 2kHz. I fear the performance of this driver, despite its bargain price of $15, is not acceptable for high fidelity use.

Larry Hitch of Madisound informs me that this driver is frequently used in automobiles, and I believe with a properly designed crossover that may be an acceptable application. I should add that my negative comments on this driver in no way reflect on other Madisound drivers. My favorable comments on their M1054 10-inch woofer in *SB* 2/85 still hold, and I have used some of their 8-inch drivers with great success.

The KEF B110 driver is a 5-inch unit with a plastic cone and a rubber surround. This driver is also disappointing. *Figure 2* shows its 9dB rise centered around 1.5kHz, and another 15dB peak around 4kHz. The most disappointing aspect of its performance is harmonic distortion.

My two samples exhibited very different distortion, with sample "B" exhibiting twice that of sample "A" at 1kHz and 2kHz. This is surprising because KEF claims to have extremely tight quality control standards. Its high price (about $50 each) makes it even less attractive. In fact, even if the B110s equaled the performance of the Audax drivers, their price would need cutting in half to make them competitive. Therefore I do not recommend the B110.

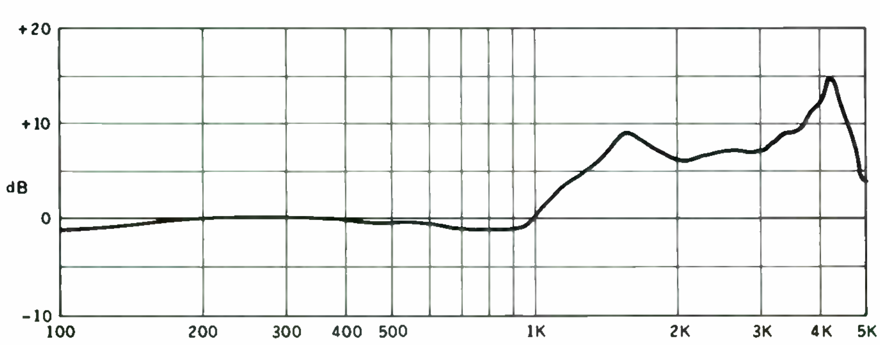


FIGURE 2: The free-field response of the KEF B110.

The Audax HD13B25H driver series consists of four 5-inch Bextrene units with PVC surrounds. The PVC surrounds are very similar in appearance and performance to butyl rubber surrounds, but I prefer the PVC. The Bextrene cones are treated with Plastiflex.®

Audax's driver numbering system can be confusing. The HD13B25H designation is followed by numbers such as 2C12 or 4C12. These designations indicate, in the 2C12 for example, a 2-layer voice coil, 12mm in length. The 4C12 has a 4-layer, 12mm voice coil. Audax also has 2C9 and 4C9 versions of these drivers. The "H" indicates a ceramic magnet 96mm in diameter. Audax's "J" series (82mm magnet) consists of four units ranging from a 2C9 to 4C12. The remaining designations refer to a 25mm voice coil, "B" for Bextrene, a 13cm overall diameter, and "HD" indicates the series.

I tested the 2C12 and 4C12 drivers. I have used both units with success in my TL-5 loudspeakers since 1982, and I've become well acquainted with their characteristics. The driver's free-field response is shown in *Figs. 3* and *4*, respectively. Both show response irregularities above 1kHz, which is common in many 5-inch cone drivers. However, both are superior to the KEF B110 in the 3–4kHz region. The 4C12 is superior in this region, and the 2C12's response falls off below 1kHz.

Mounting the drivers in the TL-5 transmission line enclosure smooths out the response peak above 1kHz (as shown in *Fig. 5* and *6*). The 2C12 errors are on the high side, while the 4C12 shows a dip in 1–3kHz response. You will need to decide which errors are the least offensive to your taste and design requirements. I prefer the 4C12.

Both Audax drivers sound very clean, and with transmission line loading they have an open-transparent character. Neither have perfect tonal balance, but I prefer the more "laid back" character of the 4C12. Both also exhibit a slight harshness in the 1–3kHz region, which is typical of many cone drivers.

I prefer transmission line enclosures for the Audax drivers. My second choice would be aperiodic enclosures, with acoustic suspension (sealed) being my last choice. I recommend 2kHz as their upper limit and 100Hz as the low. These drivers are capable of a 60Hz low response (Audax calls them "Bass-Midrange" drivers), but at greatly reduced power levels. When they are used strictly as midranges, 100Hz is a reasonable low frequency limit. I also believe the bandwidth of the driver, in actual use, should not exceed a 10:1 frequency range. This means if your upper crossover point is 2kHz you should also choose a low end crossover at 200Hz.

These drivers will operate at up to 3kHz, but I prefer a 2kHz crossover frequency for the cleanest reproduction. Of course, the builder must then select a tweeter which can operate as low as 2kHz without problems. The Audax HD100D25 1-inch dome tweeter is an excellent choice. In my opinion, the Audax tweeter is one of their best products, and at around $12 each they are a bargain.

The Audax midrange drivers, while not perfect, are far better than most of the competition and are also an excellent value. I am still searching for an "ideal" 5–6-inch midrange driver. If any of you are manufacturers, I would like to see a midrange driver with a polypropylene cone, rubber surround, and a rear-vented pole piece.

Dome Drivers

Two Philips dome drivers are covered here. My experiences with the AD-0211/SQ8 go back to 1978 when my original TL-10 loudspeaker was in its formative stages. Among its users, the AD-0211/SQ8 is considered the "classic" Philips dome. It is an updated version of the AD-0211. I have been using it in a modified version of my TL-8 loudspeaker for about two years.

Both Philips drivers contain 2-inch domes manufactured from a textile material. The AD-0211 is mounted on a slightly concave front flange. A plastic grille protects the dome from mishaps. The AD-02160 is mounted on a flat flange, and the dome is completely exposed. Each is supplied with a sub-enclosure. The AD-02160 is also available in

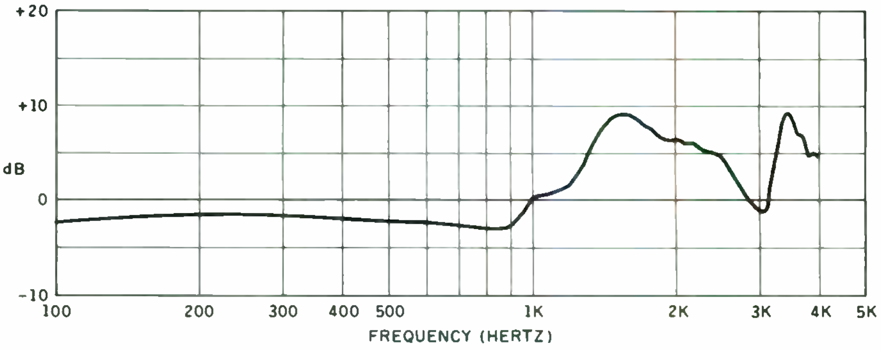


FIGURE 3: The free-field response of the Audax HD13B25H/2C12.

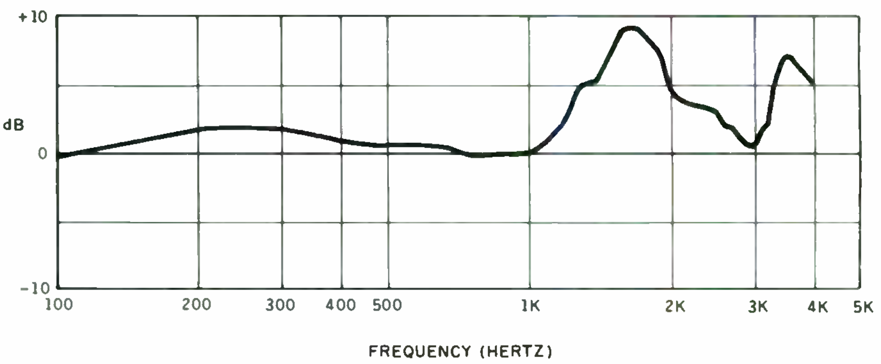


FIGURE 4: The free-field response of the Audax HD13B25H/4C12.

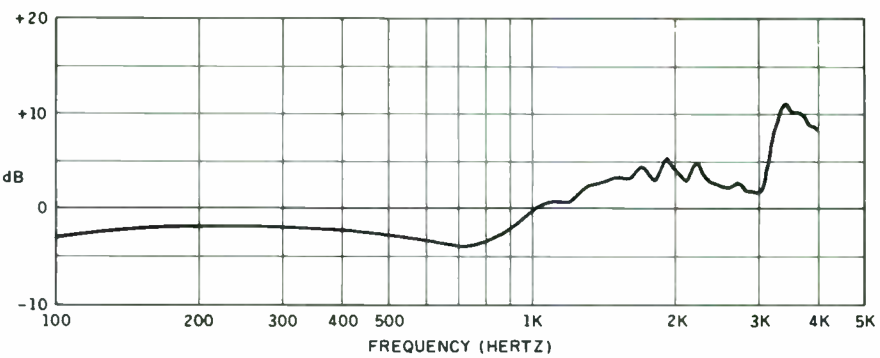


FIGURE 5: The response of the Audax HD13B25H/2C12 in a TL-5 enclosure.

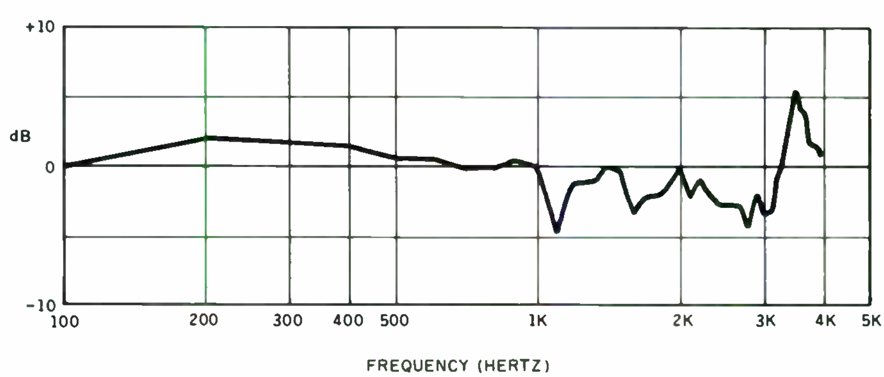


FIGURE 6: The response of the Audax HD13B25H/4C12 in a TL-5 enclosure.

an 8Ω version. The 4Ω driver is tested here.

The frequency responses and impedance curves for the two drivers are shown in *Figs. 7* and *8*. It should be obvious that these drivers are free of the response irregularities that the cone drivers exhibit. The newer AD-02160 is exceptionally smooth in the 1-5kHz region. The older AD-0211 is better below 1kHz. *Table 2* shows the newer design to have lower harmonic distortion.

Subjectively, the Philips dome drivers possess many of the qualities which have attracted loudspeaker designers to dome units. Their response is usable to 5kHz, and in this region they are smooth and well detailed. Coloration is low from 1-5kHz, but increases somewhat approaching the lower crossover frequency.

I have a few recommendations for optimum performance. Do not cross these drivers over to the woofer below 700Hz, and the roll-off must be at least 12dB per octave. First order (6dB per octave) crossovers are unacceptable, and using a lower crossover frequency, with or without a first order filter, will result in unacceptable coloration and limited power handling.

Despite Philips' claim, the satisfactory upper limit on both of these drivers is 5kHz (they claim 10kHz for the AD-02160). Again I recommend a second order (12dB per octave) crossover. I prefer even order, all-pass crossovers. If you are tri-amping, a fourth-order all-pass (24dB per octave) is the best solution at the low and high crossover points. I urge you to refer to Robert Bullock's landmark crossover series in *SB* 1, 2, 3/85, and 1/86 as the definitive source on crossover design.

These drivers perform well if properly integrated into a complete system. Although they probably do not represent the newest technology in 2-inch domes, they are a good value at their normal selling prices. If you want significantly better performance in a two-inch dome, you'll probably spend around $50 for a Dynaudio D-52.

Philips, unfortunately, provides incomplete data with their units. They omit a number of common specifications, shown in *Table 1* as NA (not available). I would like to see Philips use Audax and Dynaudio data as examples of what designers and builders need when selecting components for integration into a complete system.

I have saved the best for last. The 3-inch Dynaudio D-76 dome driver is the newest member of their line. The large diameter, combined with a fairly large sub-enclosure, results in a very low resonant frequency, which the manufacturer specifies as 220Hz.

My measured frequency response is shown in *Fig. 9a*. The D-76's most noteworthy feature is its incredibly smooth response between 1-3kHz, which is the flattest of any driver I have tested. The response does exhibit a dip below 1kHz.

Another feature of this driver is its well damped impedance peak at resonance. Impedance is 6Ω across most of its operating range, with a broad peak of 8Ω between 300 and 400Hz, as shown in *Fig.*

Continued on page 38

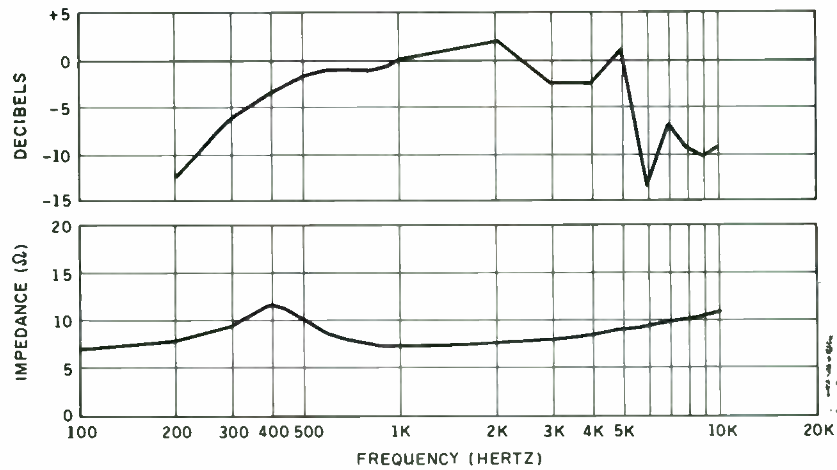


FIGURE 7: The frequency response and impedance curve of the Philips AD-0211/SQ8.

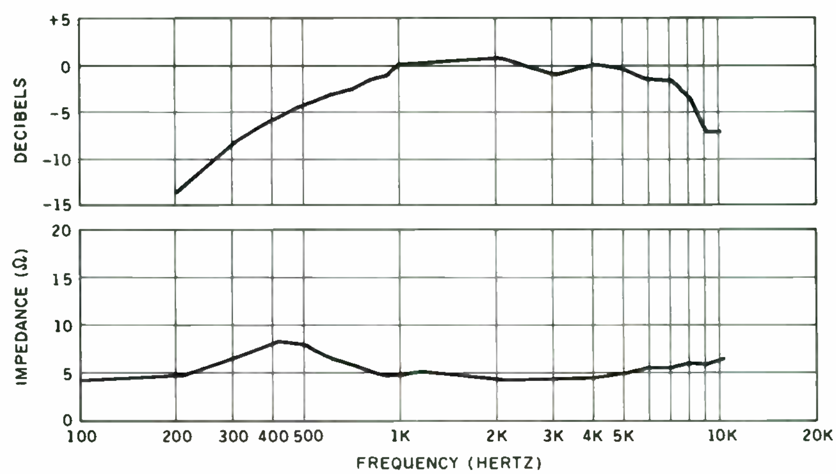


FIGURE 8: The frequency response and impedance curve of the Philips AD-02160/SQ4.

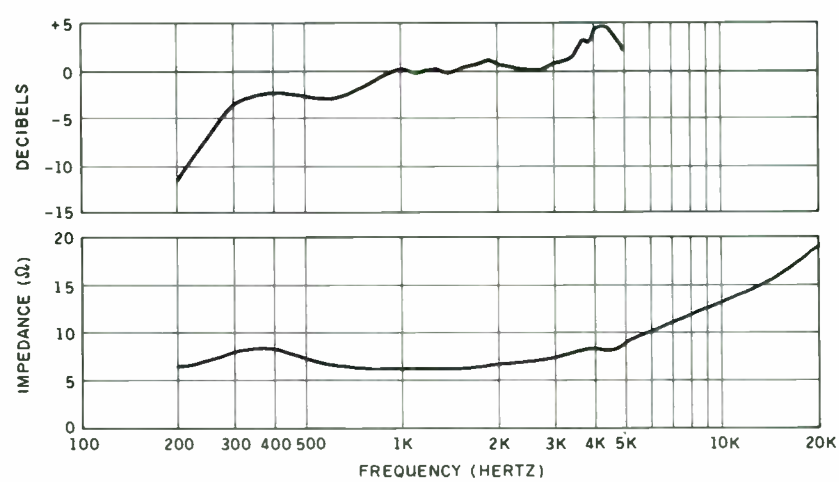
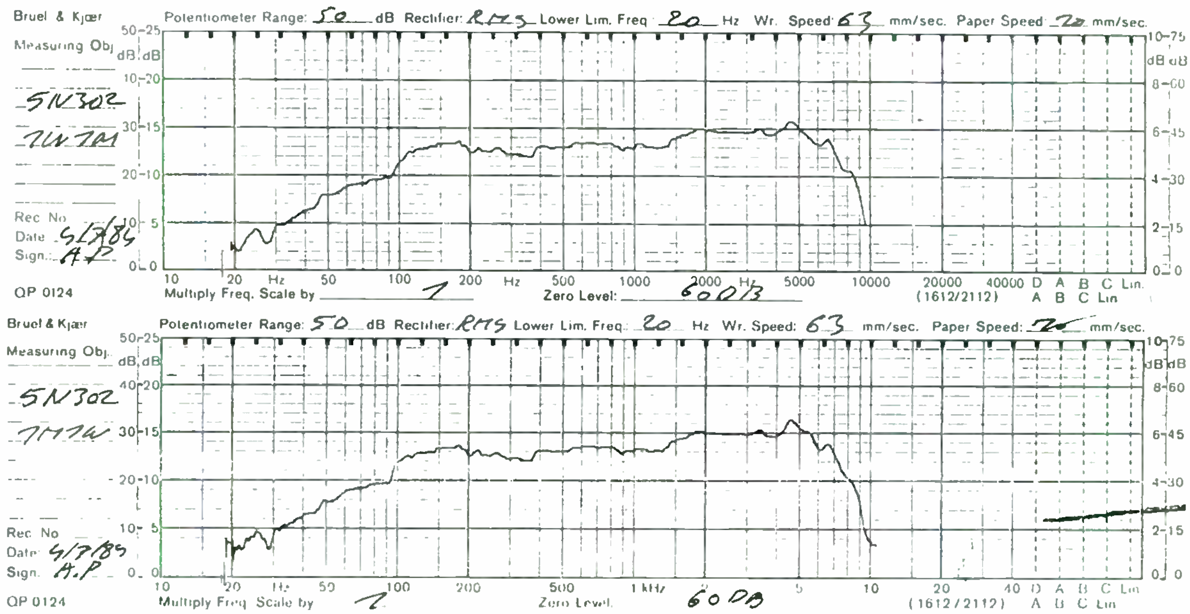


FIGURE 9: The frequency response and impedance curve of the Dynaudio D-76.

ProvenPair PRECISION DRIVE UNITS

Madisound Speakers in cooperation with France Filieres, manufacturers of Focal Loudspeakers, is proud to present Focal ProvenPair drive units.



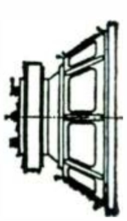
Example of 5N 302 ProvenPair Response documents:

ProvenPair units for the first time offer drive units that are carefully matched and shipped with individual response information. This can insure that the serious speaker constructor can turn his design into reality. ProvenPair drivers offer the following:

1. Frequency response matched to within 1dB throughout the range of desired use (typically one octave above resonance to the frequency where the nominal impedance is doubled).
2. Units sold only in pairs.
3. Each unit is accompanied by a signed and dated individual response curve from France Filieres. Measurement system is by Bruel & Kjaer, including a BK 4133 microphone and a BK 2307 chart recorder.

Madisound is offering the ProvenPair option on several of Focal's midranges and woofers. It is our feeling that the high frequency units do not show significant unit to unit variation, but bass and midrange speaker construction involves many hand operations and parts from several subcontractors. The quality control advantages of ProvenPair design results in a very good value for the purchaser.

MODEL		PRICE per pair
5 N 302	Neoflex midrange ProvenPair	$ 65
5 N 402dB	dual voice-coil 5″ woofer ProvenPair	68
7 N 402dB	dual voice-coil 6.5″ Neoflex ProvenPair	88
7 N 303	Lower mid 40mm flat wound coil, 93dB ProvenPair	130
7 N 501	Neoflex 6.5″ woofer; 40mm flat wound voice-coil ProvenPair	101
8 N 511	Neoflex 40mm flat wound voice-coil, cast frame, ProvenPair	106
8 P 511	Polypropylene 40mm flat wound voice-coil, cast frame, ProvenPair	110
10 N 501	Neoflex 10″ woofer, 40mm, cast frame, ProvenPair	122



MADISOUND SPEAKER COMPONENTS
8982 TABLE BLUFF ROAD
BOX 4283
MADISON, WISCONSIN 53711
PHONE (608) 831-3433

ORDERING INFORMATION: All speaker orders will be shipped promptly, if possible by UPS. COD requires a 25% prepayment and personal checks must clear before shipment. Adding 10% for shipping charges facilitates shipping procedure. (Residents of Alaska and Hawaii, and those who require Blue Label air service, please add 25%.) There is no fee for packaging or handling, and we will refund to the exact shipping charge. We accept MasterCard and Visa on mail and phone orders.

FAST REPLY #GK20

Continued from page 36
9b. With such a broad peak it is difficult to pinpoint the exact resonant frequency, but is does appear to be higher than the specified 220Hz.

For comparison see the impedance curves of the Philips domes in *Figs. 7b* and *8b*. Not only is their impedance at resonance higher than that of of the D-76, but in both cases it is sharply centered on one frequency. Note that the tested AD-02160 is a 4Ω unit.

The smooth and well-damped impedance curve of the D-76 is due, in part, to aperiodic loading of the driver. Because of the well controlled dome movement, and low-phase shift at resonance, the manufacturer claims that crossover points as low as 300Hz and 6dB per octave may be employed. I am not an advocate of odd-order crossovers for the same reasons specified by Siegfried Linkwitz, and others (see *SB* 4/80, p. 14). I am convinced that this driver, like other domes, would be better off with a second or higher order filter.

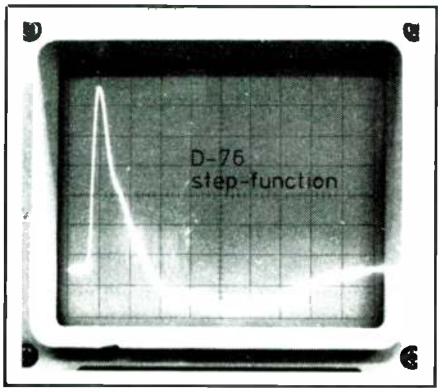


FIGURE 10a: The Dynaudio D-76's step function.*

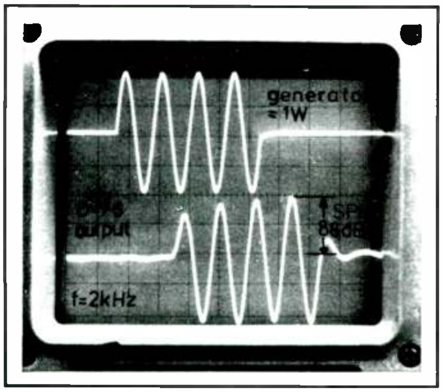


FIGURE 10b: Notice the 1W tone burst of the D-76.*

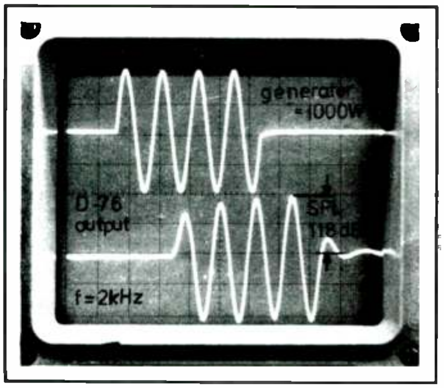


FIGURE 10c: A 1kW tone burst of the D-76. The upper trace is generator output and the bottom trace is driver response.*

*Manufacturer's photo

In my tri-amped TL-10 (revised) I use this driver with a fourth-order (24dB per octave) all-pass crossover (per Linkwitz). I am also experimenting with a second-order, all-pass, passive crossover for this driver. Since fourth-order passive crossovers are very cumbersome to design and implement, I believe second-order, all-pass crossovers are best for those who do not wish to tri-amplify. Incidentally, if you bi-amplify your system, design your electronic woofer-to-midrange crossover with the same roll-off as the passive midrange-to-tweeter crossover to avoid phasing problems between drivers.

Table 2 gives the harmonic distortion measurements for the D-76. While they are not as low as claimed by Dynaudio (less than 0.3 percent), they are significantly lower than the Philips domes. Dynaudio claims a usable response to 5kHz and, although the distortion is quite low above 3kHz, it no longer exhibits smooth response in this region. Therefore, I recommend 3kHz as the upper frequency limit for the driver.

The sound quality of the D-76 is extraordinary. In many ways it combines the best virtues of cone and dome drivers. The 1–3kHz region is detailed, and free of cone-type harshness. At the same time, between 400Hz and 1kHz, the driver does not exhibit the colorations often found in dome drivers. In this region, its sound is remarkablely "open."

What is particularly attractive about a 300–400Hz crossover frequency is that a midrange dome can now be crossed over to a larger woofer (i.e., 12-inch) without the usual problems encountered when such woofers are forced to operate up to 700–800Hz. Of course, the quality of the woofer is extremely important here and preference should be given to polypropylene or other synthetic materials of similar performance.

The D-76 is a "fast" driver, with excellent transient response and dynamic capabilities. Dynaudio's data sheet shows remarkable oscilloscope photos of step-function response, and 2kHz tone bursts at 1kW power levels (reproduced in *Figs. 10a, 10b* and *10c*). Note that the 1kW tone burst is virtually indistinguishable from the one at 1W. I did not confirm these measurements, but the D-76 is audibly superior to the other drivers listed here in transient detail and clarity on very loud material.

In short, the Dynaudio D-76 is my first choice among midrange drivers—as of this writing. It is a clean, smooth, natural sounding driver, with excellent detail and clarity even at loud listening levels. To an opera lover like myself, vocal accuracy is extremely important. Here the D-76 is also excellent. Voices are reproduced with their natural harmonic characteristics intact, free of strain or edginess.

For speaker builders, the ability to cross a dome midrange to the woofer, as low as 300Hz, opens up previously unrealizable possibilities. I hope many readers will explore them.

SOURCES

Madisound Speaker Components
8982 Table Bluff Rd., Box 4283
Madison, WI 53711

Meniscus Systems
3275 Gladiola SW
Wyoming, MI 49509
(All except Madisound M5102)

TEST RECORDS

PERFORMANCE TESTS WITHOUT INSTRUMENTS

CBS LABORATORIES' STR PROFESSIONAL TEST RECORDS

For over ten years, the original series of these high-precision test records set a standard for the audio industry. The new series sets an even higher standard. It's been revised, recut and expanded.

The new series consists of eight records for professionals and one for non-professional audiophiles.

Each record contains a complete series of easy-to-use tests to help you rapidly and accurately evaluate components and systems. Even one of these records can eliminate the need for costly, additional equipment. Each will find productive use and save you hours in the laboratory, on the production line and in field testing.

OLD COLONY SOUND LAB · OLD COLONY SOUND LAB · OLD COLONY SOUND LAB

TAKE A LOOK AT WHAT THIS ESSENTIAL TESTING SERIES CONTAINS:

STR 101 SEVEN STEPS TO BETTER LISTENING. For only $6.98, you can improve your system with CBS Laboratories' "Seven Steps to Better Listening." This high-precision test record enables you to make sure that your equipment functions properly . . . to tune your system to your ears and your room acoustics. Included is a detailed 16-page booklet by *Audio's* Edward Tatnall Canby explaining how to use the record to improve your system's performance. With the record you can perform the following "ears alone" tests: left-right identification, phasing, loudspeaker balance, tone control setting, alternate phasing, buzz and rattle elimination, lateral tracking, and vertical tracking.

STR 100 STEREOPHONIC FREQUENCY TEST RECORD. Designed for the evaluation of pickups and systems. Provides a constant amplitude characteristic below 500Hz and a constant velocity characteristic above 500Hz. Tests include: Sweep Frequency—with the sweep rate synchronized for use with a graphic level recorder; Spot Frequency—with voice announcements; Channel Separation; Wavelength Loss and Stylus Wear—to pinpoint oversize or worn-out styli, and excessive pickup tracking force; Compliance; Phasing; Vertical and Lateral Tracking: Tone Arm Resonance—to check system performance at low and subaudible frequencies and thus reveal undamped resonance which may cause equipment overloading.

STR 112 SQUARE WAVE, TRACKING AND INTERMODULATION TEST RECORD. Enables detailed study of tracking capabilities of stereophonic phonograph pickups. The square wave modulation allows a rapid appraisal of stylus-tip mass, damping, and tracking. Low frequency compliance and tracking are determined by means of 300Hz bands of progressively increasing amplitude. Intermodulation distortion measurements are made possible by graduated 200Hz intermodulation test bands. The STR 112 has been cut with vertical angle approximately 15° which is representative of current recording practice.

STR 120 WIDE RANGE PICKUP RESPONSE TEST RECORD. Makes possible the measurement of pickup response at frequencies far beyond the audible range, where elusive distortion elements can cause audible distortion. The low-frequency range includes glide-tones at twice normal level for the detection and elimination of arm resonance, loudspeaker cone and cabinet rattles. Other tests include: silent grooves for measuring rumble and surface noise characteristics; and standard level bands at 0dB for overall system S/N measurements. This record is suitable for use with a graphic level recorder to provide permanent, visible records for precise evaluation.

STR 130 RIAA FREQUENCY RESPONSE TEST RECORD. Provides RIAA frequency characteristics for the calibration of professional recording equipment and for testing the response of professional and consumer record reproduction equipment. This record is suitable for use with a graphic level recorder to provide permanent, visible records for precise evaluation. Spot frequency bands for use without automatic equipment are included.

STR 140 RIAA PINK NOISE ACOUSTICAL TEST RECORD. Designed for acoustical testing of systems and loudspeakers and for psychoacoustic tests on reproduction equipment. With the STR 140 it becomes possible to test loudspeakers in the room in which they will be used. Spot frequency tones with voice announcements facilitate the testing procedure. Continuous glide-tones in ⅓-octave bands cover the frequency range from 30 to 15,000Hz and are synchronized with a graphic level recorder.

STR 151 BROADCAST TEST RECORD. Developed especially to meet the needs of broadcast engineers, audiophiles, and other professionals seeking a convenient signal source for the testing and adjustment of all audio equipment. Tests include: phonograph pickup response and separation, speed accuracy at 33⅓ and 45 rpm, wow and flutter, rumble and hum detection, ballistic test of VU meters and many others.

STR 170 FREQUENCY RESPONSE 318 MICROSECOND TEST RECORD. Provides pickup designers and recording studios with a high-level, easily-equalized signal for frequency response and channel separation measurements. The STR 170 employs a 318 microsecond characteristic corresponding to the "test" or "flat" mode common to most disc recording equipment. Constant amplitude recording is employed in the region below 500Hz with constant velocity recording in the region above. The transition is smooth, in contrast with the STR 100 which employs a sharp breakpoint at 500Hz. The record is suitable for use with a graphic level recorder to provide permanent, visible records for precise evaluation.

SQT 1100 QUADRAPHONIC TEST RECORD. Designed for calibration, verification, and adjustment of SQ® decoding equipment. The record provides test bands for pickup measurements, for adjustment of decoder electronics and for channel identification and balance. Each band is described in terms of recorded characteristics and its intended use.

SQ is a Trademark of CBS Inc.

OLD COLONY SOUND LAB
P.O. Box 243, Peterborough, NH 03458
☐ Check ☐ Visa/MC
Credit Card # ____________________

Expiration Date ____________________

Name ____________________

Street ____________________

City __________ State __________ Zip __________

Phone Orders (VISA/MC only)
(603) 924-6371 9-4 EST, M-F

Quantities are limited. Be sure to order enough for many years of use.

Send me the following test records:

_______copies (STR 101) Seven Steps to Better Listening at $6.98 ea.
_______copies (STR 100) Stereophonic Frequency at $10. ea.
_______copies (STR 112) Sq. Wave, Tracking and Intermodulation $15. ea.
_______copies (STR 120) Wide Range Pickup Response at $15 ea.
_______copies (STR 130) RIAA Frequency Response at $15 ea.
_______copies (STR 140) RIAA Pink Noise Acoustical at $15 ea.
_______copies (STR 151) Broadcast Test at $15 ea.
_______copies (STR 170) 318 Microsecond Frequency Response $15 ea.
_______copies (SQT 1100) Quadraphonic Test at $15 ea.

Tools, Tips & Techniques

SMALL DRIVER ARRAYS—FLEXIBLE & EFFICIENT

Using an array of small, identical drivers in a series or parallel arrangement allows you to obtain the efficiency and power handling of a single large driver, while retaining the high-frequency performance of a small driver. This allows nearly full-range performance without crossovers, and added flexibility in choosing cabinet dimensions. Using this array of small drivers may result in restricted high-frequency dispersion and lobing problems in the vertical or horizontal plane, depending on their configuration (vertical line, horizontal line or "bunched").

To obtain the uniform dispersion of a single small driver, you may design a "crossover" (*Fig. 1*) that allows all drivers to function at low frequencies (so that power handling is not compromised), but only one driver to function at high frequencies.

For example, suppose you are using an array of four 3-inch full-range drivers in a very narrow, ported enclosure. The Audax HIF78BiSM and AM78GSM look excellent. From tables for vented enclosures,[1] it appears that an f_3 of 65Hz is possible, with about 1dB ripple. *Figure 2* shows the driver configuration and "crossover" design for a simple system.

The required capacitor value C can be obtained by calculating the equivalent values of R1 and R2 in *Fig. 1*. In our case,

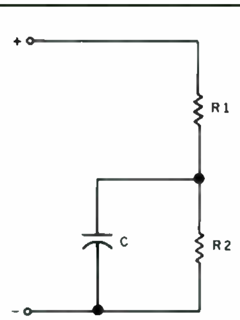


FIGURE 1: This crossover arrangement allows all the drivers to function at low frequencies, but only one driver to function at high frequencies.

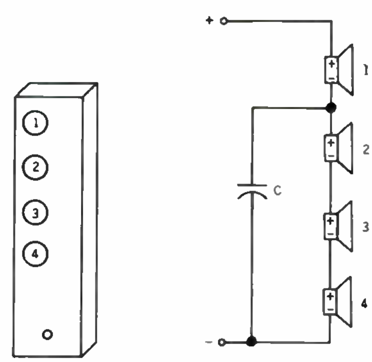


FIGURE 2: This configuration will have the same frequency response, dispersion and sensitivity as a single 3-inch driver, but four times the impedance below f_c, four times the efficiency and four times the power handling.

R1 corresponds to the impedance of driver 1 (e.g., 8Ω), and R2 corresponds to the impedance of drivers 2, 3 and 4 in series (24Ω). To eliminate drivers 2, 3 and 4 above f_c (1kHz is probably a good cutoff point), you can determine C as follows:

$$C = \frac{R1 + R2}{2\pi f_c R1R2} \qquad (1)$$

The system in *Fig. 2* will have the same frequency response, dispersion and sensitivity as a single 3-inch driver, but four times the impedance below f_c, four times the efficiency, and four times the power handling. In other words, the maximum sound-pressure level (SPL) obtainable is 12dB higher. *(Note that the load on your amplifier is changed radically, and you must take this into account.—Ed.)*

Note that in the series arrangement additional drivers increase power handling, not sensitivity, below f_c. Sensitivity remains the same for the single driver above f_c. You can further increase power handling and efficiency by inserting additional drivers and adjusting C correspondingly. Incidentally, this "crossover" is of the "constant-voltage" sort, so it does not impair the linear-phase characteristic of the system.

Another interesting arrangement is shown in *Fig. 3*. Here, drivers 1 and 2 operate above f_c and are crossed over, with first-order slopes, to a dome tweeter at about 5kHz for added high-frequency smoothness and dispersion. The 3-inch Audax driver's response becomes somewhat irregular, and its off-axis response deviates above 8kHz. Placing the tweeter between drivers 1 and 2 provides a symmetrical vertical dispersion pattern around the crossover frequency.[2]

A narrow cabinet face relegates diffraction effects to higher frequencies, where the wavelength is comparable to or smaller than the cabinet width. Since a small tweeter has good dispersion at high frequencies, you should take steps to control its diffraction. You may do this by rounding adjacent cabinet edges or restricting its dispersion with acoustic foam on the cabinet face. If f_c is chosen sufficiently far (two octaves or more) below the tweeter crossover frequency (f_T), you may use the parallel arrangement in *Fig. 3a*, with the following values:

$$C2 = \frac{1}{2\pi f_T R_T} \qquad (2)$$

$$L = \frac{0.5\, R_F}{2\pi f_T} \qquad (3)$$

where R_T is the tweeter's nominal impedance, R_F is a full-range driver's impedance, and $0.5R_F$ is the impedance of the two full-range drivers operating in parallel above f_c. *(See box—Ed.)* If necessary use impedance compensation circuits for the tweeter, or drivers 1 and 2.[3]

Referring again to *Fig. 1*, you now have R1 = R2 = the impedance of two drivers in parallel (equals 4Ω if each driver is 8Ω). Determine C1 from equation (1) for the desired f_c.

The resulting system will have all four full-range drivers operating at low frequencies: this time with a net impedance equal to that of a single driver, 6dB higher sensitivity and four times the power handling. At frequencies between f_c and f_T, impedance falls to one-half that of a single driver, while sensitivity remains at low frequencies. Above f_T, impedance is

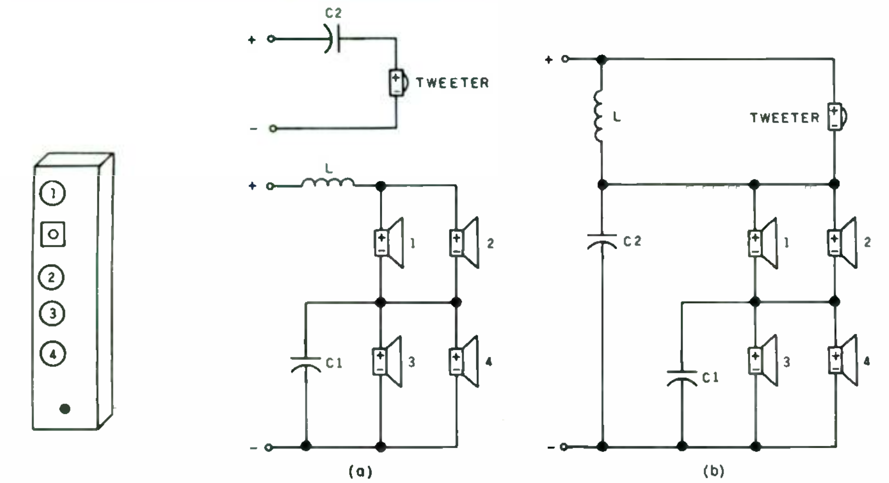
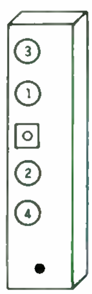


FIGURE 3: The series (a) and parallel (b) arrangements of this circuit allow drivers 1 and 2 to operate above f_c. They are crossed over, with first-order slopes, to a dome tweeter at about 5kHz.

with the tweeter, which must have 6dB higher sensitivity than a single full-range driver for proper integration. Due to the increased power handling and efficiency, SPLs may be 12dB higher than with a single driver.

If f_c is close to f_T, the series arrangement in *Fig. 3a* can be solved for an exact solution, but in this case little is gained from f_c's existence. You may omit C1 in *Fig. 3* and use an ordinary series/parallel arrangement of the full-range drivers. This still allows a 6dB increase over the sensitivity of a single full-range driver, but all four now operate up to the tweeter crossover frequency, impairing midrange vertical dispersion.



To eliminate polar axis wander around f_c, you may reconfigure driver arrangement as shown above.—Ed.

Finally, note that you may use the techniques described here to increase a midrange's or tweeter's power handling, without compromising its dispersion, by using several drivers and a suitable f_c. It helps if the drivers' flanges are small and can be packed closely. Again, if f_c is close to a "real" crossover frequency, the changing impedance may complicate the crossover. Perhaps someone can come up with suitable impedance compensation circuits. Note that you must use the impedance of the entire array to calculate the high-pass filter for a midrange or tweeter array. I would be interested in other readers' experiences with these methods.

Ralph Gonzalez
Philadelphia, PA 19143

REFERENCES

1. Bullock, Robert M., "Thiele, Small and Vented Loudspeaker Design," *SB* 4/80, p. 7.
2. D'Appolito, Joseph, "A High-Power Satellite Speaker," *SB* 4/84, p. 7.
3. Knittel, Max R., "Impedance Compensating Crossover," *SB* 1/83, p. 11.
* See also: Bullock, Robert M., "Passive Crossover Networks, Part I," *SB* 1/85, p. 13.

TWO CROSSOVER CUES

Whenever more than one inductor is to be mounted on a crossover board, their magnetic fields may interact and cause a significant change in their normal inductance value. Iron-cored inductors with open magnetic circuits are especially sensitive. To minimize the possibility of such interactions, place the inductors on the board so their axes are at right angles and space them as far apart as practical.

Until recently I used silicone to secure capacitors and inductors to crossover boards and lived with the nuisance of its long curing time. Now I use a glue gun, which lays down a thick, fast drying bead that makes it easy to hand position parts while the adhesive sets.

David J. Meraner
Scotia, NY 12302

SHY AUTHORS
SEND YOUR LETTER TO THEM VIA US—with your STAMPED ENVELOPE

Craftsman's Corner

Webb Hybrid

After much planning and an enormous effort, my idea for a combined B.J. Webb transmission line speaker/RTR ESR-6 system has become a reality. *Figure 1* is a drawing of my modification of the Webb system using only the woofer (KEF B139) and the midrange (B110). *Figure 2* shows the ESR-6 power supply and coupling circuitry. I plan to bi-amplify the system using my modified Stereo 70 for the ESR-6 supplies—with a passive, high-pass input filter so that the ESR-6 units may be driven directly. Now I drive the system's full range with the modified Stereo 70 from the 4Ω taps. The results are quite good, with a decent dynamic range. To do this, I had to modify the Webb crossover. (See *Fig. 3*.)

Figure 4 shows the almost-finished speaker. When I snapped the picture, I had not yet constructed a grille cloth frame or applied wood veneer to the sides and front edges of the enclosure. I did install Stereophile-inspired cork baffling in the ESR-6 units. I consider this essential. I also used open-pore foam to absorb rear radiation and obtained long-fiber wool for stuffing from a local handweaving shop. I ordered the KEF drivers from Wilmslow Audio in England.

Construction was delayed by my inability to find an adequate grille cloth. I had admired Dynaco's light-colored, open-weave fabric for years, but the company would not sell it in bulk, uncut quantities. Consequently, I decided to do without a grille cloth.

The results with the new speakers are phenomenal. Overall detail is excellent, with deep, clean bass and fine midrange and high-frequency reproduction. Stereo imaging and dynamic range are equally superb. The speakers also reveal gain riding, limiting and compression on recordings more clearly than ever before.

These speakers represent one of the more satisfying construction projects I have undertaken. My experience with this project leads me to believe that the original Webb design (*TAA* 1/73, p. 7) using all dynamic units is astonishingly good, and I would recommend it to anyone not wanting to deal with electrostatic speakers.

Robert K. LeBeck Jr.
Mountain View, CA 94040

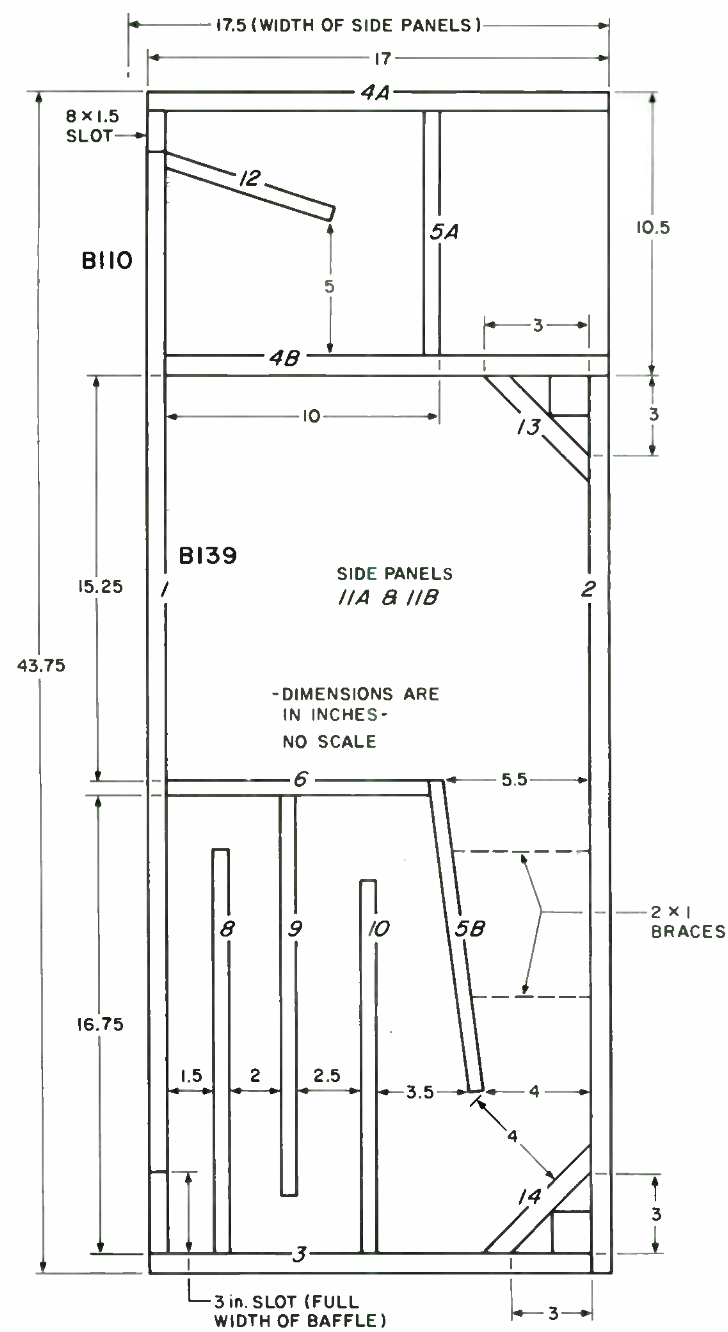
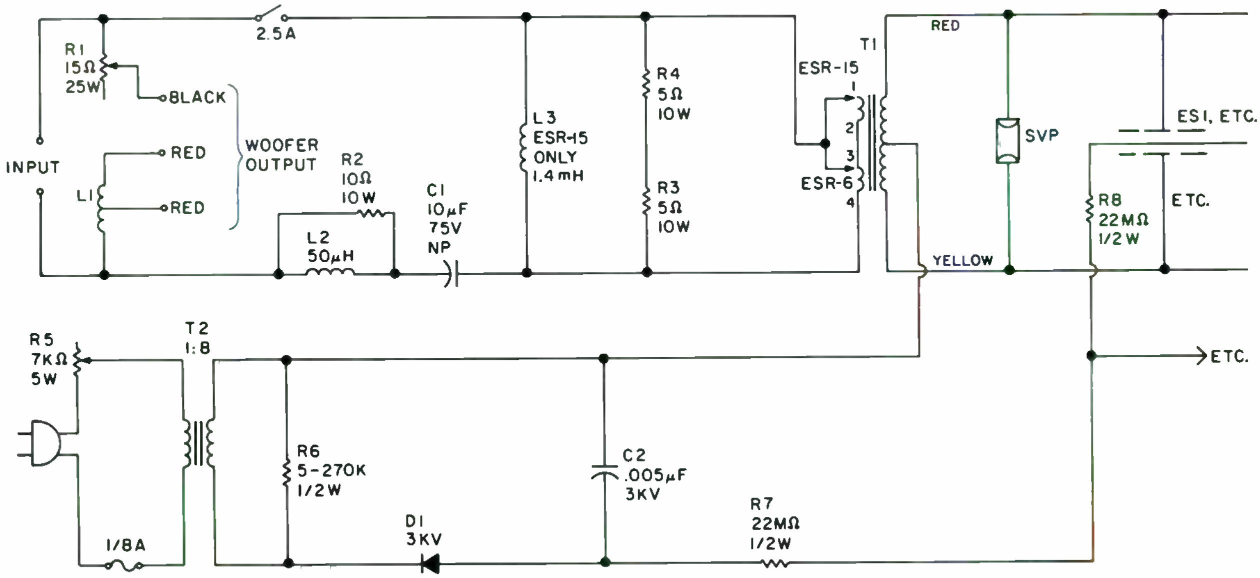


FIGURE 1: Plans for the modified B.J. Webb transmission line speaker design for use with the RTR ESR-6 power supply.



L1, 0.5mH (4Ω); 1.0mH (8Ω)
T1, 1:83 (ESR-6); 1:50 (ESR-15)
SVP, Surge Voltage Protector

Note: For 3,500Hz crossover, RTR recommended 5 to 6μF for C1.

FIGURE 2: Schematic of the ESR-6 and ESR-15 power supply.

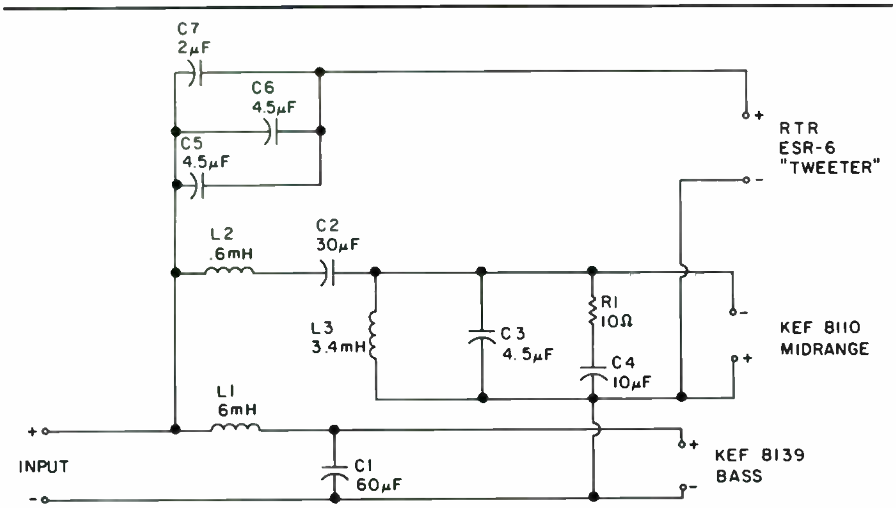


FIGURE 3: Mr. LeBeck modified the Webb crossover to use with the ESR-6s.

Dimensions of the Webb Hybrid Enclosure (See *Figure 1*)

¾" Board

1—39½ x 12⁷⁄₁₆
2—33¼ x 12½
3—16¼ x 12½
4A—17 x 12½
11A & 11B—43¾ x 17½

½" Board

5A—9 x 12½
5B—11⅝ x 12½
6—9½ x 12½
8—14¾ x 12½
9—14¾ x 12½

10—13¾ x 12½
13 & 14—5½ x 12½
12—6½ x 12½

4—½" dowels, 12½" each

T-brace made from 2 x 2, 12½" & 8"

Width of finished speaker: 14"
Standing on 1½" square feet (4): ¾" thick
Total height with ESR-6: 59"
The original spacing between B110 & B139 is preserved, relative to foam slot at top.

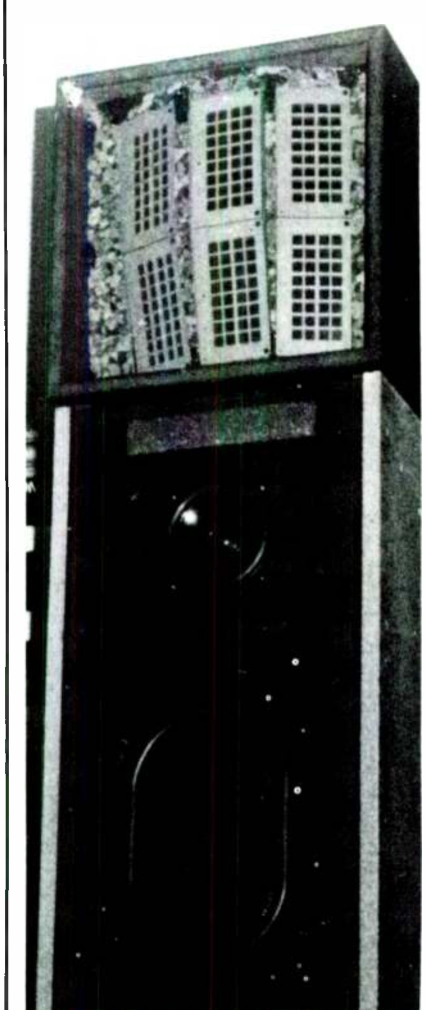


FIGURE 4: The author removed the grille cloth panel on the ESR-6 for photographic purposes only.

SB *Mailbox*

CORRECTION: EDGAR MIDRANGE HORN

Some confusion has resulted due to an error on page 14 of the 1/86 issue, in "The Edgar Midrange Horn" article. The first two lines of the last paragraph at the bottom of column three, the words "Putting the Polydax dome on ½-inch standoffs helped somewhat, but filling" should be the last two lines at the bottom of that column with the sentence completed in column one of p. 15. Please make a note of the change in your 1/86 issue for future reference.

We apologize for any confusion that may have resulted from this error.

CORRECTION: SZEKERES T,T & T

In my *Tools, Tips and Techniques, SB* 1/86, p. 34 submission, an additional capacitor, C9, must be installed between R12 and the bridge, with the positive end toward R12. This capacitor may be the same value as C6, 47μF, 10 to 15V. Also, when using the optional LED (light emitting diode) meter use a separate isolated battery supply.

Some question may arise about the value of this meter versus some commercially available meters (i.e. Radio Shack). When this meter is used with the Panasonic mike element it provides a flat response versus one with only weighted scales. No conversion corrections are needed to account for response variations. The readings are responsive to peak levels, however, this is not to be taken as instantaneous due to a slight time delay.

Greg Szekeres
Pittsburgh, PA 15236

ON COMPOUND WOOFERS

The compound woofer configuration (see *SB Mailbox* 4/85, p. 51) was invented by Dr. Harry Olson, sometime before 1951. It was extensively investigated by Olson and his colleagues John Preston and E.G. May, at RCA Laboratories, both by analog computer simulation and by building and measuring a number of prototypes. Olson's classic textbook, *Acoustical Engineering* (D. Van Nostrand, Princeton, 1957; see section 6.11, pp. 157-159), contains a complete analysis of this design.

In the 1940s, direct-radiator woofers were usually mounted on baffles or in open-back boxes and, accordingly, they had very large values of V_{AS}. For instance, a 12-inch RCA woofer (similar to that shown in Olson's drawing on p. 158) produced the $Q_{TC}=2.5$ alignment with resonance frequency about 113Hz, when mounted in a 2 cubic foot undamped closed box.

To achieve the B2 alignment, the box volume must be increased to something like 23 cubic feet and the maximum-power-handling $Q_{TC}=1.1$ alignment, which is what Olson wanted, still requires a box volume of 8.4 cubic feet. To incorporate a closed-box woofer into a "hi fidelity" console, requires a way to greatly increase the effective volume of the closed box.

Mounting a second identical auxiliary driver on an internal baffle, with the two voice-coils connected in series, greatly reduces the stiffness presented to the cone of the radiating driver by the air behind it, and is therefore, essentially equivalent to mounting the radiating driver, alone, in a very large box.

To obtain the desired response, it is now necessary to decrease the effective volume of the box, down to the required 8.4 cubic feet. Olson accomplished this by reducing the size of the auxiliary driver. His final optimized design used a special high-compliance 5-inch auxiliary driver and produced the desired $Q_{TC}=1.1$ alignment in a box of 2 cubic feet total volume, with a 3dB down frequency below 40Hz.

Olson's only design aim was to increase the effective volume of the enclosure. Olson evaluated the performance characteristics of his compound woofer design and found these as regards response, efficiency and distortion to be the "same" as those of the radiating driver, operating alone in a much larger closed box. The power handling ability of this design is, however, displacement-limited by the auxiliary driver, which, in the specific example of the 12-inch radiating and 5-inch auxiliary drivers, makes four times the excursion of the radiating driver. Thus, the entire practical effect of Olson's compound woofer design is to exchange reduced box size for reduced power handling ability.

The purported aim of Ivor Tiefenbrun's design (which is protected by US Patent #4,008,374, February 15, 1977) is entirely different. This design has two essential features. First, the two drivers are to be operated in such a way that the pressure, in the chamber between them, remains "substantially constant." Although the figures suggest that the two drivers are to be identical, Tiefenbrun tells us that the "salient feature of the ... system is that the bass drivers respond in such a way as to maintain the "Isobarik" Chamber free from sound pressure and it is conceivable that this can be done with dissimilar bass drivers...," and in other ways, and that "full-potential of the .... system is realized when the relative sizes of the various chambers are optimized." This would, of course, require simulation, based on Olson's analysis.

Second, the two drivers are to be separated by an "absorptive curtain," mounted between them or an "Isobarik" Chamber is to be filled with "absorbent material." This is supposed "to absorb distortion components resulting from the different response characteristics" of the two drivers. "Any distortion components, which the rear driver produces from its front face, will constitute virtually the only sound pressure in the "Isobarik Chamber." Tiefenbrun tells us, contrary to what Olson found by measurement, that "virtually all of the output of the rear bass driver is absorbed in creating "ideal" conditions for the front driver. The resultant effect is the production of pure and undistorted sound which is extended downward to the free air resonance of the front bass drivers and even below that level. The sound is also virtually free from coloration and anti-phase effects." In his review of this patent (see *Journal* of the

Continued on page 46

A Wealth of Useful Construction Information is Available in These Back Issues

1980

Realism Testing • Ambience Reproduction • Upgrade Mods for Small, Cheap Speakers • Diffraction and Cabinet Shape • Speaker Cables: Measurable and Audible Differences • A Three Enclosure System with Active Crossover and Delay (in 3 parts) • Electrostatic Speaker Construction Project (in 3 parts) • Double Chamber Enclosure: Deep Bass from an 8″ Woofer • Corner Midrange Horn • Thiele, Small and Vented Speaker Design: Part I •

1981

A Testing Unit for Speaker Parameters • Variable-Volume Enclosure • Thiele/Small Theory (in 3 parts) • Easy to Make Enclosure Using Concrete Blocks • The Tractrix Horn: Good Dispersion Bass From an Old Design • Diffuser Port for Small Boxes • Mini-Speaker Made From PVC Tubes • Closed vs. Vented Box Efficiency • Interview with P.G.A.H. Voigt • Dual 8″ Symmetrical Air Friction Enclosure • Thiele/Small Calculator Computations • Thiele/Small Parameters for Passive Radiators •

1984

Build an Aligned Satellite/Woofer System • BOXRESPONSE: A Program to Calculate Thiele/Small Parameters • Casting with Resins • A Phase Meter • An Interview with Ted Jordan • Building the Jordan-5 System • Self-Powered Peak Power Indicator • Closed Box Design Trade-offs • How to Build Ribbon Tweeters • Build a Dual Measurement Impedance Meter • A High-Power Satellite Speaker System • Build and Use a White/Pink Generator • Sound Pressure Level Nomographs •

1982

Transmission Line Theory • Thiele/Small Sixth Order Alignments • The Quad 63 • Table Saw Basics • AR-1 Modifications • Active Crossovers and Phase • Three Transmission Line Speakers • A Beginner's First Speaker • How Passive Networks Interact with Drivers • Horn Loaded Heil • Phase Correcting Active Crossover • Wind Your Own Inductors • Series and Parallel Networks • High Performance Corner Speaker • Using Zobels to Compensate for Driver Characteristics •

1983

Building the Two-Way Dynaudio • A Crossover That Offsets Speaker Impedance • Using a Calculator for Box Design • Choosing a Calculator • A Simple Peak Power Indicator • A Small Horn Speaker • Audio Pulse Generator • How to Use Speaker Pads and Level Controls • An Easy-to-Build Voltmeter for Speaker Measuring • Nomograms for Easy Design Calculations • Interview with KEF's Raymond Cooke • Build a Simple Wattmeter • A New Type of Speaker Driver •

1985

Compact Transmission Line Subwoofer • Bullock Crossovers, Passive and Active, three parts • Drive Attenuator, Computer-Designed • Curved Vertical Array • Sontek Powered Subwoofer • An Isobarik System • Modifying Strathearn's Ribbon • Ambience Systems • Experimental Transmission Line • Small Double-Chamber Reflex • Loose Walled Speaker • Modifying the Daline •

BACK ISSUES ORDER FORM

Please send:

- ☐ The 1985 series @ $12
- ☐ The 1984 series @ $10
- ☐ The 1983 series @ $10
- ☐ The 1982 series @ $10
- ☐ The 1981 series @ $10
- ☐ The 1980 series @ $10

☐ Check enclosed

☐ Charge to my ☐ MC ☐ Visa

My Magic Number ☐☐☐☐☐

Name ______________________

Street & No. ______________________

Town ____________ State ________ Zip ________

Card # ______________________ Expires______

Signature ______________________

Ppd. USA. Outside USA add $4 for postage. UPS delivery add $2. Remit in US funds only.

Speaker Builder Magazine PO Box 494, Peterborough, NH 03458-0494 USA
(MC/Visa phone orders (603) 924-9464)

SB Mailbox

Continued from page 44

Acoustical Society of America (64)3, Sept, 1978, p. 971), George L. Augspurger summed it all up by saying "If you believe that, let me tell you about this bridge I have for sale."

Dr. Eugene Zaustinsky
E. Northport, NY 11731

RIBBON TWEETER QUESTIONS

I would like authors Lampton and Primbsch to comment on my questions concerning their "Simple Ribbon Tweeter" article in *SB* 3/84 (p. 7). It is known that decreasing the mass of a ribbon tweeter's diaphragm by making it thinner will increase the sensitivity of the transducer. Will I also be able to increase sensitivity by using stronger magnets exerting a stronger magnetic force on the ribbon's diaphragm, and will this sensitivity manifest itself by a noticeable increase in response to transients and sound quality, or will it just increase the speaker's efficiency to the input power?

I own a pair of Strathearn midranges. If I replace the existing magnets with powerful somarium cobalt magnets, or other powerful magnets, will the sound quality improve in any way? The most powerful form of magnet is an electromagnet with the greater current used, or the more ampere-turns of wire used, to increase the magnetic strength. If such a powerful form of this type of magnet is used in place of a permanent magnet to produce a high concentration of magnetic flux, what improvements will occur?

Thomas Perrera
Ozone Park, NY 11420

Mr. Lampton replies:

Mr. Perrera is correct to identify the magnetic field as a prime area for improvement in the efficiency of ribbon loudspeakers. As Henry Primbsch and I pointed out in our article, the factors to be concerned with are conductor mass, total moving mass, area and magnetic field. Adopting a more powerful magnet should not only increase the midband efficiency, but should also extend the amplitude response to higher frequencies.

High-technology magnet materials like somarium cobalt offer two to three times more coercivity and flux density than ordinary ferrites. However, they cost ten to fifty times more than ferrites. So they are best used where a physically tiny magnet is called for, such as in phono cartridges, and where cost is not very significant. If you are trying to maximize the magnetic energy per dollar (rather than the magnetic energy per cubic inch) then you are better off with ferrites. For this reason, we suggested that experimenters build up a structure out of iron plates (as shown in the article) that allows a variety of big block magnets to be installed. Space is not usually critical in loudspeaker structures.

Electromagnets are also a possibility; they allow even more magnet energy to be put into the working gap. The limit is set by the saturation of the pole pieces (iron normally saturates at 18k gauss—very large). One can also combine a permanent magnet and electromagnet into a common structure. I have not experimented with these however.

JBLs IN AN ISOBARIK

I am in the process of building an Isobarik system using JBL 15-inch drivers (#2235H) and would like to have John Cockroft's comments on two questions. First, In selecting V_{BOX} for the inside driver, if I use the Thiele/Small closed box equations what is the relationship between Q_{TS1}, for the inside driver (V_{BOX}) using the conventional calculation Q_{TS2} for both drivers and box?

Second, Cockroft's design uses a $Q_{TS2} > 1$. For closed box, normal one driver systems, $Q_{TS1} = 0.5-0.7$ has worked well. Is this Mr. Cockroft's preference for underdamped systems, or with Isobariks $Q_{TS2} \geq 1$ is "critically" damped ($Q_{TS2} = 0.7$)? I would also like to know Mr. Cockroft's "push-pull" design.

Stephen E. Hluchan
Woodbridge, CT 06525

Mr. Cockroft replies:

When I wrote my Isobarik article I was under the impression that the outer speaker of the pair behaves, at least partially, independently of the inner speaker. However, many SB *readers have stated and shown otherwise. It would seem that the compound pair have approximately the same characteristics as a single speaker, certainly close enough to use in calculating a closed box. In view of this, my Project 2 probably is closer to 0.88 (Q_{TC}) than the 1.13 I stated in my article.*

I have no preference for any particular Q_{TC}. It is a matter of what can be done to provide the best possible environment for a given raw driver. This includes considering a whole Pandora's Box full of possible parameters and adjustments besides just Q_{TC}. It all depends on the parameters of the woofer in relationship to the parameters one desires in the system (or as is usually the case, the best possible compromise).

It is true that closed box systems with $Q_{TC} = 0.5-0.7$ work well. It is also true that closed-box systems with higher Q_{TC}s are also successful. Roy Allison states that his systems have a Q_{TC} of 1.0. When the AR 9s first came out, an AR representative told me they had a Q_{TC} of 0.8.

Incidentally, critical damping occurs at Q_T 0.5 not at 0.7 as you stated in your letter. $Q_T = 0.7$ is the curve with the longest straight line section. Compared to the critically damped curve, which is 6dB down at F_C, $Q_T = 0.7$ is 3dB down. This is because Q_T (actually the figure is 0.707) is underdamped enough to cause the curve to "peak" 3dB above the −6dB position, thus straightening out the curve a little, giving a longer straight line. As this is a function of Q_T only, it would be the same with any kind of system.

Regarding my push-pull system, SB *has my manuscript. I believe it will be published soon.* [Scheduled for 3/86.—Ed]

DALINE DESIGNER WRITES

When an envelope postmarked Puerto Rico, addressed with magnificent calligraphic artistry, dropped on our doormat there was much excitement in our domestic quarters. To the UK, that's a long way. The letter asked if I had any objections to the eventual publication of an article which described grafting an LS3/5A baffle onto the B110 Daline. Objections? Certainly not. It sounded like an excellent idea (why didn't I think of it first?). Carlos Bauza wrote this letter, of course, and I am pleased that he found the Daline worthy of such treatment. (*SB* 4/85)

The B110 Daline is now about 15-years old. The whole thing was born out of two things: frustration and a long-lived dream. The frustration was caused by the inability of compact loudspeaker systems to reproduce the lowest fundamentals of bass instruments. The dream? I suppose I did not think I could have realized it. I simply wanted to create a speaker system that would be compact and able to reproduce 20Hz at high volume. Ah, yes, that is not difficult is it? But I also wanted no bass resonance in the audio range.

I came to the conclusion after studying the then conventional forms of bass loading—sealed boxes, reflex, horn and transmission line—that there seemed to be no possibility of realizing the objectives using any of those systems. That was in 1971 and, to be brutally frank, I do not believe that any other development in speaker system design since then has made sufficient progress to change my mind.

The R & D took about a year and the system displayed some extra benefits—acoustic reinforcement of the bottom octave and a half of the working range, much improved power handling in the deep bass and significantly improved midrange clarity. Whatever else was achieved, just removing the bass resonance would have been enough on its own. I could not understand then, and I cannot understand now, why audiophiles are quite happy about having a resonance right in the middle of the bass in their loudspeakers, but would be mortified to have the slightest indication of anything resembling it in the cartridge, amplifier, CD player or whatever.

Circa 1971, the bass midrange unit that showed the most promise in terms of clarity and definition was the then fairly new B110. The fact that it has endured 16 years or so and still earns respect is a worthy testament to its designer. In the original B110 Daline I used the KEF T15 tweeter and Coles 4001 Supertweeter. The very month the Daline+ B110 article was published in *Hi-Fi News*, KEF announced cessation of production of the T15. This put me under pressure to find an alternative unit for the 3-way Daline, which led to the Peerless K010DT. To my mind it never matched up to the performance of the T15.

Did I meet the original design objectives? Apart from the maximum SPL (sound-pressure level) capability, yes. However, during the last two years or so, I have been able to develop the system still further, primarily for the power speaker market. Incidentally, this is a particularly interesting area of the market for the designer who likes a challenge. We are all familiar with the inadequate sound produced by a lot of power loudspeakers, and also impressed by the hi-fi-like sound of the top quality systems. Getting this sound quality from a power speaker is a lot more difficult than it is from a hi-fi speaker.

The amplification of synthesizers, electronic drums and bass guitar presents some interesting problems. (Did you know the majority of bass guitar speaker systems do not reproduce the fundamentals of the bottom E string of the instrument?) Do you know of a loudspeaker system that is compact and able to move as much air as a bass pedal drum, and without losing the sound character in uncontrollable "boom?" Synthesizers can produce sounds that extend out of the hearing range at both ends. They need speakers that do the same. However, I have not forgotten the home constructors, so it is likely a new "son-of-Daline" may be published in the not too distant future.

Robert Fris
Hampshire, England

Mr. Bauza replies:

It is a pleasure to build and hear the Daline. We have it because of Mr. Fris' generosity in sharing the design published in Hi-Fi News, *and Editor John Atkinson's permission to publish the revised version.*

The slight emphasis of the upper bass is not part of the Daline loading. It results from the intentionally large resistance and inductance values of the LS3/5A crossover, and is sufficiently mild to be a builder's choice. In any case, the Daline+ B110 is a true full-range design, which is very gratifying in the power context of a miniature speaker.

"AND RE-VISITED" REVISITED

Mr. D'Appolito's letters in *SB* 4/85 were excellent, but he made one statement without much explanation. In his letter "And Re-Visited" he stated, "these results [same f_S, Q_M, Q_E and reduced N_O] are true regardless of the voice-coil interconnection, which may be series or parallel." Assuming both drivers in the composite are identical they will have the same flux density, B, and equal lengths of wire in the magnetic field. When placing the voice-coils in series, B is constant, L doubles and R_E doubles causing $(BL)^2/R_E$ to double. So far, so good.

But when the voice-coils are placed in parallel, B is still constant and L (the length of the wire in the magnetic field) doubles, but R_E is one-half of its value for a single driver. This would imply $(BL)^2/R_E$ for the

SB *Mailbox*

composite driver would be eight times that of a single driver. Thus:

$$Q_E' = Q_E/4$$

and

$$N_O' = 2N_O$$

using your notation.

My only assumption has been that the source is a constant voltage source with a small internal resistance compared with R_E.

Charles Smith
Champaign, IL 61821

Mr. D'Appolito replies:

Q_E is a measure of the electrical self-damping of a loudspeaker driver. This damping is caused by the back voltage generated by the voice-coil velocity, which in turn causes a damping current to flow that is limited only by the voice-coil resistance R_E by Mr. Smith's assumptions. When a driver is analyzed from its electrical side, you must look into its terminals to find both its equivalent internal resistance and voltage source causing the damping current.

For a single driver the internal voltage source is BL_V, where V is the cone velocity. The internal resistance is, of course, R_E. This equivalent circuit is shown in Fig. 1a, *with two voice-coils in series, both internal sources and resistances are in series as shown in* Fig. 1b. *The equivalent circuit, thus, has an internal voltage generator of $2BL_V$ and an internal resistance of $2R_E$ so that $(BL)^2/R_E$ doubles.*

The situation is quite different when the voice- coils are parallelled as shown in Fig. 1c. *The separate internal generators no longer add. The equivalent voltage source is just BL_V, but the internal resistance has dropped to $R_E/2$, which causes the motor factor to double. The equivalent BL product does not double in the parallel connection due to the way the two internal voltage sources interact. The net result of all this discussion is that Q_E is the same for either voice-coil connection and the same as that of a single driver.*

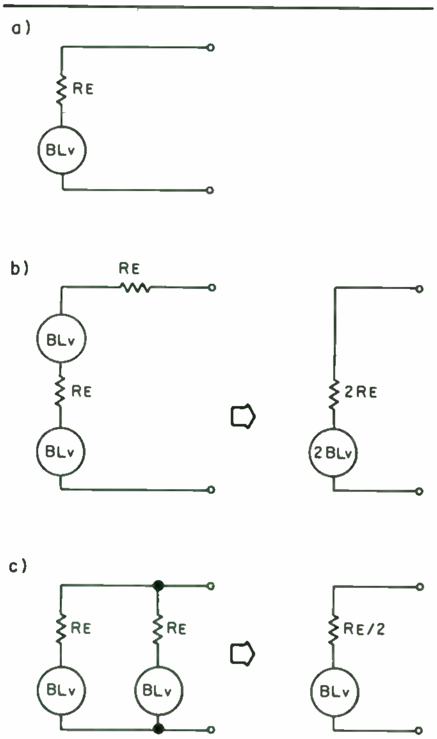


FIGURE 1: The equivalent circuit (a), with two voice-coils in series (b) and the paralleled voice-coils (c).

SATELLITES UNDER SCRUTINY

Mr. D'Appolito states in his satellite article (*SB* 4/85, p. 7), that the best type of crossover to use in the three driver two-way design, for the desired spherical radiation pattern, is the odd-order Butterworth 18dB/octave, which would cause a 90 percent phase shift between the drivers. This means the sound from one driver, will arrive sooner than the others on axis.

To correct this error one of the drivers must be arranged ahead of the other on the baffle so both the woofers and the tweeter will be acoustically in phase with each other on axis, which is also known as Time Alignment® Why doesn't Mr. D'Appolito do this in his design? Does this phase shift not cause distortion or some type of unnatural presentation of program material? How much distortion does the Butterworth crossover produce as compared to a first-order crossover, which is known as "transient perfect?" I have heard there is some ringing on transients, is this very noticeable?

I would like to build a three-way loudspeaker using the D'Appolito configuration in the following manner: one or two modified Panasonic Leaf tweeters flanked on both sides by two or three 50mm Jordan Modules, also flanked by two Dynaudio 8-inch woofers, one on top and the other at the bottom with the same driver arrangement as the satellites.

The Jordan Modules are small 4-inch midrange drivers, the Panasonic Leaf tweeter is also small, so the two outer Dynaudio woofers will not be too far from each other in this arrangement. I also intend to tri-amp using the odd-order Butterworth 18dB/octave slope as suggested. Each driver's sound pressure level can be adjusted by using an active crossover and eliminating the need for equal sensitivity. Will I still be able to produce the desirable spherical radiation pattern in this arrangement? Would an odd-order 36dB/octave crossover produce this spherical pattern?

My last question is that Mr. D'Appolito states he wanted his satellites to produce a SPL of at least 110dB, why would he want such a loud playing unit for home use?

Thomas Perrera
Ozone Park, NY 11420

Mr. D'Appolito replies:

Time Aligned® phase-coherent loudspeakers are a Holy Grail which is rarely, if ever, achieved in practice. Most loudspeaker systems displaying flat frequency response are over all non-minimum phase, and thus have phase distortion. This non-minimum phase response is caused by the summing action of the crossover network.

No definitive evidence exists that this type of phase distortion is audible. (This includes the so-called "ringing" that many underground press reviewers love to conjure up.) Quite the contrary, all published experiments of which I am aware indicate that the all-pass phase response of these systems cannot be detected on program material. (See the early Linkwitz articles in SB 1980 for more detail on this point.)

With regard to your first question, a 90° fixed interdriver phase shift does not imply that the drivers are out of alignment with reference to time. In building a loudspeaker system, the drivers are first Time Aligned® and then the amplitude and phase characteristics of the crossover network are imposed upon the individual drivers to obtain the over all desired frequency response from the combination. All odd-order Butterworth crossovers (6, 18, 30 etc. dB/oct.) produce a 90° interdriver phase shift. This includes the first order, "transients perfect" crossover.

Once the drivers are aligned with respect to time, the constant 90° phase shift between the drivers coupled with the attenuation of 6dB/octave is just what is required to have the individual driver responses add to the transient-perfect response. The wavefronts from the drivers do not arrive at the same time, they arrive in the proper Time Alignment® to add up to the original input signal. Although drivers aligned thus are acoustically in phase, the crossover network puts the drivers out of phase in just the correct proportion to add up to the transient-perfect response.

Due to the limited bandwidth of most drivers, it is almost impossible to achieve the first-order transient perfect response. Since my satellites should have an interdriver phase shift of 90°, I had to go to the next higher odd-order Butterworth crossover which is 18dB/octave. More justification for my crossover choice is well covered in my original article.

The vertical dimension of the system you propose spans more than one wavelength at frequencies above 700Hz, and thus looks like a line source with a cylindrical radiation pattern above this frequency. Many authors consider this a desirable pattern. Due to driver placement symmetry, the pattern will be *stable, i.e. no lobing error. The absence of lobing error is more important in my opinion than the spherical versus cylindrical wavefront issue.*

A 36dB/octave crossover is even-order, not odd. The drivers will be 180° out of phase at the crossover frequency producing a large response null. Due to the large vertical dimension of your proposed system at 12kHz, even-order in-phase crossovers will produce multiple off-axis nulls in the frequency region around crossover. I suggest that you try odd-order Butterworths.

Many sources of live music produce short term SPLs of 110dB, and most audiophiles are interested in producing these levels realistically. Even if you prefer lower levels, however, the stable radiation pattern of my system coupled with the lower distortion levels achieved due to reduced driver excursion required when multiple drivers are used is, I believe, sufficient justification for the design.

BUGS IN MY STRATHEARNS

I was so excited by the Strathearn ribbon article in *SB* 3/85, p. 22, I ordered a pair of them despite having never seen or heard them. Mr. Spangler's and Mr. McKenzie's article is lucid and informative and I am grateful for the benefit from their research. However, I see a couple of small flies in the ointment. I hope the authors can swat them for me.

The disclaimer in the *Author's Note* at the end of the article filled me with trepidation. When I asked the folks at Audio Concepts about quality control and DC resistance problems with the Strathearns, they told me that none of the units they had received in some time were up to 0.5Ω, and the best they could do was to match units for resistance before shipping me a pair. I figured that some Strathearns would sound better than no Strathearns at all, so I bought a pair in hopes of using the modification even if it meant adjusting the LC values to optimize the prefilter.

When the units arrived, I measured their DC resistance (0.28–0.30Ω), and AC impedance for the driver/transformer combination (approximately 6Ω using the "middle" tap). I then attempted to measure their (more-or-less) near-field response. Not having a computer, let alone an IQS 401, I improvised with ⅓ octave pink noise bands from a Denon compact disk. The measuring apparatus consisted of an electret condenser microphone borrowed from my Audio Control C101 to feed a venerable GR 1554-A analyzer, reading in all-pass mode. After correcting for the 1554's response, I received the response curve in *Fig. 1*, which indicates a tremendous peak centered at 13kHz, a wider trough at about 3kHz and a smaller peak at about 1.5kHz. Varying the microphone's position caused some changes in the response, but the basic pattern is pretty much the same.

I hope Spangler's and McKenzie's modifications will work on these beasts if

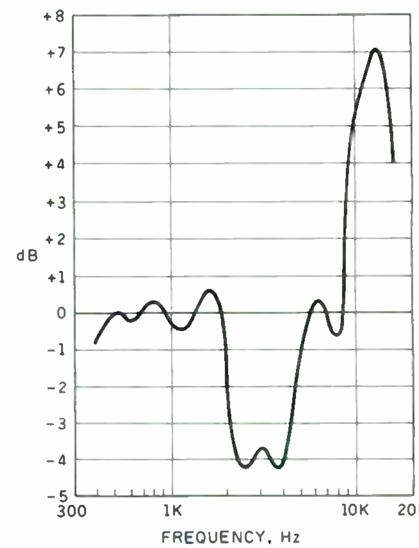


FIGURE 1: The 1554's response curve after corrections.

SB Mailbox

I play with the values in the prefilter until I get the high end right. Before I void the warranty by tearing into them, I would like to know whether the modifications will work and whether the units may be defective. They will play loudly, but not very loudly.

Through an 18dB active 300Hz crossover, they will play loudly enough to make spoken conversation difficult in the room, but will begin to crackle and distort badly with the cleanest signal at volume levels far below the threshold of pain. Is this normal? Am I asking too much from 0.000080-inch thick plastic-wrap diaphragms? This is not a problem for me, just a matter of concern before I begin to rip, tear and glue.

Jack W. Brent
Seattle, WA 98102

Mr. Spangler replies:

I have been curious whether Strathearn was back in business and producing good units. As for Mr. Brent's loudness problem, from my experience his units are doing fine. The single panels, which are preferred for smooth response, do not produce levels capable of splitting eardrums. I have not found it necessary to have any higher sound levels with the sources I listen to, however, and I like loud music.*

One must use care when playing the Strathearns loudly, and I was surprised that Mr. Brent has not melted a trace, if he heard crackling. I would suggest Mr. Brent use the outside taps on the transformer since they measure better when passing square waves.

Mr. McKenzie replies:

In addition to Mr. Spangler's comments, the crackling you heard could be caused by the diaphragms excursing so far as to touch or slap against the magnet structure. This would represent the maximum design sound pressure level achievable if the diaphragm is properly centered. To check for proper centering, remove the rear fiberglass panels and peer into the magnet gaps with a flashlight to check for equal clearance on the front and back side of the driver. Also check for equal spacing at the center of the magnet structure and at the two ends of the magnet gaps. If the spacing is equal to naked eye tolerances, then the driver is not defective. It is a good idea to check each unit for proper spacing and clearance before you modify them, and remember how the spacing should look after you have installed the foam pieces.

The lower than specified DC resistance is not a problem to properly load and operate the prefilter. The prefilter component changes will not be needed to give good and effective results. We mentioned some of the drivers' low resistance, not because the low resistance was a problem, but because those low resistance drivers also had frequency/resonant problems which were not changed by our modifications.

The data Mr. Brent generated is very close to the expected response of a "good" Strathearn. I do not expect the prefilter network will need trimming, but he seems to have the right equipment to do so if he wishes.

*Note: The Strathearn Company's assets and patents have been purchased by a West German firm, hifi-sound, Lautsprecher Vertrieb, Saerbreck & Moravia, Judefelderstrasse 35, 4400 Munster. They will market the driver under the name Stratek SLC II MRHF. List price is DM698. Presumably these will still be available from current U.S. distributors.—Ed.

EXAMINING THE ESL

Over the past several years, I have been doing considerable research on the subject of electrostatic loudspeakers (ESL). While I do not consider myself an expert in the field of acoustics, perhaps you will permit me to comment on Mr. James Rice's concerns in *SB* 4/85, p. 58.

His first question concerns the round versus the square ESL. Although not too many round ESL drivers have been built, the two most notable are the Voght and the Malme units. Information about the Voght unit, designed in Germany, can be found in *Wireless World*, May 29, 1929. The capacity of the Voght unit, which seems to be one of several, was 1kpF and the polarizing voltage was between 500–700V. Based on these details the spacing between the diaphragm and the plates should be approximately 0.02-inches.

A more up-to-date ESL was designed and built by Charles Malme, a research assistant at the Massachusetts Institute of Technology. The results of his efforts were reported in the *Journal* of the Audio Engineering Society, January 1959 (Vol. 7, No.1), and patented on December 19, 1961 (#3,014,098).

Malme's most interesting design features ¾-inch diaphragm-to-plate spacing. This is probably the largest spacing ever used and it created some special requirements for the amplifier. The driver, 20-inches in diameter, had a capacity of 270pF. With the amplifier directly coupled to the speaker (no transformer) it delivered a voltage output of 4.5kV RMS (root mean square). It necessitated the use of special high-voltage tubes.

Mr. Rice also expresses concern about the performance of a round versus a square radiator. If two speakers (a round and a square) are of equal size with the same mass and diaphragm tension, then the configuration, as a radiator, should not make a difference. One way to determine whether this concept is valid is to calculate the resonant frequency for each type of driver.

The formula for the resonant frequency of a circular diaphragm is:

$$Fr = (0.382/r) \times (T/M)^{1/2} \quad (1)$$

where:

Fr = resonant frequency in hertz
r = the diaphragm radius in centimeters
T = tension in dynes
M = mass in grams.

As an example, suppose a 20-inch circular diaphragm has a mass of one gram. If the diaphragm tension is 10 to the sixth dynes than the resonant frequency is:

$$Fr = (0.382/25.4) \times (10^6)^{1/2} = 15\text{Hz}.$$

For a square speaker to provide the same results it must have an equal area. The area of the round speaker mentioned above is 314 square inches. A square speaker of equal area will be 17.72-inches on each side. The formula for calculating the resonant frequency of the square diaphragm is:

$$Fr = (0.705/a) \times (T/M)^{1/2}$$

where:

a = one side of the unit in centimeters

With the same tension and mass, as the round ESL, the square diaphragm's resonant frequency is:

$$Fr = (0.705/45) \times (10^6)^{1/2} = 15.7\text{Hz}$$

Due to the two results being almost identical, the calculations indicate the shape of the driver is not important. As further evidence of this, Olson states in his book on electrical engineering, "in the study of sound sources the sound waves depend upon the amount of air that is moved and not on the shape of the radiator."

I do not know of any specific investigation which covers Mr. Rice's question about the relationship between a speaker's frequency response and the diaphragm spacing. Drivers that are intended for bass reproduction generally have a large plate to diaphragm spacing. The KLH-9 for instance, had a spacing of 0.012-inches for the tweeter and 0.100-inches for the bass panels.

I have not found literature to indicate that spacing directly alters the frequency response. It does affect the bass region sound pressure, as well as the efficiency, capacity and amplifier loading. Some of this was reported by R.J. Matthys in his article, "Telstar Shaped Electrostatic Speaker," *Audio*, May 1961.

Mr. Rice also raised the question of ESL distortion. If the speaker uses the push-pull technique this should not be a significant problem. The distoriton level of most and perhaps all ESLs of this type is under one percent. I have measured the distortion from my own ESLs, using a B & K condenser microphone and a Hewlett-Packard selective level voltmeter, and the results show a distortion level which is about the same as the source.

In this case the value was 0.03 percent for both the second and third harmonic. Perhaps if I had used another oscillator, which had lower harmonic distortion, I may have been able to separate the speaker's distortion from the generator's. However, since the measured value was far below any detectable audio level I did not believe it necessary to pursue this any further.

In reference to Mr. Rice's concerns about beaming versus driver size; all drivers have a tendency to beam as the frequency is increased. The actual point where beaming occurs is a function of the speaker's diameter and the highest frequency that it must radiate. Most books on acoustics will show a directivity curve as a function of a parameter called "kr." The value of "r" is equal to the radius of the speaker while "k" is equal to the wavelength (speed of sound/frequency). A plot for different values of kr is shown in *Fig. 1.* The critical point is the one at which kr = 2. Above this point the speaker is no longer mass controlled, and more energy is fed to the radiation resistance.

Commercial manufacturers use different techniques to extend the performance of their drivers. One method is to reduce the size of the cone area. This is usually a mechanical function that is designed into the driver. The second method is similar except it uses a separate driver, with a suitable crossover network, to maintain an acceptable kr value.

Malme's design reduced the radiating area of his ESL by a factor of two for each doubling of the frequency. Other designers have maintained the width at a value equal to or less than 1.3 of the speaker's uppermost frequency wavelength.

While the above explanation is related to the speaker, it is not the only factor which affects directionality. In Martin Colloms' book, *High Performance Loudspeakers,*" he indicates on pages 11–15 that directionality is also a function of the type of baffle. Another factor which must be considered is the measuring distance. Close to the speaker the test results can show extreme directionality. However, sound spreads out as it disperses from the source, so the directionality may be much less in the far field.

The last two items I would like to comment on for Mr. Rice are the use of passive crossovers and the concept of curved speaker. Most commercial ESL speakers use passive crossovers (see the article on the Quad Model 63 in *SB* 1/82, p.10). The Malme article also mentions that he used resistors as a crossover to the various sections of his loudspeaker.

As far as a curved unit is concerned, although I am not familiar with John Civitello's speaker, the Pickering Company manufactured a speaker called the Isophase during the late 1950s. This speaker never seemed to succeed commercially; perhaps due to its vertical directionality. When you were seated the unit sounded fine, but when you stood up the output dropped considerably. The curved concept, however, has been used successfully by many horn designers as well as those designing tweeter arrays. Usually these devices are curved so the sound radiation is projected both horizontally and vertically. A tall ESL, which uses a planar array does not need vertical curvature. You can achieve horizontal dispersion by using more than one driver panel.

Some of the above information is taken from my book on ESLs, to be published later this year by TAB Books. In addition to the theory of how an ESL works, the book provides information on how to build an ESL as well as where to obtain some of the materials.

Continued on page 52

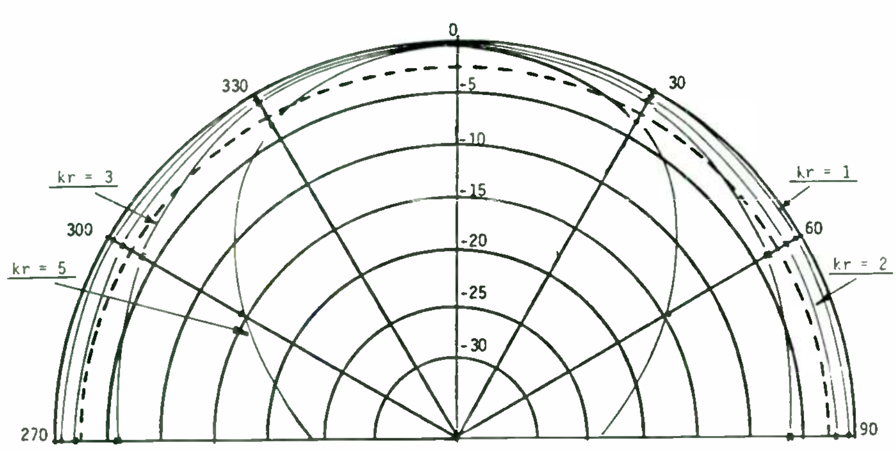


FIGURE 1: A plot for different kr values.

SB Mailbox

Continued from page 51

[John Civitello's curved electrostatic to be announced at the summer CES in June is covered by U.S. patent #4,289,936, issued Sept. 15, 1981. The driver consists of triangular elements making up a geodesically curved surface.—Ed.]

Robert Wagner
Fremont, CA 94539

REFERENCE

1. Gayford, M.L., *Acoustical Techniques*, MacDonald and Evans Ltd., 1961, pp. 50–51.

BASS MUFFLERS

I was wondering whether any readers have ever experimented with coupling a bass driver to an auto exhaust muffler(s)? Any results or behavior predictions would be appreciated.

James Y. Pann
5818 Santa Cruz Ave.
Richmond, CA 94804

Why don't you build one and take it out for a test spin.—Ed

TWIN DRIVERS

I originally thought that Thiele/Small-aligned cabinets were far too complicated for my precalculus mind. After some heavy number-crunching and a very good calculator purchase, however, I have found the alignments to be relatively easy.

One item I have failed to find is a simple formula for Thiele/Small alignments using two identical drivers in the same cabinet space. The current rage in some circles is to use twin drivers, usually 6½ inches with high power-handling capability, with one crossed over at about 350 to 500Hz. Designs based on this concept by B&W, Polk and others seem to sound very good.

I would like to be in contact with anyone who has information about this interesting concept.

Robert Conner
Omaha, NE 68131-2404

Robert M. Bullock replies:

You can achieve the Thiele/Small alignment of a multidriver woofer system by replacing the array with a single equivalent driver. This means that the model predicted response will be valid only at those frequencies low enough for you to treat the array as a single, simple source. If you plan to cross this system over at a high frequency (i.e., to a tweeter), you must treat it as a multiple source and deal with the complications that entails. To avoid this problem, I suggest crossing the system over at no more than a few hundred hertz.

Mr. Conner's idea of crossing over all but one driver at a low frequency alleviates the problem of multiple sources, but it also creates a sensitivity matching problem. A commercial manufacturer is likely to use a comprehensive trial-and-error design strategy to deal effectively with either of these problems, but the home builder usually must settle for the more conservative strategy of problem avoidance.

The most direct method for determining the Thiele/Small parameters of the equivalent driver is to connect the component drivers in the arrangement you plan to use and treat the resulting array as a single driver accessible only at the input terminal pair. Then measure its parameters as described in my article in SB 1/81 (p. 12). This technique is not always practical, however, especially when you are trying to measure V_{AS}, and it does not allow you to make preliminary pencil-and-paper feasibility studies based on a manufacturer's driver specifications. You must be able to infer the equivalent driver parameters directly from those of the array elements.

When all the drivers in the array are identical, this is quite easy to do. Regardless of how the component drivers are electrically connected and how many there are, the equivalent driver resonant frequency, electrical Q and mechanical Q are the same as those of one array element. If the array contains N identical drivers, the volume equivalent acoustic compliance of the equivalent driver is N times that of one element. Thus, you can use the same target alignment for the N-driver system as you would for one component driver and simply increase the box volume by a factor of N.

Using identical drivers in a multidriver woofer not only simplifies system alignment, but also produces the best performance in two and four-driver systems, as the work of Greiner and Allie shows ("Acoustical and Electrical Interaction in Multi-driver Arrays," Preprint No. 1818 (B-6), *presented at the 70th Convention of the AES, October 30 to November 2, 1981). In practice, "identical drivers" probably means all of the same make, model and size. Parameter values will inevitably vary from sample to sample, so it is best to use their average value in determining the corresponding parameter of the equivalent driver.*

If you want to equalize the impedance of the equivalent driver, you must know its voice-coil resistance and inductance. These depend on how the component drivers are connected and follow the usual series/parallel rules for combining resistances and inductances. With two drivers in parallel, the equivalent resistance and inductance are half those of one driver. If the drivers are in series, the equivalents are twice those of one driver. If four drivers are used, the factor is four for series connection, one for series-parallel or parallel-series, and one-fourth for parallel.

The sensitivity of the equivalent driver also depends on the electrical connection. For a series connection of two drivers, the equivalent sensitivity is the same as for one driver, while it is 6dB more for a parallel connection. With four drivers, series connection leaves sensitivity unchanged, series-parallel or parallel-series increases it by 6dB, and parallel increases it by 12dB.

If you are interested in studying the large-signal response with BOXRESPONSE, an Apple program of the Thiele/Small models (SB 1/84, p. 13), you should know that the equivalent thermally limited input power and cone area are additive, and the linear excursion limit is the same as for one driver, regardless of the electrical connection. For example, two drivers of 100W input power, 5mm linear excursion and 500cm² cone area are equivalent to a single driver of 200W input power, 5mm linear excursion and 1,000cm² cone area. It follows from the cone area relationship that the equivalent diameter for two identical drivers is $\sqrt{2}$ times the diameter of one driver, while for four drivers, the factor is 2.

HP PROGRAM FOR SIXTH-ORDER T/S ALIGNMENT

My program for the HP-41C will provide the design information you need to build a filter-assisted, vented-box speaker, given the Thiele/Small parameters for a particular driver. It calculates each of the three classes in the Chebyshev to Butterworth to sub-Chebyshev line of sixth-order alignments. To use the program, your HP-41C must have three memory modules and a printer (or you may use a printer with an HP-41CV).

The program asks for the following parameters:

f_{AS}—the driver resonance frequency in hertz;
Q_{TS}—the driver total Q;
V_{AS}—the equivalent volume of the driver's compliance in liters;
Q_E—the driver's electrical Q;
S_d—the area of the diaphragm in square meters;
X_{max}—the maximum displacement of the diaphragm in millimeters;
Q_B—the total Q of the box.

The calculator then calculates the driver's reference efficiency, as well as the following:

K—the Chebyshev constant;

R—the amount of ripple (if any) in the pass band;

V_B—the box volume, in cubic inches and cubic feet;

f_B—the frequency to which the box should be tuned;

D_v—the minimum diameter of the vent to prevent wind noise;

f_{aux}—the frequency to which the auxiliary filter should be tuned;

Q_{aux}—the Q of the needed auxiliary filter;

a_{aux}—the damping of the filter (1/Q);

f_{pk}—the frequency of the peak in the auxiliary filter response (if any);

M_{pk}—the magnitude of the peak (if any);

P_{AR}—the estimate of the acoustic power output available;

f_3—the frequency at which the composite response of the filter and speaker is 3dB down; the cutoff frequency.

The program should be available soon from the Hewlett-Packard Users Library on magnetic cards or bar code with further documentation. It is derived from the work of R.A.R. Bywater and H.J. Wiebell [*JAES* (Volume 30, Number 5), May 1982, pp. 306–317] and J.E. Benson [*AWA Tech Review* (Volume 14, Number 4), 1972, pp. 369–484].

For details about the program, write to me at the address below.

Dan T. McGillicuddy
34 Shears St.
Wrentham, MA 02093

STIFFER, LIGHTER ENCLOSURES

I enjoyed Siegfried Linkwitz's "Excursion-Limited SPL Nomographs" (*SB* 4/84, p. 24) very much. In the last paragraph of his article, Mr. Linkwitz touches on the problem of cabinet resonances excited by high SPLs. He suggests the use of very stiff panels or extremely well-damped boxes.

Many speaker builders use ¾-inch high-density particle board for speaker cabinets and apply heavy coats of damping material to the interior. While this works, I think it is a "brute force" solution. A 2-cubic-foot cabinet built as above is almost impossible to move without a forklift.

For reasons I do not understand, I know of no one who is making any serious attempt to make enclosures stiffer and lighter. I know from my own experience as a cabinetmaker that it is possible to construct a corrugated panel out of ½ or ⅜-inch plywood that is many times stiffer than ¾-inch particle board (for the same area) and is a fraction of the weight. Assuming that I built an ultralight, ultrastiff speaker enclosure, would there necessarily be any adverse sonic effects from such a box?

Kenneth P. Miller
Mexico, MO 65265

Mr. Linkwitz replies:

Mr. Miller raises some important questions about the construction of cabinets with reduced panel resonances. I have done some experimentation in this area, and here are my findings.

Initially, I built some LS3/5A size boxes out of ¾-inch mahogany. (See *SB* 4/81, p. 32, for a report on the LS3/5A.—Ed.) *The side panels showed resonances at about 430Hz with a rather high mechanical Q of around 40. A 430Hz tone took 180msec to decay by 60dB after I shut off the electrical signal. This is a slow decay. I then applied thick layers of roof-patching tar thickened with sand to the inside of the cabinet. This lowered the resonance frequency to 300Hz because of the increased mass of the panels. The decay time decreased to 60msec, corresponding to a Q of 8.4. The whole approach was rather messy and the result not completely satisfying because a Q of 8 is still high, at least to an electrical engineer. The problem was that the tar's stiff-*

SB Mailbox

ness was too low to restrain the vibration of the wood panel.

You can achieve a better match in stiffnesses by using ¼-inch plywood panels instead of the thicker particle board. Without the damping layers, such a box is unacceptably transparent, but with a ½-inch coat of tar and sand on the inside, the enclosure is acoustically quite dead. The damping layer dominates the motional behavior of the walls, while the plywood provides the outer skin. Again, the construction of such a box is messy. Getting the thick tar layer to adhere to the walls can be a problem, particularly when the temperature rises. Also, the effectiveness of the damping is temperature dependent. For these reasons, I have gone to a different enclosure design.

The idea is to raise the panel resonance frequencies by making the enclosure extremely stiff so that the resonances occur either above the frequency range of the drivers or fall into a frequency range where it is easy to decouple the driver from the enclosure. For the enclosure mentioned above, with a midrange-to-tweeter crossover at 1,600Hz, the resonances are pushed to 1,800Hz and above. They are, therefore, not excited by the midrange driver. The tweeter does not transmit significant vibrational energy to the enclosure because the kinetic energy decreases with the square of frequency for constant SPL from the driver.

To achieve the desired high-resonance frequencies using ¾-inch particle board, I use the following rule of thumb based on actual measurements: keep all unsupported panel areas to less than 20 square inches. I use ¾-inch particle board walls for internal bracing of the enclosure, forming separate internal chambers with openings between them. This makes for a very stiff enclosure. Knocking on it with my knuckles makes a pinging sound like a brick. A better test is to apply shaped tone bursts and observe the panel motion with an accelerometer.

I would imagine that you could push the resonances even higher using a plywood construction, as Mr. Miller suggests. Using at least ½-inch plywood would not make a light box because of the internal walls. Since I have not worked with plywood, I cannot advise how small the unsupported panel areas must be.

I have one final but very important point. When you mount a driver by its basket in the standard fashion of screwing it to the front panel, it will exhibit a nasty resonance. This resonance is caused by the springy-ness of the basket and the mass of the attached magnet. With the basket clamped to the panel, the spring-mass system is free to resonate. The resonance occurs in the 150 to 250Hz region for typical woofers and midranges. The Q is extremely high, and the system rings like a bell when excited. Further, the mechanical impedance is so high that you cannot dampen the ringing even when holding the magnet firmly with your hand. You can avoid the resonance by mounting the driver from the magnet and having only a compliant air seal between the basket and the front panel. The magnet should rest in a half-circle support structure, which is part of the internal bracing, and be firmly strapped to it.

This has turned out to be a rather lengthy reply, but I hope it provides some useful information. After all, hobbyists can use some construction techniques that are unfeasible commercially.

REAR-LOADED HORN CABINETS

I am interested in designing and building my own rear-loaded horn cabinets using either a 12 or 15-inch driver similar to the JBL 4530. I don't see many rear-loaded bass cabinets around. Is there some drawback to this design concerning performance? Can anyone recommend available books or information explaining the criteria involved in determining proper dimensions for rear panels, angles, and so on? I would appreciate any help.

Gilles R. Grignon
Hogansburg, NY 13655

Bruce Edgar replies:

The rear-horn-loaded enclosure, such as the JBL 4530, is an attempt to get around the bandwidth restrictions of front-horn-loaded enclosures. With a front-loaded bass horn, you are lucky to get response up to 400Hz, but with a rear-loaded cabinet, the front of the driver is radiating into free space and can provide response up to several kilohertz. As the theory goes, the sound coming off the back of the driver propagates through the horn to produce the bass.

The reason you do not see many rear-loaded enclosures on the market (JBL no longer lists the 4530 in its current professional catalog) is that the design has some drawbacks. First, at some frequency in the upper bass, some sound cancellation will occur because the sound coming out the bass horn will be 180 degrees out of phase with the sound wave being radiated from the front of the driver. Generally, you can live with that problem. A second and more severe criticism is that to match the output from the front radiator with the horn output, you must shorten the horn. And by shortening the horn, you introduce some horrible resonant peaks into the bass response.

The JBL design does attempt to deal with these problems by using one or two 15-inch drivers to provide a large front radiator surface area and a large mouth on the rear-loaded horn. I would recommend using the JBL drivers (2205, E-140) in the enclosure because they have the large magnet size and low-mass cones required to drive the horn. A typical 12 or 15-inch driver designed for a box will produce poor results.

You can find the plans for the JBL 4520 (two 15-inch drivers) in Electronics World *(January 1962, p. 40) as a sidelight to an ar-*

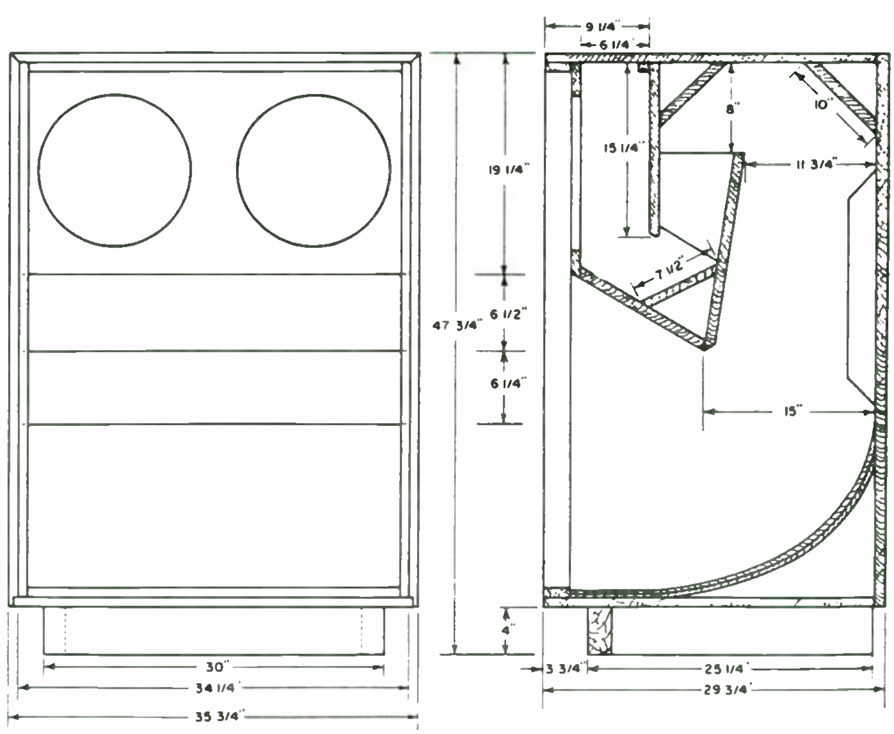


FIGURE 1: Plans for the JBL 4520 enclosure (two 15-inch drivers). From a rough scaling of these drawings, Mr. Edgar has determined that the rear horn is 70.5 inches long with a flare rate of approximately 26Hz. Reprinted, with permission, from *Electronics World*. Copyright © 1962 Ziff-Davis Publications. All rights reserved.

ticle by George Augspurger. From a rough scaling of the drawings (Fig. 1), I have determined that the rear horn is 70.5 inches long with a flare rate of approximately 26Hz. When placed against a wall, the 4520 enclosure has a mouth cutoff frequency of approximately 60Hz. In an old catalog, JBL states that the 4520 has a maximum loading down to 50Hz and is usable to 30Hz. These figures agree somewhat with my calculated response parameters. The 4530 is only half the width of the 4520 and has only half the mouth area, so I suggest using the 4530 in a corner to obtain the best bass response.

I have not heard the JBL rear-horn-loaded enclosures, but the dynamic range and efficiency ought to be very good. I can only suggest that you try building an enclosure and report to SB how the experiment fares.

BOX DESIGN ROADBLOCKS

I have read Bob Bullock's articles in *SB*, especially those regarding box design, with great interest. In trying to comprehend the physical relationship among several of the design parameters, however, I have come to several logic roadblocks. I could use some clarification on these items to alleviate my confusion.

The first area of confusion is compliance (V_{AS}) versus box volume (V_B). Both Small alignment tables and the formula from David Weems' book *(Designing, Building and Testing Your Own Speaker System, 1st Edition)* indicate a direct relationship between these two parameters, but my logic (which has been known to be wrong on occasion) indicates the relationship should be indirect. For example, for a given driver, increasing the compliance also increases the damping. This should require a smaller box volume with its increased air stiffness to compensate for the increased damping and maintain optimum control over the driver diaphragm. Where am I missing the boat in this cause-and-effect description? Is my understanding of compliance wrong? Doesn't high compliance always mean high damping?

My second problem is with the computation of V_B. According to the information in Mr. Bullock's article in *SB* 4/80 (p. 7), V_B equals V_{AS}/α, where α is defined in the Small alignment tables for various values of Q_{TS}. This formula produces results that are significantly less than those achieved when using the formula from the Weems book, where V_B equals $15(Q_{TS})^{2.87}$ (V_{AS}).

For example, for the Dynaudio 21W5406, Q_{TS} equals 0.29 and V_{AS} equals 82 liters. Using Mr. Bullock's formula and the Small alignment table for $Q_L = 10$, for a Q_{TS} of 0.29, α equals 3.1843. Therefore, V_B equals 82 liters/3.1843, which equals 25.75 liters. Using the Weems method, V_B equals $15(0.29)^{2.87}$ (82 liters), which equals 35.24 liters. Since the h values from the Small alignment tables correlate exactly to the Weems tuning factor chart, I assume that both methods are based on the same mathematical concepts. Bob White used the Weems formula in the follow-up article to BOXRESPONSE in *SB* 1/85 (p. 28). Where am I going wrong here? Am I using the Small alignment tables incorrectly?

Finally, when determining a driver's V_{AS} using the test procedures described in Mr. Bullock's article in *SB* 1/81 (p. 12), should the volume of air trapped under the inverted driver cone in the test setup be included with the known test box volume (V_{TB}), or is it inconsequential?

Why is the formula

$$V_{AS} = V_{TB}\ [(f_C Q_{EC})/(f_S Q_{ES}) - 1]$$

better than

$$V_{AS} = V_{TB}\ [1.15(f_{cc}/f_S)^2 - 1]$$

as stated in the same article?

Robert E. Davis
Belle Mead, NJ 08502

Mr. Bullock replies:

Mr. Davis' analysis of the relationship $V_{AS}/V_B = \alpha$ is not incorrect, but it misses the point because he treats α as a constant. With this constraint, an increase in V_{AS} must be adjusted for by a decrease in V_B—i.e., an indirect relationship. But in practice, α is treated as a variable and V_{AS} as a constant. The objective is to choose V_B so that the value of α is the one specified by the target alignment. In this context, higher damping (lower Q_{TS}) is accompanied by larger α values—i.e., higher damping requires smaller box size. You can see this in the relationship between Q_{TS} and *α in the alignment tables and formulas.*

High compliance in and of itself does not necessarily imply high damping. Driver damping depends on several driver parameters. For example, doubling V_{AS} with no other changes will decrease Q_{TS} by a factor of $\sqrt{2}$, but doubling both V_{AS} and M_{MS} will leave Q_{TS} unchanged.

I mention the formulas found in Weems's book in my SB 4/80 article (last paragraph of Small Alignments section, p. 13), but I attribute them to Hoge. Saffran also presented them in a letter in SB 1/81 (p. 34). Alignment formulas for vented boxes are necessarily approximations to the table values, since there is really no elementary *relationship between the alignment parameters. The usual procedure is to take table data and fit an elementary function to it by least squares. The accuracy of the fit depends on how many data points are used and what type of function is fitted. Using too few data points or an inappropriate function can lead to large errors. I presume this is the cause of at least part of the discrepancy Mr. Davis found. Part of it could also be due to the formulas assuming a different Q_L value than Mr. Davis used.*

What you must keep in mind is that formula values are not "incorrect" in any sense; they merely produce a different alignment than the tables. To decide whether it is a good alignment, I would run it through BOXRESPONSE (SB 1/84, p. 13) and look at the response curve. Generally, I think that the table alignments should produce flatter responses, but the differences may be negligible even with a sizable difference in box volume. The excess box volume in the test box with the drivers inverted is probably insignificant, but it would not hurt to include it.

The first formula Mr. Davis gives for computing V_{AS} is exact, while the second is based on an estimate. When you mount a driver in a box, the effective moving mass increases because of an additional air load. The amount of mass increase depends on what proportion of the baffle area the driver occupies. The first formula incorporates the actual mass increase into the calculation, but the second formula assumes an "average driver" that occupies about one-third of the baffle area. Cobb gives a derivation of this approximate formula in the JAES *(Volume 19, Number 1, pp. 53–55).*

PHASE INFORMATION

I ran across Siegfried Linkwitz's reply to John Kasowicz's inquiry in *SB* 1/84 (p. 37) for the second time. But this time, because of my recent experiments, it had a special significance for me.

I am referring to the symmetrical driver configuration (W-T-W) and its ability to exhibit a stable, on-axis direction of the radiation pattern's central lobe, regardless of interdriver phase differences. Mr. Linkwitz tested for audible effects of different crossover networks. I wanted to do an experiment from which I might calculate *inherent* interdriver phase differences. My experiment proceeded as follows.

With my sound-level meter 3 feet in front of the tweeter and on its axis, I measured the response of the tweeter alone (without a filter); the woofers, in parallel, alone (without a filter); and the woofers and tweeter in parallel, phased either way (without filters).

Mr. Linkwitz has made several subjective evaluations of the perceptibility of nonlinear phase behavior and has concluded that only in very special circumstances and/or only with very special kinds of signals is nonlinear behavior perceptible. He seems to have been the only person to point out that if the drivers are not excited by in-phase signals, the main radiation lobe axis will wander as a function of frequency. Has Mr. Linkwitz evaluated the perceptibility of such behavior, and if so, what conclusions has he reached?

My intuition tells me that stable lobe axes and uniform magnitude are important at listening positions where the direct sound waves are dominant because that supports correct spatial perceptions. On the other hand, where the reverberant contribution is dominant, the sense of directionality is

SB *Mailbox*

essentially lost. Consequently, stable radiation axes contribute only trivially, and constant-power output may be the most important characteristic.

I calculated an apparent phase angle from the following formula:

$$S^2 = T^2+W^2 + 2TW \cos\theta$$

From one phasing, I calculated θ as 77 degrees, with a total variance of 8 degrees over seven frequencies. From the other phasing, I calculated θ equal to 124 degrees, with a total variance of 15 degrees. I expected the sum of these angles to total 180 degrees instead of 201. Can anyone explain this?

Then I postulated a 180-degree sum and used both sets of data to calculate θ from the relationship

$$\cos\theta = \frac{(S1)^2-(S2)^2}{4TW}$$

I obtained 117 degrees, with a total variance of 11 degrees. Is this a valid approach to the evaluation of inherent inter-driver phase differences? My reason for being interested is my growing conviction that without this information, designing crossovers rationally is a hopeless task.

David Meraner
Scotia, NY 12302

Mr. Linkwitz replies:

It surprises me that the results of Mr. Meraner's phase measurements show such inconsistency because the theory is certainly correct. For example, if the measured SPLs had been T' = 82dB, W' = 87dB, and S1' = 89dB, then the phase between T and W would be 77 degrees. Reversing the polarity of T should produce an S2' of 87.3dB. I assume that Mr. Meraner converted the decibel readings into linear terms. If you normalize to T' by subtracting 82dB, you obtain $T=1$, $W=10^{(87-82)/20}=1.78$, $S1=2.24$, *and* $S2=1.84$. *With these values, the phase becomes 77 degrees.*

As far as the perceptibility of nonlinear phase is concerned, I have come to the following conclusions. There are no audible effects from first or second-order all-passes such as 18dB or 24dB/octave acoustic crossovers, provided that the main radiation lobe axis does not wander. I derived this conclusion from experiments with W-T-W and W-T driver arrangements, and 18dB, 24dB and delay-derived crossovers where the crossover is at 1.5kHz. The phase behavior at the low-frequency roll-off is, however, quite important for natural-sounding bass reproduction. Since there is no way around the high-pass behavior of loudspeakers, the roll-off should be as gradual as possible. I think that 12dB/octave with a Q of 0.5 is probably as good as you can do.

Finally, if the reverberant contribution is dominant for the sound balance, a constant-power output would seem important. But I cannot see how such a system could reproduce stereo sound accurately.

ECONOMICAL UPGRADES

Upgrading an old pair of speakers can be an inexpensive way to get high-quality sound. I paid $5 for my first pair of Zenith speakers. They came with a cheap 6-inch woofer (5-ounce magnet, stiff cloth surround) and a large paper-cone tweeter (2 ounces) with a single capacitor crossover. The small 0.65-cubic-foot enclosure did not limit the woofer's excursion enough to prevent bottoming, which resulted in audible distortion and very limited power handling.

A few years back, I decided to replace the drivers to see what sort of reasonably priced upgrade I could get. I purchased a 6½-inch, butyl-surround woofer (20-ounce magnet, vented voice coil). Then I replaced the peaky paper-cone tweeter with an Audax tweeter and the 3.3μF capacitor with a second-order crossover with air-core inductors. I then mounted ¾-inch-thick strips of wood on the interior of the cabinet to reduce panel vibration. The magnets mounted on the baffle and on the grille frame hold the screen tight and look attractive.

Later, I mirror-imaged a pair of L-pads and mounting plates on the baffle. I read about diffraction rings in *SB* and have found that they work well and are easily constructed. The little rubber feet on the bottom of my speakers protect the veneer from scratches and are available from Radio Shack (Cat. No. 64-2342).

Now I have tight, clean bass and smooth, crisp extended highs. The system would benefit from a subwoofer, but I do not have room for it. The other equipment in my setup is a Mitsubishi DP-EC10 turntable, a Carver M-400 amplifier and a Great American Sound preamp.

Greg Cahill
Mt. Prospect, IL 60056

CERWIN-VEGA SYSTEM

It would be nice if A.B. Krueger (*SB* 4/84, p. 38) supplied details of his 17Hz woofer using the Cerwin-Vega D-189. Cabinet dimensions, port dimensions and other details would be most welcome. Incidentally, I question his ability to obtain 115dB out of an AR-1W speaker with 50W of input power.

Dick Nelson
Simi Valley, CA 93063

Mr. Krueger replies:

Building a Cerwin-Vega D-189 subwoofer is pretty easy and straightforward. First get a D-189E from Empirical Sound (1234 East 26th St., Cleveland, OH 44114). They sell rebuilt units for about $185 each. DLC Design (24166 Hagerty Rd., Farmington Hills, MI 48024) measured the Thiele/Small parameters of my rebuilt D-189, set in a

Continued on page 58

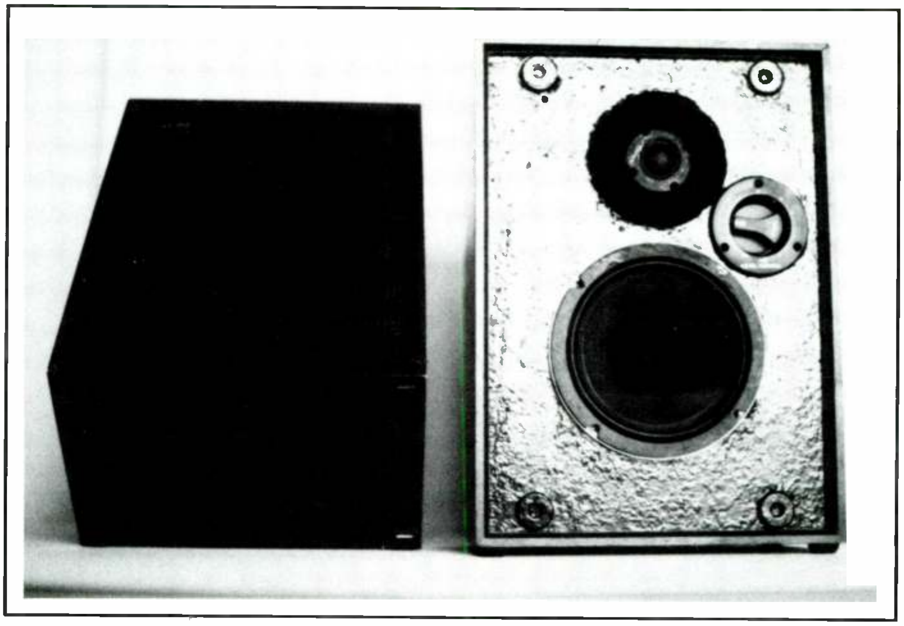


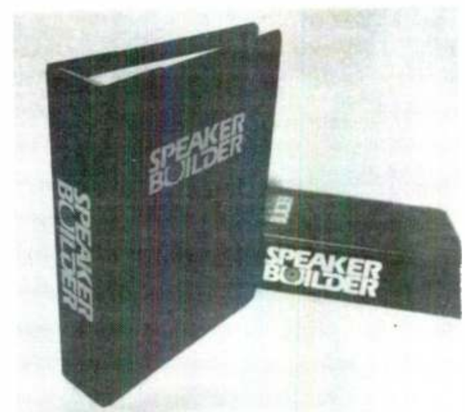
PHOTO 1: Mr. Cahill paid $5 for this pair of Zenith speakers, which he then upgraded by replacing the drivers, adding a crossover and making a few other inexpensive modifications.

SPEAKER BUILDER BINDERS & BOXES

Rich blue, heavy vinyl binders imprinted in white with ***Speaker Builder's*** logo. Spaces for twelve issues on loose wires will store your valuable ***Speaker Builder*** data in a readily accessible format.

Each ***$7.00***
Two binders, ea. ***6.50***
Three or more, ea. ***6.00***

Postpaid in USA. Outside USA please add $2. per binder for shipping. Canada must add $4. for one and $2. for each additional binder.



Boxes: Library quality boxes. 4¼x9¼x12" high, bright orange printed, folding corrugated boxes with handle/label holder to identify contents. Holds three years of ***Speaker Builder*** or other valuable magazines in your technical library. Each order supplied with preprinted labels for ***Speaker Builder*** and ***Audio Amateur.***

Two for ***$7.00***
Four for ***$10.50.***

Postpaid in USA. Outside USA please add 25% for shipping. Canada must add $3. for each set of two boxes and $1. for each additional set of two.

ORDER BLANK
Speaker Builder
PO Box 494, Peterborough NH 03458-0494

Charge card no.__________
Charge card expiration date__________
Signature __________

Send______ binder(s) @ $______ ea. Name__________
Send______ boxes for $______ ea. Street & No.__________
Total $______ ☐check; ☐M.O., ☐MCard; ☐Visa Town______ St.______ Zip______

SB Mailbox

Continued from page 56

14-cubic-foot box. We obtained the following results:

F_S = 23.8Hz
DCR = 4.67Ω
Q_{MS} = 4.14
Q_{ES} = 0.36
Q_{TS} = 0.33
V_{AS} = 24,489 cubic inches
Eff. = 1.45 percent

Plugging these values into DLC proprietor Dave Clark's computer, we obtained the following possible applications:

- *B4 alignment: box = 7.3 cubic feet; port area = 58.5 square inches; impedance "saddle point" = 29Hz; 3dB point = 31Hz.*
- *B4 alignment: box = 13.5 cubic feet; port area = 64.1 square inches; impedance "saddle point" = 24Hz; 3dB point = 24Hz.*
- *B6 alignment: box = 13.5 cubic feet; port area = 64.1 square inches; impedance "saddle point" = 20Hz; unequalized 3dB point = 30Hz. Equalizer: second-order high-pass, Q = 2, design frequency = 20Hz. Resultant 3dB point: 20Hz.*

Because these 3dB points are so low, "in-the-room" response will depend a lot on the listening environment.

Some risk factors are involved when using rebuilt loudspeakers. They are not consistent, and they are not always available. You might, therefore, wish to consider the JBL 15 and 18-inch subwoofers. They are very good and are more consistent than D-189E rebuilds, although the latter are lovely and will generally be less expensive.

Once you have obtained your subwoofer, build a sturdy box with a volume of between 7 and 14 cubic feet and a hole in it for the loudspeaker and one port. Line the box with 4-inch fiberglass batts. The required port area is satisfied by an 8-inch square hole. Port length will vary between 8 and 14 inches. Be aware that if the inside end of the port comes near a cabinet wall, it will extend the effective length of the port.

Tune the port for an impedance curve "saddle" around 20 to 250Hz. The better woofers, such as those recommended here, have a low Q_T of between 0.25 and 0.35, which allows considerable latitude in box design and tuning. Beware of high-Q_T woofers sold in the mass market. Radio Shack sells a 12-inch polypropylene unit with a Q_T of about 0.9. No reasonable box is really suited for it. For about the same price, Madisound will send you a Peerless TA-305F, which is quite nice.

Find an amplifier with a working 200W channel. At really low frequencies, nothing is perceived until things get quite loud. The ear is logarithmic, but the gut, hams and feet must be log-log. Several of us have remarked that music below 25Hz seems to be ignored until it gets to a certain loudness, and then it "snaps in."

Consulting Don Lancaster's Active Filter Cookbook, *build a high-pass filter with a corner frequency of about 20Hz and a Q of around 2. As you are probably aware, this is a B6 alignment equalizer. Some builders fret unnecessarily over the phase and power effects of a B6-aligned woofer. First, no sources of music below 100Hz are phase coherent unless operated in a very large anechoic chamber. That just never seems to happen. Second, the power savings from cutting things off sharply at 10 to 12Hz more than offset the extra power required at 20Hz.*

Phase incoherence at low frequencies is common in the real world. One demonstration of the audibility of phase shift at low frequencies used 40Hz tone bursts of four cycles, lasting one tenth of a second each. Forty-hertz musical tones last several seconds. The demonstration proves only that if you distort the time frame of music by a factor of ten, without distorting the frequency domain equivalently, you will have problems. Phase matching between audible sources is important, but that only means that you must be phase consistent, not phase perfect.

The filter might need some tuning of its Q, depending on how the standing waves line up in your listening situation. Connect a Panasonic P-5532 microphone ($1.95 from Digi-Key, PO Box 126, Thief River Falls, MN 56701; flat ±1dB from 20Hz to 20kHz) to a voltmeter with a 1mV range and a pink-noise source such as a CD player and bands 80–82 of Denon Audio Technical CD 38C39-7147. This allows you to verify that things are essentially flat, perhaps a decibel or two up at 25Hz. That means output at 16.66Hz will be within 3dB or so. The ear only notices variations of response on the order of 5 to 8dB at that frequency, so close hits score.

Use a 24dB/octave crossover at 100Hz for subwoofing. This frequency is not too high with sharp cutoffs. Suitable devices are available at a very reasonable price from Alpha Electronics (1415 Yorkshire, Grosse Pointe Park, MI 48230). The owner, Bob Klazca, is a friend, so take my recommendation with a grain of salt, but I think his circuit cards are first rate. He has a card for building 24dB/octave, phase-aligned Linkwitz-Riley-D'Appolito crossovers, two-way and up. One card is needed for a two-way subwoofer crossover. The phase corrector is not needed with a two-way unit and can be wired into a B6 filter.

The Cerwin-Vega has a sensitivity of about 98dB/watt, and 200W is 23dBw. Figure on 115dB down to 20Hz, including the effect of the 6dB or so boost for the B6 alignment and depending on the listening room. The Cerwin-Vega woofer is thermally limited to around 300W continuous, and my experience has been that it just will not bottom. With the volume all the way up and 105dB plus peaks, maximum voltage across the subwoofer voice coil was 6V, leaving 16dB headroom before amplifier clipping at 42V.

Mr. Nelson questions the output of my AR-1W speakers. The 110dB output with a 50W input is an actual measurement using my entire system, which consists of two AR-1Ws, two Morel MDM-75 3-inch dome midranges and two Morel MDT-27 tweeters. Amplifier power for the subwoofer was one channel of 200W, for the woofers two channels of 50W, for the midranges two channels of 200W, and for the tweeter two channels of 50W. For the measurements, I used a Radio Shack SPL meter at my listening location, with C weighting, slow meter response, musical program material and no amplifier clipping. As you can see, more than just the AR-1Ws were contributing to the measurement.

Many people are surprised to find that old AR-1Ws like mine are quite efficient by contemporary standards, measuring out at around 96dB/watt. Fifty watts is 17dBw, for a maximum output of 113dB at 1 meter. There is an approximately 3 to 4dB loss as the sound travels back to my listening location, which is about 10 feet away. A 110dB output from one AR-1W is possible, but I was not claiming that.

TRANSMISSION LINE

Continued from page 18

mended for "difficult" rooms. However, since the radiation pattern is designed to be narrow vertically, preferred placement is to raise the tweeters to ear height of the seated listener.

BUILDER REACTIONS. The stereo array of twenty sound sources (eight woofers, eight ports, and four tweeters) provides a delightfully diffuse, "speakerless" quality to the sound. Sometimes a multitude of sound sources also produces a clouded, indistinct effect. However, in the 2 x 4, the column sources, and cone loading by matched transmission lines, resulted in sound with striking clarity and crisp definition. The clean sound extends across a range of music, from folk songs to pipe organ. When the music calls for sound power, this two cubic foot system easily delivers more than my ears can handle with no noticeable distortion.

All in all, the listening pleasures have closely matched the original goals.

REFERENCES

1. Queen, D., "The Effect of Loudspeaker Radiation Patterns On Stereo Imaging and Clarity," *AES Loudspeakers*, (Volume 2), 1984, pp. 69-80.

2. Schroeder, M.R., Gottlob, D. and Siebrasse, K.F., "Comparative Study of European Concert Halls," *JAES*, (Volume 56, No. 4), October 1974.

3. Crowhurst, N., "How to Select and Install Your Own Speakers," TAB Books, 1979, pp.101-110.

4. Klepper, D.L. and Steele, D.W., "Constant Directional Characteristics From a Line Source Array," *AES Loudspeakers*, (Volume 1), 1980, pp. 100-110.

5. Allen, S. "Wood Finishers Handbook," Sterling Publishing Co., Inc., 1984.

SOURCES

Madisound Speaker Components
8982 Table Bluff Road
Madison, WI 53711

J. Ebbert
431 Old Eagle School Road
Strafford, PA 19087

Albert Constantine & Son Inc.
2050 Eastchester Road
Bronx, New York, NY 10461

SOMETHING NEW

Lots of magazines—most of them, in fact—offer readers special information services for products mentioned in news columns and advertisements. Usually only large magazines offer such services. We think we are one of the first small publishers to do so—and on our own computer.

But we need your help to make it all work. Remember to write your subscriber number (what we are calling your "magic number") on the ***FAST REPLY*** cards you mail in—and on any other correspondence about your subscription or orders for other items. That "magic number" is actually your record number on our computer. We have designed our little system to use that number in addition to the number of the vendor or vendors from which you wish to hear. Our clerk will enter only the numbers from your returned card, and our computer will do the rest.

If you have a friend who wishes to use the second ***FAST REPLY*** card, that's fine with us, even though your friend is not a subscriber. Just ask the person to check the non-subscriber box at the top, and we will take care of the rest.

Classified Advertising

TRADE

SPEAKERS, CROSSOVERS and accessories. We carry home, auto and professional speakers, we also feature Performance Sound Points. Free catalog. ***MARKEL PERFORMANCE SPEAKERS***, 18602 Kewanee Ave., Cleveland, OH 44119. T2/86

LONG HAIR WOOL carded, cleaned for stuffing speakers. $13.50/lb. including shipping. ***J. EBBERT***, 431 Old Eagle School Rd., Strafford, PA 19087, (215) 687-3609. TTF

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

ULTRA HIGH-END SPEAKER SYSTEMS using the best drivers from Europe: Dynaudio, VIFA, Siare, Audax, M.B....Low prices on kits or fully assembled systems for home—car—truck. Save $$. For free info-pak write ***DNL SOUND CO.,*** 10 Bellam Blvd., San Rafael, CA 94901. T3/86

ELECTRONIC CROSSOVER, 24dB/octave, fourth order, in-phase, low noise preamp. Power amplifier. More. Bare boards, kits. Free catalog. ***MARCHAND ELECTRONICS,*** 1334 Robinhood Lane, Webster, NY 14580. T2/86

ALPHA ELECTRONICS active crossovers: 24dB/octave, amplitude, phase correct. Two through six way, perfect for subwoofers. Boards, $39.95; kits, $97.50; complete systems, $199. ***ALPHA ELECTRONICS***, Box 15026, Detroit, MI 48215. T4/86

"IN PHASE" 24dB/octave electronic crossover minimizes irregularity in radiation pattern through crossover region. 0.0008% distortion, 5 year warranty. Single frequency with power supply (specify) $523. ***DB SYSTEMS***, Main St., Rindge, NH 03461. TTF

PYLE DRIVERS. America's finest, most complete line of raw speakers for stereo, car or guitar. Always discounted at the ***SPEAKER SHOP***, 318 South Wasatch, Colorado Springs, CO 80903, (303) 475-2545. T2/86

AUDIOPHILE ACCESSORIES

DBP-2J (5) SWITCH BOX..59.95, AU (5) gold jacks.....$71.95
Selects between up to 5 phono inputs. Used with DBP-6 or 6MC allows for selectable loading of cartridges

DBP-6 PHONO EQUALIZATION KIT.....$36.95
Allows adjusting the input capacitance of the phono input of every preamp and receiver with low loss Polystyrene Capacitors

DBP-6MC RESISTIVE LOADING KIT.....$36.95
Allows adjusting load resistance from 10 to 200 Ohms for moving coil cartridges. Gold plated phono plugs in both kits

DBP-9AU BANANA PLUGS Eight gold plated, solderless .$16.50

DBP-9H BANANA HANDLES Four red, four black .$5.50

DBP-9P GOLD PLATED DUAL BANANA PLUGS 2 pk $17.95

DBP-9J GOLD PLATED DUAL BANANA JACKS 2 pk $17.95

DBP-10 PHONO ALIGNMENT PROTRACTOR.....24.95
Allows adjusting the lateral tracking error of a mounted cartridge to within ¼ of one degree. Non-technical instructions & case included

DBP-12 AUDIO CABLE 10 meter (33 ft.).....$70.00
Low capacitance (400pF) stereo interconnect cable terminated with rugged gold plated phono connectors. Custom lengths available

DBP-12X AUDIO CABLE 10 meter (33 ft.).....$95.00
Low capacitance (400pF) stereo interconnect cable terminated with premium DBP-13PX phono connectors. Custom lengths available

DBP-13J GOLD PLATED PHONO JACKS (¼") 8 pk $14.95

DBP-13JR GOLD PLATED PHONO JACKS (3/8") 8 pk $19.95

DBP-13P GOLD PLATED PHONO PLUGS 8 pk .$9.95

DBP-13PM GOLD PLATED PHONO PLUGS, 4 red, 4 blk .$21.95

DBP-13PX GOLD PLATED PHONO PLUGS 2 pk $14.95

DBP-14 GOLD PLATED SPADE LUGS 8 pk $5.95

DBP-15 GOLD PLATED "Y" ADAPTORS 2 pk $12.95

DBP-CK CRAMOLIN AUDIO KIT contact treatment $16.50

DBP-SC SOUTHER RECORD CLAMP.....$10.00

ELECTRONIC CROSSOVERS ..6, 12, 18, 24dB Inquire

At your dealer or direct. Orders under $45 add $2.50 Handling
MC/VISA

DB SYSTEMS
Main St., Rindge, NH 03461 (603) 899-5121

PRIVATE CLASSIFIED ADVERTISING SPACE up to 50 words in length *is open to* Speaker Builder's *subscribers without charge for personal, non-commercial sales and for seeking information or assistance. The publishers reserve the right to omit any ad. Any words beyond 50 are* **20 cents per word.** *Please type or neatly print ad copy on a separate sheet or card with your full name and address.*

TRADE CLASSIFIED ADVERTISING RATES: 45 cents per word *including name, address and zip code—prepaid only. 10% discount for four insertions.* Speaker Builder *cannot accept responsibility for the claims of either the buyer or the seller.*

SIDEREALKAP. Not just another generic capacitor with a wonderful name, the Sidereal-kap was designed from its inception to be the finest sounding capacitor available for audio today. Find out what the music lover's capacitor doesn't sound like. Free literature and price information upon request. Dealer inquiries invited. ***SIDEREAL AKUSTIC***, 1969 Outrigger Way, Oceanside, CA 92054, (619) 722-7707. T2/86

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.

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.

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.

YOUR 50 WORDS
in our private classifieds ARE FREE
OVER 50: Send money (20 cents/word)

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 incorporate and include "live" music occassionally. We are presently meeting every 5-6 weeks (subject to change). Audiophiles interested contact Frank Manrique, President, IAS, 1219 Fulbright Ave., Redlands, CA 92373, (714) 793-9209.

CENTRAL FLORIDA AUDIO SOCIETY

meets monthly in Orlando. Come and meet others who share your interest in music reproduction as we audition equipment and recordings or discuss audio related topics.

Contact: Bill Boswell (305) 678-3724
P.O. Box 579, Orlando, FL 32802

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 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.

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 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.

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.

FOR SALE

Apt Holman HP preamp $495; Apt 1 Holamn amplifier, $495; mint condition, original boxes and manuals. Kenneth Mark, (703) 938-7220.

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

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

ReVox B77 tape recorder, $895; Denon DP 2500 with 103D cartridge, $325; JBL O77 slot radiators, $95 pair. Wesley Kellie, 3805 Sherwood Dr., Coeur d'Alene, ID 83814, (208) 765-8521 after 5 p.m. PST.

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

Classified Advertising

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

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

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

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

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

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

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

Stromberg-Carlson turntable rebuilt with Shure M91 cartridge, $40; BSR SX100 spectrum analyzer with microphone, $100; Realistic 12-band equalizer with image expander, $80; muffin fans, 3 and 5-inch, $8 each; high-voltage power transformers, SASE for list. WANTED: good AM tuner such as Mckay-Dymek. TRADE: 12AX7s for 6V6GTs. Ben Poehland, 14 Carol Lane, Malvern, PA 19355, (215) 644-3677.

Audionics Space & Image Composer, (SQ and surround-sound system), very good condition with new R/C, $400; dbx 3BX-R remote control, new, never used, $100; Signet MK12T moving-coil transformer, very good condition, $75. All items shipped UPS prepaid. William Sommerwerck, 133 West Locust, Apt. 217, Mechanicsburg, PA 17055.

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

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

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

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

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

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

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

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

WANTED

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

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

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

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

Advertising Index

Old Colony Sound Lab's

BOOK SERVICE

PENGUINS

P-1 THE COMPLETE PENGUIN STEREO RECORD & CASSETTE GUIDE (3RD ED.) *by Edward Greenfield, Robert Layton, and Ivan March.* Drawing on profound technical knowledge and on vast musical and historical learning, this newly revised and updated guide to recorded classical music deals with more than 4,000 disks, cassettes, and compact discs, giving details of title, performers, record number, label and price range. 1387pp., softbound. *Please add $1.00 for shipping.*
Each **$12.95**

P-2 A NEW DICTIONARY OF ELECTRONICS *by E. C. Young.* This remarkably compact reference covers electronics from A-Battery to Z-parameters with succinct, concise definitions and illustrations. A quick reference completely revised and updated with lots of added charts and reference data. 618pp., softbound. Each **$5.95**

P-3 A NEW DICTIONARY OF MUSIC (3RD ED. 1973; FIRST PUB. 1958) *by Arthur Jacobs.* Alphabetically arranged entries covering composers, individual musical works, orchestras, performers, conductors, musical instruments, and technical terms. 458pp., softbound. Each **$5.95**

P-8 INTRODUCING MUSIC *by Otto Karolyi.* Even if you've listened to music for years, you may be surprised at how little you know about its fundamentals. A beautifully basic and spare introduction to the grammar and vocabulary of music—enough to understand the language without speaking it. It will deepen almost any non-musician music lover's pleasure in listening. 174pp., softbound. Each **$4.95**

McGRAW-HILL

MH-1 HANDBOOK FOR ELECTRONICS ENGINEERING TECHNICIANS (2ND ED.) *by M. Kaufman and A.H. Seidman.* A comprehensive compendium of electronic facts. Component selection, circuit analysis, power supplies, IC uses and characteristics, op amps, transistors, batteries and tubes. A one-volume encyclopedia on how components work and how to choose the best of them for your application. Includes seven new chapters on such topics as logic analysis, fiber optics, active filters, digital test gear and microprocessors. 752pp., hardbound. Each **$39.50**

MH-2 HOW TO MAKE PRINTED CIRCUIT BOARDS *by Joel Goldberg.* All the topics on making your own circuit boards are covered both in theory and in practical advice: design and layout, artwork preparation, photo layout, silk screening and etching. Each **$7.95**

WILEY BOOKS

W-1 HIGH PERFORMANCE LOUDSPEAKERS (3RD ED.) *by Martin Colloms.* A thorough revision of the 2ND ED. (1980) updates and expands this authoritative volume. The author surveys new developments in the four-year interim and adds some highly original proposals for low frequency alignments, listening tests and much else. For the speaker builder, Colloms' book is one of three indispensable volumes. Hardbound.
Each **$29.95**

HOWARD W. SAMS

S-10 IC OP-AMP COOKBOOK (3RD ED.) *by Walter G. Jung.* Jung's popular classic in a revised and expanded 1986 edition. One hundred pages are added to cover new devices, applications and manufacturer's data sheets. With over two hundred practical circuits with component values, this cookbook is one of Old Colony's best selling technical volumes. It belongs on every electronics bookshelf. 580pp., softbound.
Each **$21.95**

S-11 HOW TO BUILD SPEAKER ENCLOSURES *by Alexis Badmaieff and Don Davis.* The "whys" and "hows" of speaker enclosures. Drawings and instructions cover infinite baffle, bass reflex and horn types plus combinations. 144pp., softbound. Each **$6.95**

S-19 SOUND SYSTEM ENGINEERING *by Don and Carolyn Davis.* A thorough introduction to sound systems for halls, studios, outdoor locations and much else. 245pp., hardbound, 8½x11. Each **$24.95**

DOVER BOOKS

D-1 REPRODUCTION OF SOUND *by Edgar Villchur.* An elegantly simple 92-page paperbound primer on the subject by the man who invented the bookshelf speaker. Loan or give it to friends who want to build a system as good as yours.
Each **$6.00**

D-3 MUSIC, PHYSICS AND ENGINEERING *by Harry F. Olson.* A thorough introduction to the physical characteristics of sound and the relationship of sound to musical instruments by the former head of staff at RCA's lab for acoustical and electromechanical research at Princeton, NJ. A classic by one of the giants in the audio field. Good, easy to read chapters on acoustics, mikes and recording, recording and playback systems, as well as an electronic music chapter. 2ND ED. (1967) 460pp., softbound. Each **$14.25**

Old Colony's

BOOK SERVICE

To order from **Old Colony Sound Lab**, please write each book's number below with quantity of each and price. Total the amounts and remit by check, money order, MasterCard, or Visa. *For orders under $10 add $2 service charge.* Please add $1 for postage on the first book; 50¢ per additional book. Canadians please add 10% for postage. All remittances must be in U.S. funds. *Please use clear block capitals.* NOTE: *Prices are subject to change without notice.*

Number of Books		Price
....	Book No.	$..........
....	Book No.	$..........
....	Book No.	$..........
....	Book No.	$..........
		Postage $..........
		TOTAL $..........

Name ______________________________

Street & No. ______________________________

City ______________________________

State ______________________ ZIP ______________

PO BOX 243 • PETERBOROUGH, NH 03458 • CHARGE ORDERS (603) 924-6371 • 9-4 M-F EDT

McGee Radio

Over **50 years** in business.
Over **50 million** speakers sold!

For over 50 years, McGee Radio has given the US people the very *best* buys in speaker systems.

BUILD YOUR OWN CABINETS AND SAVE BIG!

New for 1986: Our REFERENCE LINE

Instead of all the speaker verbage, we'll get right to the point. This Subwoofer Satellite System has been rated as outperforming systems with prices up to $2,300.00. For only **$249.95** we will deliver to you anywhere in the continental United States with no extra costs (no shipping, no handling) the following system components:

- 1 12" Dual Voice Coil, 8 OHM, 40 oz. Magnet Sub Woofer.
- 2 6-1/2" Polypropylene Curve Linear Cones, Butyl Edge , 30 oz. Magnet, 1.33" VC Mid-Bass.
- 2 5-1/4" 10 oz. Deep Basket Mid-Range.
- 2 1" 10 oz. Soft Dome Tweeters.
- 2 12 dB (Second Order) Phase Coherent Crossovers, 150 Watts
- 1 Dual Input Subwoofer Crossover.

ALL FOR $249.95 !

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

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

Speaker:	Size	VC	Imp	Mag	fs	O+S	VAS (Cu.Ft)	Power*	Eff	Retail Price:	McGee Price:
TP165F	6-1/2"	1.25	4&8	15oz.	53	.42	.55	80	89	27.95	**15.95**
TP165R	6-1/2"	1.25	8	15oz.	33	.321	1.41	80	80	38.95	**21.95**
TX205F	8"	1.25	4&8	20oz.	42	.40	1.73	80	90.5	34.95	**19.95**
TD205R	8"	1.5	8	27oz.	32	.326	2.12	100	90	47.95	**26.95**
TX255F	10"	1.25	4&8	20oz.	28	.527	4.31	100	88	40.95	**22.95**
TD255F	10"	1.5	8	27oz.	23	.252	5.47	150	90	48.95	**27.95**
TA305F	12"	2.0	.8	38oz.	25	.306	6.64	200	89.5	54.95	**30.95**
TA305FSW	12"	2.0	2- 4	38oz.	21	.332	8.62	200	89	75.95	**42.95**

* Power rated in RMS undistorted music program material.

49er SPECIAL!

This speaker system normally retails at $399/Pair. System Frequency 49 to 20 KHz. Great with a 60 Watt Amplifier! The 49er System includes all of the following:

- 1 #AD7066W8 6-1/2" Butyl Edge European Woofer, 16 oz. Magnet, 1" VC.
- 1 #AD5061/SO 5" Closed Back Mid-Range, Chamber Dampened, 10 oz. Magnet, 1" VC
- 1 #AD0163/T 1" Polymer Dome, 10 oz. ABF Magnet, 1" Low Mass Voice Coil, Made in Europe.
- 1 Crossover. 2nd Order, Phase Coherent, 150 Watt.

ALL FOR ONLY $49.00 !!!!

CABINET (WITH GRILL) AVAILABLE FOR $49.00 IN ADDITION TO THE SPEAKERS.

WE CARRY THE COMPLETE LINE OF PYLE RAW DRIVERS.

SALE PRICES GOOD THRU APRIL 15,1986

McGee Radio. The FIRST in speakers. Send $2.00 for your McGee catalog. Not just a price list
All pertinent data listed. $2.00 refunded on first order of $20.00 or more.

1901 McGee Street Kansas City, Missouri 64108-1891 PH 816-842-5092

TO PLACE VISA, MC, or AE CHARGE ORDERS CALL 1-800-34MC GEE (1-800-346-2433)