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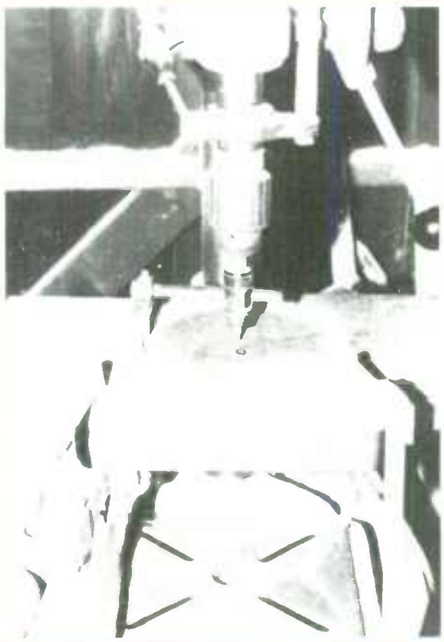
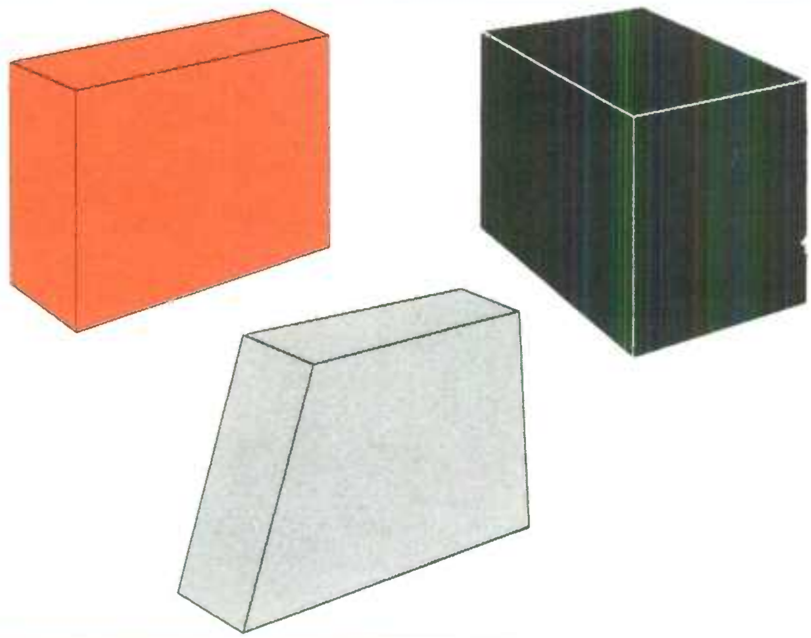
Speaker Builder

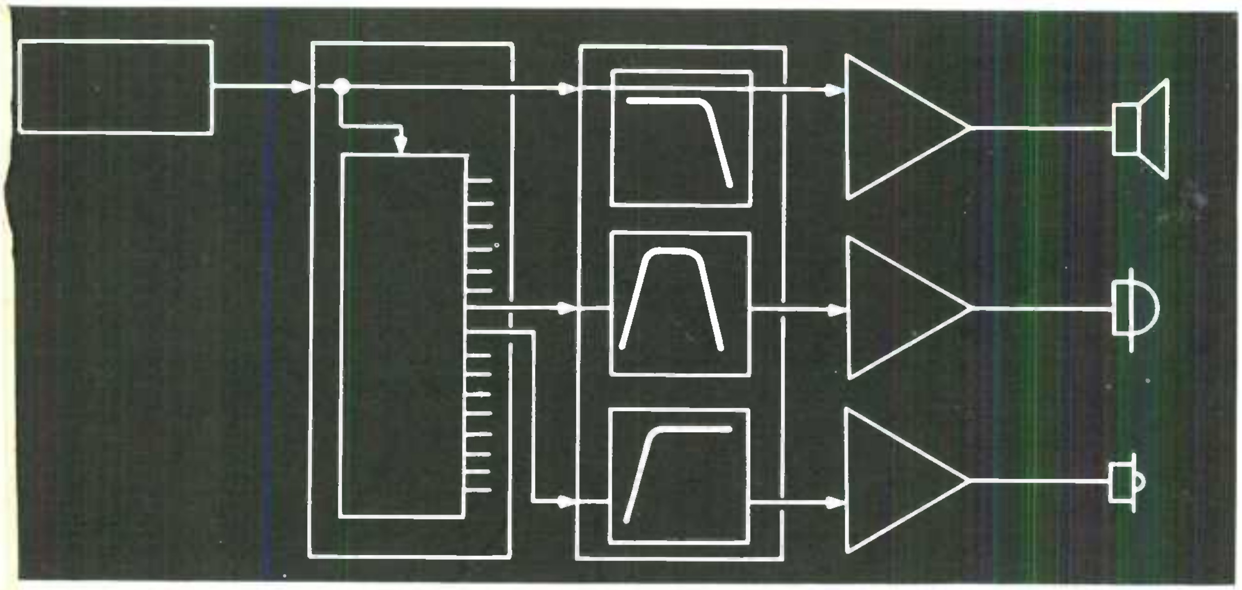
THE LOUDSPEAKER JOURNAL





AR-3 RE-DO

SHAPES & VOLUMES



ELECTRONIC TIME DELAY

SPEAKER CITY U.S.A. ANNOUNCES NEW KEVLAR® & CARBON GRAPHITE

Speakers by Versa Tronics

Designed and manufactured in France to the highest quality European standards. Versa-Tronics had them designed to incorporate the latest state-of-the-art technology in woven Kevlar® and Carbon Graphite cone materials. The true form shaped cones are a decided improvement over the sandwich types which do not allow for any other cone shape other than a straight side. The possibility of curvilinear cones does not exist with sandwich honey-combed types.

The Carbon Graphite cloth is treated on the rear with a hard resin and the front of the cone is impregnated with a butyl-latex. This provides excellent internal damping along with the desired rigidity-mass ratio. The woven Kevlar® has a rear treatment of a hard lacquer type resin along with the face treatment of a specially formulated latex. Both the Kevlar® and Carbon Graphite diaphragms provide open sound without coloration. This plus many duo-voice coils, die-cast baskets and brass-capped pole pieces are offered in these new Versa-Tronics products.

F013R-KV 5¼-Inch

Impedance	8Ω
Music Power DIN	80W
Sensitivity 1W/1M	90dB
Resonance Freq.	60Hz
Upper Freq.	8kHz
Voice Coil	1″ Nomex
Cone Material	Kevlar®
Surround	Rubber
Magnet Wt.	550 gm
Vas	8.86 L
Qts	0.316
Qes	0.365
Qms	2.4

F017R-KV 6½-Inch with Bullet

Impedance	8Ω
Music Power DIN	100W
Sensitivity 1W/1M	90dB
Resonance Freq.	45Hz
Upper Freq.	8kHz
Voice Coil	1″ Nomex
Cone Material	Kevlar®
Surround	Rubber
Magnet Wt.	550 gm
Vas	53 L
Qts	0.33
Qes	0.37
Qms	2.77

GQ20R-KVC 8-Inch Die Cast Basket

Impedance	8Ω
Music Power DIN	140W
Sensitivity 1W/1M	93dB
Resonance Freq.	45Hz
Upper Freq.	7kHz
Voice Coil	1½″ Kapton
Cone Material	Kevlar®
Surround	Rubber
Magnet Wt	780 gm
Vas	45 L
Qts	0.39
Qes	0.60
Qms	1.05

F017R-CG DB 6½-Inch Duo-Voice Coil

Impedance	8Ω
Music Power DIN	100W
Sensitivity 1W/1M	89dB
Resonance Freq.	50Hz
Upper Freq.	5kHz
Voice Coil	1″ Nomex
Cone Material	Carbon
Surround	Rubber
Magnet Wt.	550 gm x 2
Vas	20.57 L
Qts	0.627
Qes	0.766
Qms	3.46

F020R-CG 8-Inch

Impedance	8Ω
Music Power DIN	100W
Sensitivity 1W/1M	90dB
Resonance Freq.	35Hz
Upper Freq	8kHz
Voice Coil	1″ Aluminum
Cone Material	Carbon
Surround	Rubber
Magnet Wt.	550 gm
Vas	75 L
Qts	0.35
Qes	0.41
Qms	2.52

GQ25F-CGC 10-Inch Die-Cast Basket

Impedance	8Ω
Music Power DIN	120W
Sensitivity 1W/1M	90dB
Resonance Freq	28Hz
Upper Freq.	5kHz
Voice Coil	1½″ Aluminum
Cone Material	Carbon
Surround	Foam
Magnet Wt.	780 gm
Vas	181 L
Qts	0.512
Qes	0.63
Qms	2.76

SPEAKER CITY U.S.A.

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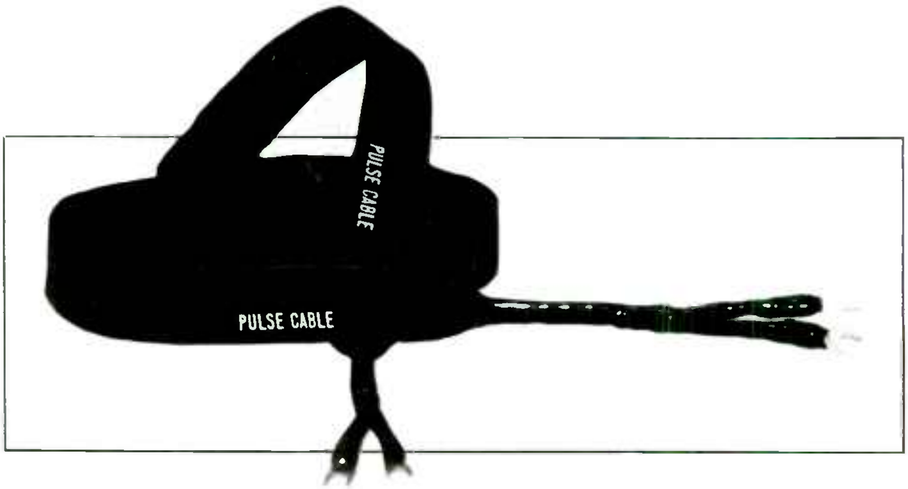
Reply #GC683

Good News



A new high performance speaker cable is available from **PULSE TECHNOLOGIES**. Pulse Cable is a flexible, 1¼ by $^1/_{16}$-inch flat cable made of copper with polypropylene sheathing, engineered to improve power transfer and frequency response. The terminations (pin or spade types) are military standard gold, insulated with red and black vinyl boots.

The cable is designed with regard to waveform linearity and phase response. Pulse Technology says the test results are measurable, minimizing high frequency attenuation, phase interaction and capacitative effects.

Prices per pair: 10 feet, $41.80; 15 feet, $56.70; 20 feet, $67.60. Contact James Vidican, Pulse Technologies, 125 St. Louis St., Elwood, IL 60421-0329, (815) 722-1515.

Fast Reply #GC78

PRECISION LOUDSPEAKERS introduces the TD255F-SW 10-inch dual voice coil subwoofer.

Designed for a center-channel, medium-size subwoofer for home or auto applications, it utilizes Precision's proprietary mineral filled polypropylene cone material for low coloration and distortion; high-temperature aluminum former voice coil for high power handling; and polyurethane-foam surround for controlled stiffness. The motor assembly is optimized for reflex applications, but it can be used for high-level, efficient closed-box systems.

Thiele/Small parameters are listed as: F_s, 27Hz; V_{as}, 130 liters; Q_{ms}, 4.03; Q_{es}, 0.37; Q_{ts}, 0.34; SPL, 90.7dB (2.83V, 1M, 1 coil driven).

For more technical information and applications contact Dick Pierce at Precision Loudspeakers, 2-B Columbia Dr., Amherst, NH 03031, (603) 883-7050.

Fast Reply #GC46

A new stereo speaker system designed by audio pioneer Henry Kloss is available from **CAMBRIDGE SOUNDWORKS**, his new company. Ensemble is a four-component system which consists of a pair of compact low-frequency 12 by 21 by 4½-inch units and a pair of satellite units, 5½ by 8 by 4 inches.

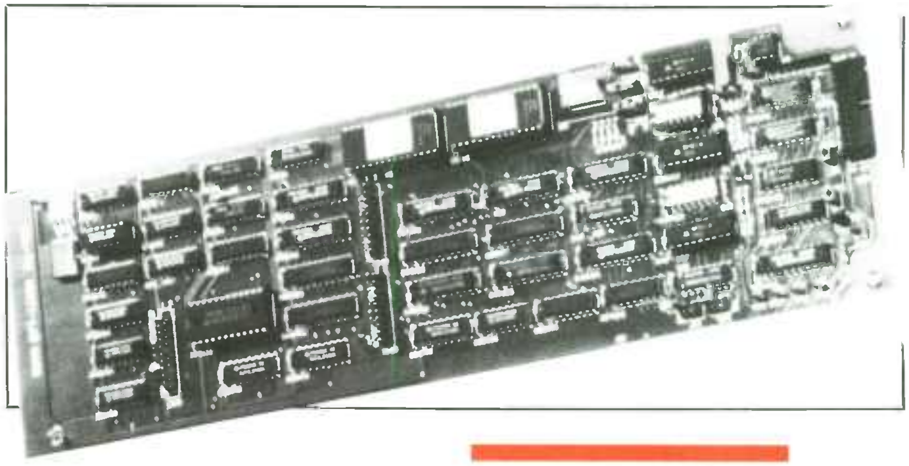
The subwoofers use 8-inch, long-throw acoustic suspension drivers with butyl surrounds, 12dB/octave LC low-pass networks with 170Hz nominal crossover frequency; system resonance 54Hz, Q of 0.7.

The satellites use 3½-inch cone drivers with impregnated fabric surrounds; 12dB/octave LCR low- and high-pass networks with 170Hz and 2.3kHz nominal crossover frequencies. The 1¾-inch direct radiator tweeters have ⅜-inch center domes, butyl surrounds and R/C frequency contouring networks.

Cabinets are high density particle board, finished in black laminate and dark gray Nextel with metal grilles. Power requirements are 25–100W per channel.

The complete system is sold directly to consumers for $499, plus shipping. Contact Cambridge Soundworks, Tom DeVesto, 154 California St., Newton, MA 02158, (617) 332-9461 or 1-800-AKA-HIFI.

Fast Reply #GC44



The R712, from **RAPID SYSTEMS**, is a multifunction analog/digital I/O card for the PC, XT, AT, or compatible computer. This card offers 12-bit resolution in both A/D and D/A conversion. Also on the card are a three-channel programmable interval timer/counter, and 16 digital inputs and outputs. Sample programs are included with the user manual and controlling software is available.

Features include: 16 single-ended analog inputs 10MΩ input impedance, A/D conversion up to 33kHz, 2 analog outputs, D/A conversion up to 833kHz and 12 bits of accuracy.

Order from: Rapid Systems, 433 N. 34th St., Seattle, WA 98103, (206) 547-8311.

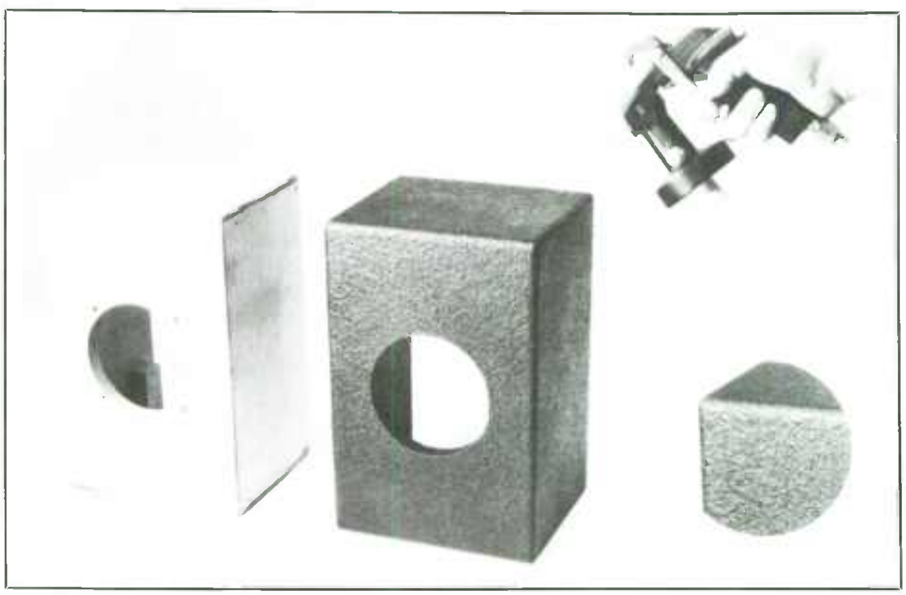
Fast Reply #GC948

In April, **NEW ENGLAND AUDIO RESEARCH, INC.** (NEAR), purchased manufacturing and distribution rights to all **BOZAK** speaker products, and all products will accordingly carry the NEAR logo.

Former Marketing Director of Bozak, William Kieltyka, will direct operations as President of the new firm. In May, speaker products for the consumer market and the commercial sound market were introduced. In addition, a newly developed indoor/outdoor, high-end compact system is now available. The AES-2 is easily portable and the metal alloy cone material makes it impervious to moisture or heat for year-round outdoor use. Retail price is $275.

Dealer and consumer inquiries are welcome: contact NEAR, Bill Kieltyka or Robert Adams, 5 East St., New Britain, CT 06051, (203) 225-7816.

Fast Reply #GC76



Texturelac, new from **ARD**, is a paint product designed for speaker cabinets that gives a spray finish that looks like pebbles, stucco or leather. The different patterns are created by adjusting air and fluid pressures using standard Binks and DeVilbiss guns.

Texturelac dries in 15 minutes and eliminates sanding, spackling and multiple finishing coats. It also hides blemishes and imperfections, for use on a variety of lower cost materials, including metal. It is washable and available in a variety of colors.

For more details contact: Abilene Research and Development Corp., PO Box 294, Hewlett, NY 11557, (516) 791-6943.

Fast Reply #GC66

E.J. JORDAN USA announces initial response to their Owners Club provided excellent member contributions to potential purchasers of Jordan drivers and related products. The club continues to seek members and welcomes hobbyists of all experiences, as many Jordan systems utilize other manufacturers' drivers. The Club will also update new technologies and components. For information send a card to: E.J. Jordan USA, Box 301, Suite 252 Bldg. B, Princeton, NJ 08540.

Fast Reply #GC553

AUDIO CONTROL has completed another in their series of technical papers, this one exploring recording processes. "How Recordings Are Made" was written by Rick Chinn, a working sound designer with 15 years experience in all facets of recording, sound reinforcement and acoustical design.

The technical paper intends to educate a broad audience within the industry, particulary dealer personnel and the more avid consumer.

Technical paper #105, price, $1, is available from: Audio Control, 22313 70th Ave. W., Mountlake Terrace, WA 98043, (206) 775-8461.

Fast Reply #GC123

MAHOGANY SOUND introduces a new synthetic long-fiber sound absorption material for high-end speaker systems, which they claim has the acoustic purity of wool.

Acousta-stuf uses a multi-directional crimp in the fiber to give it more surface area to break up and disperse the speaker's rear wave. Recommended for use in transmission lines and reflex enclosures as an alternative to the higher cost of wool, it requires no extra "care and feeding." Price is $7 per pound.

Contact Mahogany Sound, Larry Sharp, 2430 Schillingers Rd. #488, Mobile, AL 36695.

Fast Reply #GC73

MARRS DEVELOPMENT, INC., announces it is beginning initial production of its Super Compliant Modules for use in high performance speakers. The modules, made from an ultra-thin, high-tech polymer, fit inside a loudspeaker to provide a softer cushion for the internal bass waves than air alone. Marrs claims the modules will double low-frequency performance in a compact speaker.

The process for the technology has been developed and patented since 1980, and the company now plans to license the production to manufacturers.

For more information contact Ralph Marrs, 5765 Winfield Blvd., Unit 4, San Jose, CA 95123, (408) 629-8520.

Fast Reply #GC77

SPEAKER BUILDER

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About This Issue

Our horn expert, Contributing Editor **Bruce Edgar**, proves he has more than one string in his bow, giving us a thorough analysis of the venerable Acoustic Research 3-A's strengths and weaknesses and then a masterful renovation plan which produces a re-incarnation superior to the original (page 9), full of construction technique bonuses as well.

Mark Rumreich tackles one of the knottiest problems in speakerdom: how to align the drivers so they speak together. Rather than tilting the box or angling the front panel, Mark chooses the electronic route, using a series of filters to get the driver relationship right. Another breakthrough for speaker builders.

Lubos Palounek is intrigued by the relationship between cabinet shape and performance. His article (p. 22) gives you a method of assembling a "library of shapes" for box volumes.

Ralph Gonzalez takes a close look at a commercial product that apparently has solved the multi-driver phase problem (p. 34) and offers suggestions about how builders may achieve similar results in their projects.

Tom Cox noticed that his room was giving his system (and his ears) a lot of overactive, bounced frequencies so he decided to modify the room with movable panels that make the waves one-way. His controlled set of measurements of various absorptive materials gives you the benefit of his tedious homework.

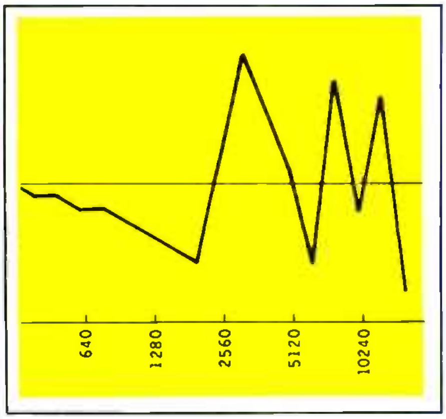
Subwoofers are a popular panacea for emaciated low frequencies but adding one or a pair of them to your system is often more difficult than it seems. **Jim Franc** has suggestions on how to go about the procedure.

If you'd like a driver made of a candle or a gas jet, **Peter Muxlow** has the technology for you (p. 51). **Gary Galo** reviews two helpful books on maintaining your computer (p. 53). Don't miss the tips by readers **Gitto**, **Bock**, **Hansen** and **Schneider** starting on page 47. As a bonus, this time we include a sample issue of our industry newsletter, *Voice Coil*. We're happy to say it is becoming widely accepted in the industry with over a third of the market subscribing.

By the time you receive this, we will have moved to new quarters. Same town, same postal address, and the same telephone numbers. After a short break, your next issue of *Speaker Builder* will be mailed at the end of July.

SpeakerBuilder
THE LOUDSPEAKER JOURNAL

VOLUME 9 NUMBER 3 MAY 1988



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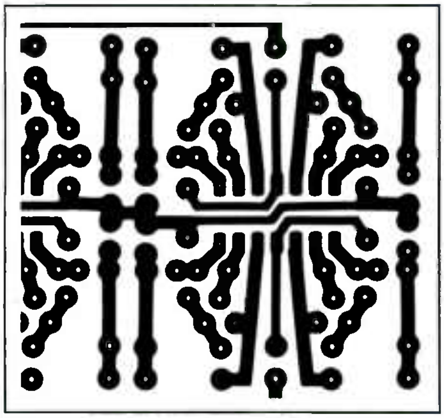

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FEATURES

DEPARTMENTS

Editorial

Underfoot Fi

We speaker builders spend a majority of our avocation time thinking about boxes or horns or diaphragms in frames as a way to get our reproduced music back into space. Lately I have noticed more discussion of the enclosure for the enclosures: the room.

All of us are aware, in a remote, back-burner way, that our rooms are a factor in how the system sounds. Few of us are in a position to change things very much.

Recording professionals spend a fair amount of time thinking about their studios—the environments for the performers and for those in the control room. On this latter point, most audio enthusiasts think the engineers spend far too little thought and care on the monitor systems and the environment in the control room—but that's another subject.

We are seeing a definite rise in the study of rooms. Witness *Audio*'s recent pair of articles on the Don Davis' "Live End/Dead End" concept. And who has not dreamed of the legacy that makes building a new listening room addition to the house possible? I had the good fortune years ago to have a friend in that enviable position. He built a marvelous extension to his house on Cape Ann on the Massachusetts north shore to house a small pipe organ and his audio system. No two walls were parallel and the acoustics were controllable by hinged wall panels to adjust for the presence of a small audience. The project was a remarkable success.

Most of the discussions I hear on the topic, however, talk about walls; sizes, shapes, absorption factors and so on. Nobody talks about floors. Once in a while we correct for the floor problem in trying to site a phonograph so the floor does not affect it. This clue tells us most floors are more like walking on a spring-mounted panel than on the solid surface we mentally suppose them to be.

We live in the first age where we are beginning to discover that the earth's crust under our feet is not the solid, inflexible ground we have always considered it to be. It still hurts to fall down—but in the larger picture, the earth is more like an ovoid, jelly-filled egg, flying through space, whose relatively thin surface plates are still sliding around looking for a final resting site.

I recently moved from an apartment which is part of a four story post-and-beam barn built in the 1820s. In a high wind you could feel and almost see it move. Since I was on its top floor the sensation was somewhat nautical. I longed to mount my Ivor Tiefenbrum special on a shelf hung on a bearing wall but in rented space you don't do that.

I have moved to a 30-year old slab house with a solid concrete floor throughout. My new listening room is 28 x 13 x 7.5 feet. The pleasure of jumping up and down within a foot of the phonograph without the slightest noticeable effect in the sound reproduction is novel.

I have no data to offer as yet, but I believe the solid floor (even with its shaky underpinning of the New England plate beneath it) is giving me a remarkable difference in low-frequency response. I look forward to finding the time to make some measurements when my system is fully in place. Unfortunately, the walls of my much younger house are nothing as substantial as those of the old barn I vacated. The panelled walls are far more flexible than the old plaster in my former abode. But we shall see what effect this may have on quality.

I am certain some of you have given thought to room effects but I do not recall seeing floors discussed as other than a bad, mushy underpinning for phonographs. Your input is welcome, of course. It isn't a problem your local hi-fi emporium is going to solve by selling you a replacement or a mod kit. But it is a factor which must be part of the mix if we are to fully understand what our systems are doing.

Think Small

The quality and complexity curve of *Speaker Builder* articles over the last three years has been rising at a steeper and steeper rate. Articles are getting better, more complex and more demanding. But new *Speaker Builder* readers are joining us weekly. We are growing rapidly and a lot of the new people are arriving in the middle of the party—or the seminar—and feel a little lost.

Some of you would-be authors are obviously feeling a little intimidated by the complexity and polish of what we are publishing these days. You should not be. Let me encourage you, all of you, to think seriously about sharing your speaker building adventure with your fellow readers, no matter how simple you may think they are. Our *Tools, Tips and Techniques* is perpetually empty these days. We need your Craftsman's Corner items and an account of your simple projects.

We also need small projects suitable for those just starting out on the road to speaker literacy. And we are at work commissioning articles which tell you how to get started in woodworking, simple electronics, and how to stuff a line and other related topics.

If you are a new arrival, don't hesitate to ask about what you don't understand. All we ask is that you type or write legibly, include a stamped and addressed envelope so we can reply easily, and state your question as clearly as possible. Good questions are valuable because they remind those of us who have been building these gadgets forever, how much we take for granted that we have learned along the way.

Help us keep the ladder to good speakers firmly planted on the ground. I welcome your work which will help new arrivals to get on board and under way.—E.T.D.

A LOW-COST UPGRADE FOR THE AR-3

BY BRUCE C. EDGAR
Contributing Editor

As a speaker builder, you will no doubt run into this situation: Your in-law, neighbor, or friend, for example, says, "I have this old pair of AR-3s that don't work well anymore. Can you repair them and bring them up to date?"

Usually, with AR-3s, the open core rheostats used for pads have oxidized, and the tweeters have stopped working. A simple fix is to carefully pry out the woofer and spray the pots with contact cleaner. Or, replace the pots with L-pads or with new ones from AR[1] However, the tweeter may be blown; in which case you can replace it through AR, or retrofit a more modern driver.

In previous *SB* articles, Walt D'Ascenzo suggested ways of upgrading the Acoustic Research AR-3[2,3] But, my friends wanted an inexpensive upgrade to their AR-3s, which some aspects of D'Ascenzo's imaginative approach did not cover. Thus, I describe a relatively inexpensive and straightforward upgrade for the AR-3. In addition, I will review some interesting aspects of the AR-3 design I learned from former AR employees, who were present during and after the early "Villchur" years, when co-founder Edgar Villchur ran the company with Henry Kloss.

THE AR-3 DESIGN. The AR-3 loudspeaker (*Photo 1*), designed by Edgar Villchur[4] in the early fifties, is now recognized as a classic American loudspeaker design[5,8] Its main components are a 12-inch woofer in an "acoustic suspension" sealed-box enclosure, a dome midrange, and a dome tweeter (*Photo 2*). The modest size, compared to other large speakers in the late fifties, offered clean bass in an unobtrusive bookshelf speaker.

When a lab colleague first asked me to repair his early model AR-3s, I was

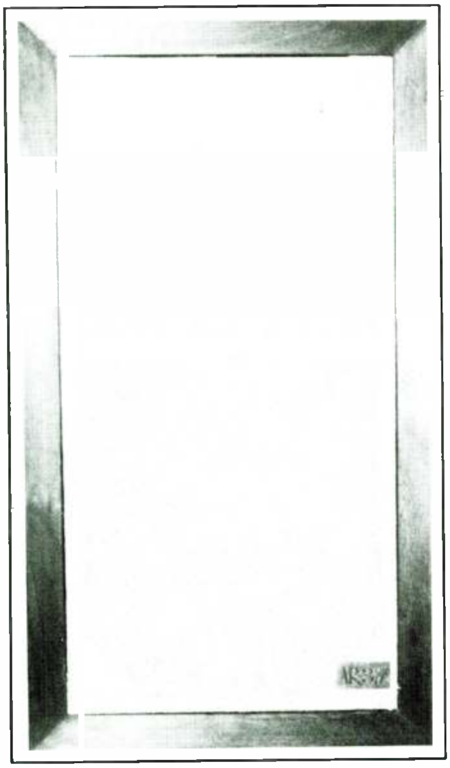


PHOTO 1: The "Classic" AR-3a speaker.

intrigued. When I was a teenager, the AR-3 was the ultimate speaker to own, but it was well beyond my limited resources. Here was my chance to finally check out its inner workings.

The 12-inch woofer was impressive with its massive die-cast frame and large alnico magnet. Certainly today a consumer speaker manufacturer could not afford to build a woofer in the same manner.

I was surprised by the effort it took to remove the tweeter and midrange. After prying them out, it was apparent why I had some difficulty. The magnets are huge! The midrange magnet weighs 5 pounds and looks like it should go on a woofer. The tweeter magnet weighs 3½ pounds.

The large magnet mass of the midrange and tweeter drivers (shown on the left in *Photos 3* and *4*) intrigued me. Villchur[6] designed the dome drivers with the suspension in the gap, so it had to be wider than normal. The wide gap required a large magnet structure to maintain efficiency. According to one of my sources, AR designers at the time did not know how to optimally size magnets. In some cases the magnets had to be demagnetized somewhat. You can see from *Photos 3* and *4*, the magnet sizes are considerably reduced in the AR-3a.

The AR-3 design tailored the frequency response, using a "shelved response" where the midrange sensitivity was set at several dB lower than the woofer, and the tweeter sensitivity was several dB lower than the midrange. The overall frequency response plot resembled a descending set of steps or shelves.

The rationale for this tailoring was the high frequency content of the records of the time did not warrant a "flat" response. This tailoring was in contrast to one AR ad that I remember which

1. Teledyne Acoustic Research, 330 Turnpike St., Canton, MA 02021.
2. D'Ascenzo, W., "The AR-1 Rejuvenated," *SB* 2/82, pp. 7-10.
3. D'Ascenzo, W., "Modifying the AR-3," *SB* Mailbox, 1/86, p. 52.
4. Villchur, Edgar M., "Revolutionary Loudspeaker and Enclosure," *Audio*, 100, Oct. 1954, pp. 25-27.
5. Augspurger, G.L., "Theory, Ingenuity, and Wishful Wizardry in Loudspeaker Design—A Half Century of Progress?" *Journal* Acoustical Society of America, April 1985, pp. 1303-08; also *Audio*, April 1987, pp. 51-55.
6. Villchur, Edgar M., "New High Frequency Speaker," *Audio*, 104, pp. 38-42.
8. Dell, E.T., "Conversation with Henry Kloss," *TAA* 3/71, pp. 4-7, 20.

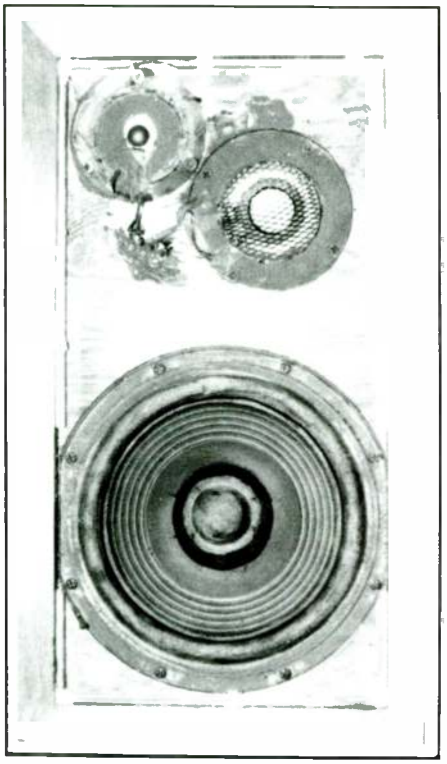


PHOTO 2: The AR-3a with grille removed.

showed a plot of the woofer, midrange, and tweeter responses all adding up to "flat" response. Another source said the individual driver responses were superimposed on each other to give an artificially "flat" response in the ad—an example of creative advertising.

MODIFYING THE AR-3. One of the many options to improve an AR-3 is to replace pots, drivers and so on with AR speaker parts (*Table 1*).

The replacement units are made at the AR plant in Canton, Massachusetts in exactly the same way as the original units. When Ed Dell and I made a tour of the plant several years ago, we saw how old style drivers were assembled in small limited production runs. In contrast, AR presently buys its drivers from Japan.

If you believe in historical restoration, you should replace the nonfunctioning drivers with AR units. However, most people aren't interested in museum pieces; they want better sound in their living rooms. So, the following sections outline a low-cost procedure for upgrading the AR-3 with a new midrange, tweeter and crossover.

DISASSEMBLY. First remove the grille cover by prying it loose with a screwdriver. Be careful because the grille frame is easily broken. In the older AR-3s, the grille sits on a groove. The later AR-3a grilles are held on by dabs of resin glue.

Next, remove the tweeter, midrange and woofer mounting screws and pry the drivers loose from the sealant putty, which should still be pliable. If you have the older model AR-3, don't be surprised at the effort it takes to pry out the midrange because of its heavy magnet. Remove all the stuffing and clip the wiring.

The existing crossover capacitors will be replaced so remove them. In the AR-3 there are several oil-filled capacitors held in by mastic and straps. In the AR-3a, the capacitors are large, tar-covered paper types held in by straps. The original AR-3a crossover board should now look like *Photo 5* with only the coils, rheostats and input posts left.

You can still use the two larger coils in the original AR-3a crossover in the new crossover. In the older AR-3, discard the coils because they are too small. The plastic coil forms are held down by adhesive and a clip. Drive a wood chisel in between the coil form and the masonite mounting board. Open up a gap, then use a small pry bar or hammer to pry the coils from the masonite. Leave the rheostats where they are. If you remove them, you will have to reseal the holes.

Finally, remove the binding posts, and note the order of all the washers and nuts. I put new solder lugs on terminals 1 and 3 and reassembled the binding posts. You may want to replace the old posts with banana jacks. The middle terminal (2) is not used in the new setup (the AR-3s had a separate terminal for woofer-only connections). I simply re-

PHOTO 3: The AR-3 midrange (L) as compared to the AR-3a midrange (R).

PHOTO 4: The AR-3 tweeter (L) and the AR-3a tweeter (R).

TABLE 1

REPLACEMENT PARTS

PART NO.	COST EST.
1210003-0 12" woofer	$73.25
1200010-1 1½" midrange	68.75
1200013-1 ¾" tweeter	32.30
1802008-0 pot	5.00

versed the mounting hardware using only the screw, nut, and washers to seal up the hole.

MODIFYING THE FRONT GRILLE. If you examine the front grille assembly, notice the front edges extend out about an inch (actually, the side of the box extends out ½," and the walnut molding extends another ½"). You can remove the walnut molding with a chisel as shown in *Photo 6*, and make a new grille frame, which I did in three AR-3 upgrades, but it amounts to several hours of work. Unless you are a skilled woodworker, I would leave the front grille as is.

If you want to make a new grille front, then remove the walnut molding by driving in a wood chisel at the glue joint at a corner. The molding should break off cleanly, however, you may have to chisel off the molding bit-by-bit. There is a secondary molding still left which I removed in past upgrades, but you should not because it adds strength to the front baffle.

TWEETER AND MIDRANGE. I replaced the tweeter and midrange units with the Audax TW-51A and the Peerless K040MRF. The total cost for all the drivers (for a stereo pair) is $50-60. The units I used were leftovers from other projects. The AR-3 woofer has a sensitivity of 90dB which fortunately matches the sensitivity rating of the Audax tweeter and Peerless midrange. You may use these units or any of the other higher grade 90dB or greater tweeters and midranges from Morel, Dynaudio and others.

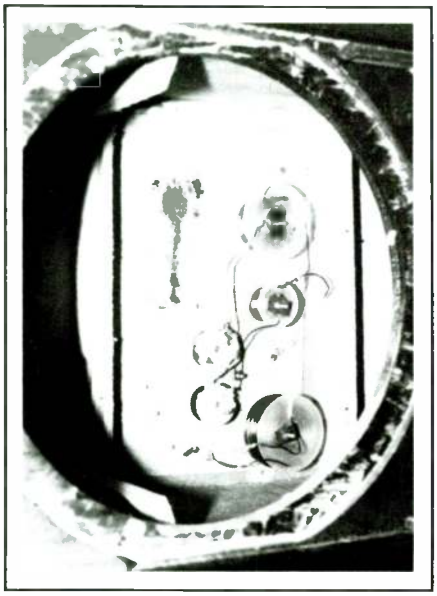


PHOTO 5: The AR-3a crossover after the woofer has been removed. L1 is the coil in the lower right hand corner and L2 is in the upper right hand corner.

PHOTO 6: The AR-3a box with drivers removed.

ADAPTER RINGS. The original holes are unfortunately larger than required for the replacement's drivers, so I made adapter rings out of scrap, ¼" tempered masonite, as shown in *Fig. 1*. I used a heavy-duty Sears Craftsman circle cutter in a drill press to cut the rings (*Photo 7*), but you can also cut out the rings with a saber saw. Be sure to clamp the material securely and wear eye protection when using the circle cutter. Also, a note of caution: a dull cutter blade will bind up in the the tough masonite.

Begin by cutting out a masonite disk for the midrange adapter ring. Measure the distance (5 ¹¹⁄₁₆" diameter) to the inner edge of the cutter blade. Also, be sure to screw or nail down the inner disk to the mounting board underneath. Allow the cutting blade to fully penetrate the mounting board for a clean-edged cut. Then with the disk still screwed to the mounting board, reset the circle cutter so the outer edge of the blade will cut a 4" diameter circle (*Photo 8*).

After cutting the ring, use the old AR-3 mid-dome as a guide and mark the location of the mounting holes. Drill and countersink ⁵⁄₁₆" holes for the flat head mounting screws. For the tweeter adapter ring, follow the same procedure (*Photo 9*) but with the diameters given in *Fig. 1*. Finally, check the rings for fit with the holes and drivers.

To seal the adapter rings I applied a thick bead of white bathroom caulk around the recessed lip, positioned the adapter ring, and screwed down the rings with the mounting screws.

I then soldered several feet of 16-gauge speaker wire to each driver, stuck Mortite caulk around the mounting flange of each driver, and screwed each driver down. (Note: The corners of the Peerless midrange stick over the edge of the adapter ring and the adapter ring is usually not flush with the cabinet surface. I used a small chisel to cut away pieces of the mounting board so that the midrange driver will lie flat.) The final driver mounting configuration is shown in *Photo 10*.

NEW CROSSOVER. I disagree with D'Ascenzo that the crossover values don't need changing. The older AR-3 had the midrange/woofer crossover point above 1kHz, because the old dome midrange could not go down further (the AR-3a's crossover point was somewhat lower). But the Peerless midrange can go below 500Hz and the woofer sounds better with a lower crossover point, so, I set the crossover point at 600Hz. The upper crossover point was set at 5kHz which matches the Peerless and Audax units quite well.

The new 6dB crossover schematic given in *Fig. 2* is based on Robert Bullock's formulation for a 4Ω woofer and 8Ω midrange and tweeter. I used a good

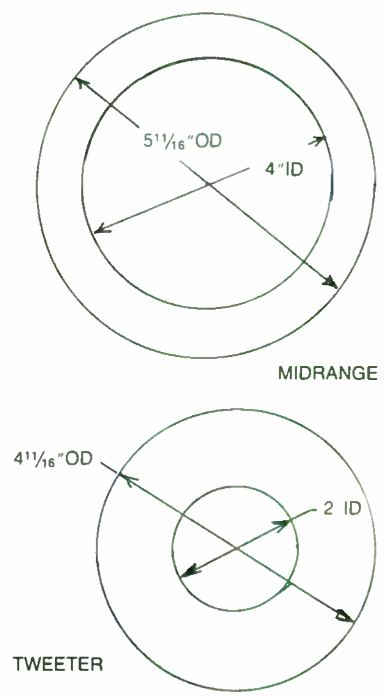


FIGURE 1: The midrange and tweeter adapter rings.

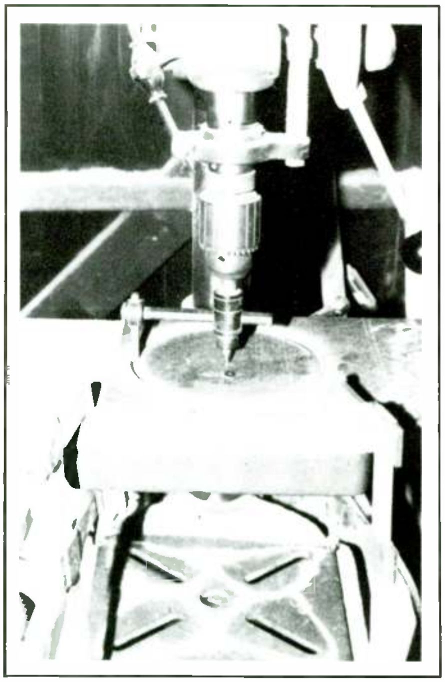


PHOTO 7: Cutting the outer diameter of the midrange adapter ring.

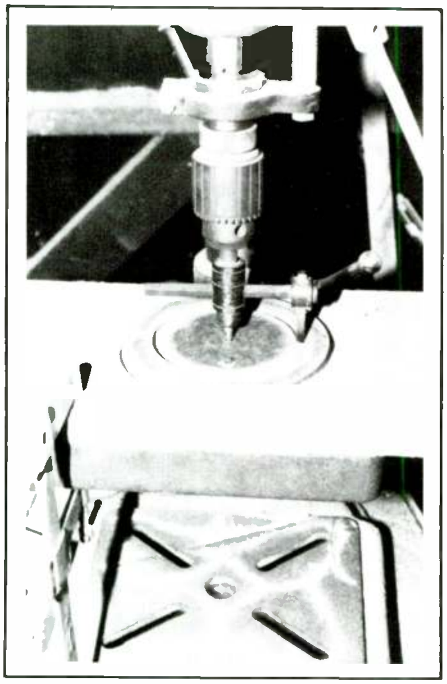


PHOTO 8: Cutting the inner diameter of the midrange ring.

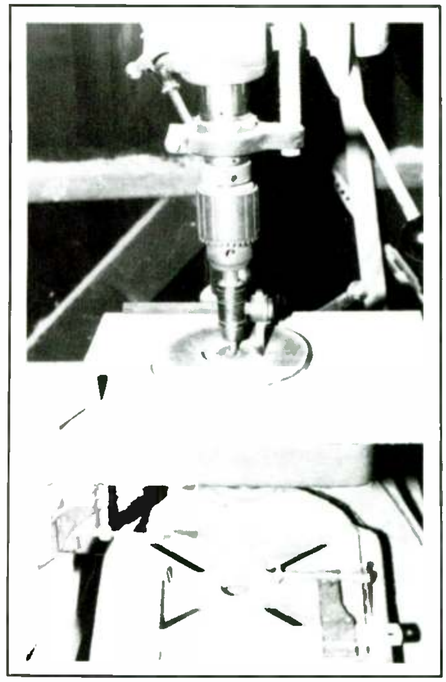


PHOTO 9: Making the tweeter adapter ring.

quality nonpolarized electrolytic for the 40μF capacitor and a Mylar for the 4μF capacitor. I constructed the crossover on 8 by 6 by ½-inch particle board and mounted it in the space the large capacitor behind the woofer had occupied.

Use the coils from the AR-3a original crossover. Take off 50 turns from the larger of the two big coils (it becomes L1). This coil originally was at 1.8mH and when modified becomes 1.1mH. The other large coil was 0.9mH. Take off 90 turns to reduce it to 0.22mH for L2. Epoxy the coil forms to the crossover board.

With the woofer still unmounted, wire all the drivers and the input terminals to the crossover. Restuff the speaker with the original stuffing or use fiberglass or Dacron batting. D'Ascenzo[2] presents some interesting alternatives for stuffing, but I was always rushed to finish the job so I've never tried them. Then stick mortite caulk on the back mounting flange and remount the woofer. All of the old caulk should have been cleaned off.

I glued ½" thick blocks in the corners as shown in *Photo 11* for the velcro fasteners. I constructed a grille frame from 14 by 25 by ½-inch particle board with cutouts and a 1½" border to affix grille cloth. *Photo 12* shows the completed AR-3 upgrade with new drivers and grille.

RESULTS. The frequency responses of an unmodified AR-3a and a modified AR-3a are plotted in *Fig. 13*. The response was taken by a hand-held Audio-Source RTA-One octave band sound

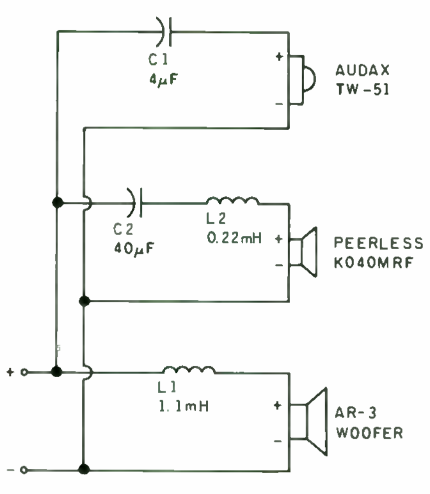


FIGURE 2: The new 6dB crossover for the modified AR-3.

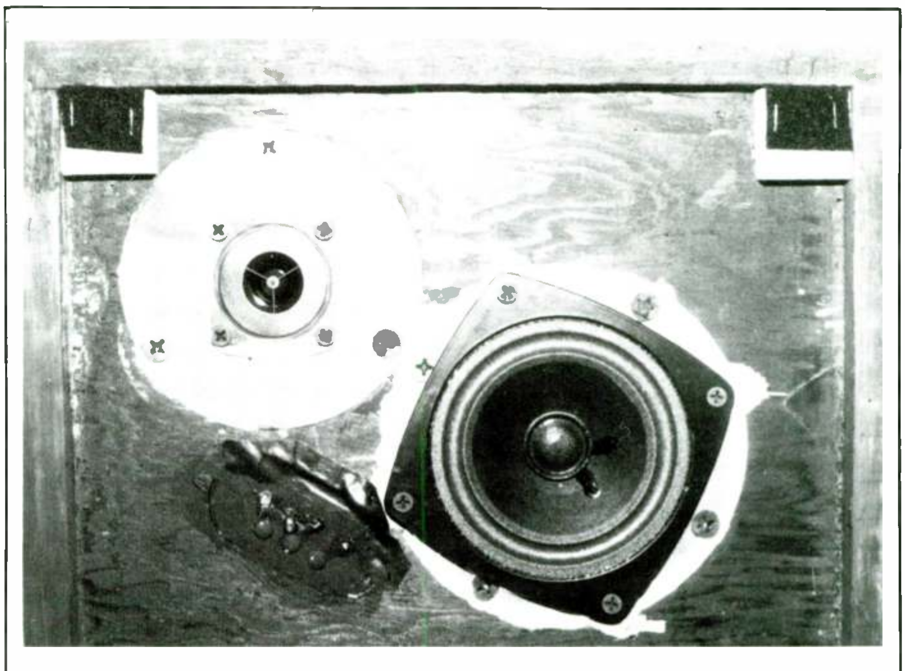


PHOTO 10: The Peerless K040MRF and Audax TW51A units installed in the AR-3 box.

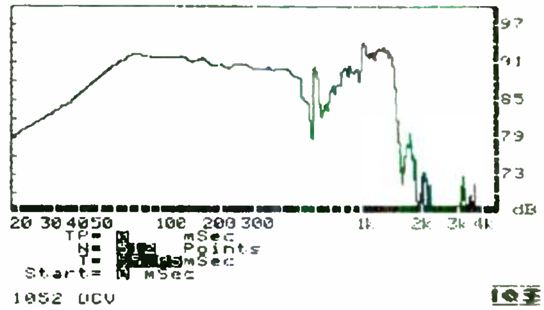
WOOFERS WOOFERS WOOFERS

MADISOUND 1052 DVC
10" Dual Voice Coil Woofer 8Ω/8Ω

Fs	21 Hz +/-2Hz
Mmd	46 Grams
Cms	1.3 X 10^{-6} CM/D
Vas	212 Liters
Rscc	5.7 Ohms
Z min	6.5 Ohms
Z max	86 Ohms
vcL	.7 mh
Qms	4.11
Qes	.29
Qts	.27
Xmax	6 mm pk
VD	204 cm^3

Surround: Foam
Magnet: 30 Oz. Ceramic
Power handling: 100 watts 50/50
Frequency response: 25-2.5K Hz
Efficiency: 91 db 1w/1m
Uses: Home or Autosound Subwoofer
Coils in parallel for 4 ohm result

Price: $34.00



MADISOUND 1252 DVC BASS REFLEX ALIGNMENTS

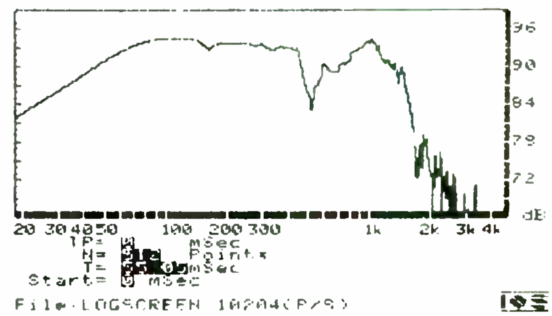
BOXVOLUME:VB	30 LITERS SEALED	56 LITERS SEALED	70 LITERS SEALED	120 LITERS VENTED	170 LITERS VENTED
BASS 1/2 PWR: F3	48.7 Hz	43.8 Hz	40.8 Hz	28.7 Hz	25.5 HZ
FILLING IN BOX	Y	Y	N		
QTC	.92	.73	.78		
PEAK AT RES: R db	+.8	.0	-.3	+1.2	+.3
BOX-VENT RES FREQ: FB		26 Hz	23 Hz		
PORT: DIAMETER Inches		3.0	3.0		
LENGTH Inches	4.5	3.8			

MADISOUND 10204 DVC
10" Dual Voice Coil Woofer 4Ω/4Ω

Fs	22.5 Hz +/-2Hz
Mmd	53 Grams
Cms	.9 X 10^{-6} CM/D
Vas	165 Liters
Rscc	3.5 Ohms
Z min	3.5 Ohms
Z max	46 Ohms
vcL	1..2 mh
Qms	3.35
Qes	.27
Qts	.25
Xmax	6 mm pk
Surround:	foam

Power handling: 200 watts 100/100
Frequency response: 30-1500 Hz
Efficiency: 90.4 db 1w/1m
Uses: Home or autosound subwoofer

Price: $40.00



MADISOUND 10204 DVC SUGGESTED ALIGNMENTS

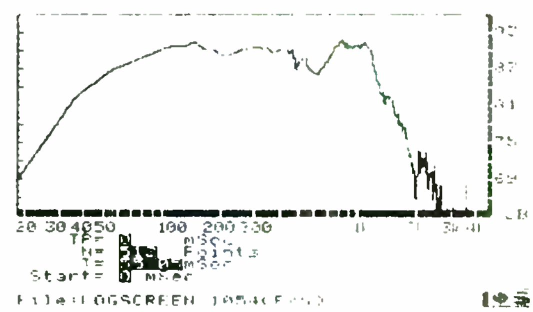
BASS VOLUME: VB	21 LITERS	28 LITERS	36 LITERS
VENTED	VENTED	VENTED	VENTED
BASS 1/2 POWER: F3	54 Hz	48 Hz	44 Hz
BOX-VENT RES FREQ:FB	40 Hz	39	36
PEAK at RES: R db	+.8 db	+.4	0
PORT: Diameter:Inches	2.5	2.5	2.5
Length: Inches	5.8	4.0	3.7

MADISOUND 1054
10" Polypropylene Woofer 8Ω

Fs	22 Hz +/-2Hz
Mmd	45 Grams
Cms	1.17 X 10^{-6} CM/D
Vas	206 Liters
Rscc	6.2 Ohms
Z min	6.9 Ohms
Z max	100 Ohms
vcL	1.5 mh
Qms	4.34
Qes	.28
Qts	.27
Xmax	3.5 mm pk
VD	119 cm^3
Surround:	foam

Magnet: 30 oz. ceramic
Power handling: 125 watts
Frequency response: 25-2.5K Hz
Efficiency: 91.7 db 1w/1m
Uses: home hi fi, autosound woofer

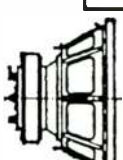
Price: $32.00 EACH



MADISOUND 1054 BASS REFLEX ALIGNMENTS

BOX VOLUME: VB	1.5 CUBIC FT. 42.5 LITERS	2.25 CUBIC FT. 63.7 LITERS	*Optimum 2.5 CUBIC FT. 72 LITERS	3.0 CUBIC FT 85 LITERS
BASS 1/2 PWR: F3	48 Hz	38 Hz	36 Hz	34 Hz
VENT RES FREQ: FB	36 Hz	32 Hz	30 Hz	29 Hz
PEAK at RES.: R	+1.7 Db	+.5 Db	+ 1 Db	-.5 Db
PORT: D diameter	D 3"	D 3"	D 3"	D 3"
L length	L 7.8"	L 6.0"	L 6.1"	L 5.3"

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, Canada and Hawaii, and those who require Blue Lable air service, please add 25%). There is no fee for packaging or handling, and we will refund to the exact shipping charge. We accept Mastercharge or Visa on mail and phone orders.



MADISOUND SPEAKER COMPONENTS
8608 UNIVERSITY GREEN
BOX 4283
MADISON, WISCONSIN 53711
PHONE (608) 831-3433

PHOTO 11: The completed modified AR-3.

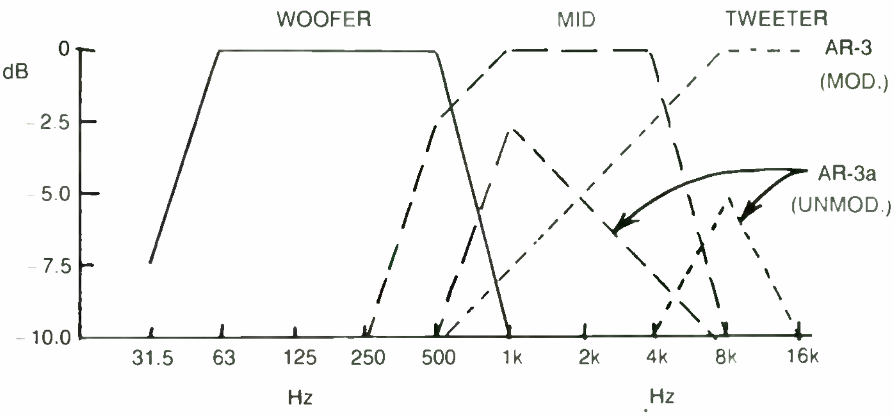


FIGURE 3: Frequency response plot of an original AR-3a versus the modified AR-3.

analyzer[7] Even though the spectrum band analysis is rather coarse, *Fig. 3* shows the shelved response of the AR-3a mentioned earlier, with the reasonably "flat" response of the AR-3 "mod." My clients were very pleased with their new sounding speakers. The modified AR-3a now has a definite balanced sound with real mids and highs instead of the recessed midrange and rolled-off highs of the original AR-3a.

There are other "mods" to try, such as replacing the original stuffing. The TW51 tweeter is sometimes "hotter" than 90dB and may require a 1 or 2Ω resistor in series with it for equalization.

Also, you may want to put foam strips around the tweeter and/or midrange to control unwanted diffraction.

Modifying and upgrading an AR-3 was an enjoyable exercise in audio "archeology." I have presented only a "low cost" upgrade, but variations are possible depending on your wallet and junk box. The further rewards from the gratitude of your friends when you give them back a better sounding speaker make all your efforts worthwhile.

ACKNOWLEDGEMENTS

I thank Don Croley, Connie O'Neill, and Al Kopania for letting me upgrade their AR-3s, Terry Schoessow for taking the photographs and Rosemary Cavy for typing the manuscript.

7. Edgar, B., "AudioSource RTA-One," *SB* 4/86, p. 39.

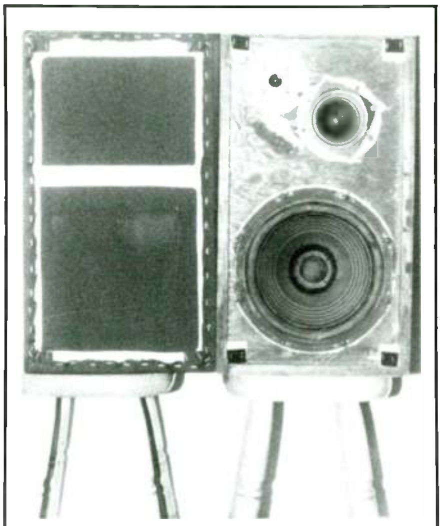


PHOTO 12: Modified AR-3 with new grille.

ELECTRONIC TIME DELAY LINE FOR SPEAKERS

BY MARK RUMREICH

Using active crossovers offers the opportunity to "delay align" the drivers in a loudspeaker without physically staggering them. Delay alignment corrects time arrival differences between drivers (due to their acoustic centers which are not equidistant from the listener's ear).

Physically staggering drivers not only complicates cabinet design, but degrades dispersion characteristics of the high frequency drivers. Also, staggering provides delay alignment on-axis only (off-axis, the drivers are no longer in a vertical line). Electronic delay provides the ultimate solution to these problems.

In this article I discuss the design and construction of a line suitable for loudspeaker delay alignment. I'll also examine setting the proper delays for your system.

A delay line for delay alignment of loudspeakers should:

- Have flat amplitude and delay response from DC to 20 kHz;
- Introduce no audible noise or distortion;
- Be capable of time delays in the hundreds of microseconds (1 inch corresponds to approximately 75μsec).

For additional convenience and flexibility, it should also provide:

- Selective delay with adequate resolution;
- Unity gain and low output impedance regardless of delay selected;
- Multiple outputs of different delay if desired (for a three-way system for example).

Analog delay lines suitable for high fidelity audio are generally either charge-coupled devices (CCDs) or delay filters. Because of noise limitations associated with CCDs (an approach typically used for electric guitar effects), I ruled out this possibility. To avoid the many disadvantages of inductors, I chose an active filter approach.

FILTERS. Cascaded Bessel (maximally flat delay) filters is a common approach to delay networks. A similar filter type, *linear phase with equiripple error*, provides more delay and greater delay bandwidth for the same number of stages.

Unfortunately, both filters are low-pass, and their amplitude response rolls off where the delay is still flat. These filters require amplitude compensation at high frequencies *to take advantage of their delay bandwidth.*

For example, to produce a 200μsec delay (with 20kHz delay bandwidth) would require approximately[1] (10) fourth-order networks, (6) sixth-order networks, or (4) eighth-order networks.

Because each filter's amplitude response rolls off at a lower frequency than 20kHz, it would be necessary to compensate through the use of an amplitude "peaking" circuit. The amount of amplitude boost at 20kHz required (with respect to low frequencies) is: For fourth-order networks, 4dB per stage = 40dB total; 7dB per stage = 42dB total for sixth-order networks; or 14dB per stage = 56dB total for eighth-order networks.

Using amplitude peaking in turn affects the delay flatness, but more importantly adds filter stages. As we will see shortly, the filter complexity required for low-pass filters is excessive compared to all-pass filters.

ALL-PASS FILTERS. An alternative to the low-pass delay filters' problem of amplitude rolloff at high frequencies, is a filter type known as *all-pass* which has the property of flat amplitude response for all frequencies. I thought I could design an all-pass filter to approximate a time delay and therefore eliminate the problems previously discussed.

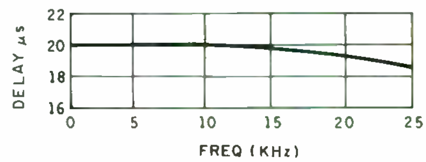


FIGURE 1: Simulation of 20μsec delay: Pade approximation (C = 0.68).

To meet the requirements for "selectable delay with adequate resolution" and "time delays in the hundreds of microseconds" I decided to cascade a number of identical small delay elements for the best results. Through numerous computer simulations, I opted for a "tapped" delay line composed of 16 identical delay sections of 20μsec each, providing a maximum delay of 320μsec (4.3 inches). Each section has unity gain and low output impedance to eliminate compensating for gain when selecting delay.

PADE APPROXIMATION. Pade developed a synthesis procedure for an all-pass approximation to a pure time delay.[2] The Pade approximation matches the maximum number of terms of the Taylor's series expansion of the mathematical equivalent of a time delay.

To provide 20μsec of delay per stage, I considered the second-order Pade approximation. The results of the computer simulation of this are shown in *Fig. 1.*

1. Zverev, Anatol I., *Handbook of Filter Synthesis*, John Wiley & Sons, 1967, pp. 92-97.
2. Roberge, James K., *Operational Amplifiers: Theory and Practice*, John Wiley & Sons, 1975, pp. 530-35.

A MODIFICATION. Although the Pade approximation appeared acceptable for this application, I wanted to be sure it was the optimum second-order filter. Tradeoffs generally exist in filters between passband ripple and bandwidth, and the second-order Pade approximation exhibits no ripple.

Constraining myself to second-order, all-pass networks with 20μsec of delay (at DC), I defined a *shaping factor*, C, which allows adjustment of the appropriate coefficient ratios in the transfer function. The defining equation for the shaping factor, C, is:

$$H(s) = \frac{Cs^2 - (4/T)s + (8/T^2)}{Cs^2 + (4/T)s + (8/T^2)}$$

where T is the time delay.

This allowed me to experiment with reoptimizing the delay response. *Figures 2* and *3* show delay curves for C values of 0.75 and 0.85. (The Pade approximation corresponds to a C of 0.68.) The plots show that increasing C adds high frequency delay peaking, which extends the delay bandwidth at the expense of delay flatness.

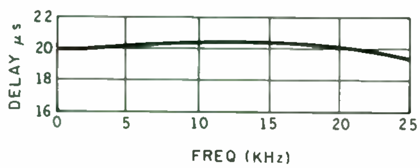


FIGURE 2: Simulation of 20μsec delay: modified Pade approximation (C = 0.75): 1.5% peaking; cutoff, 20kHz.

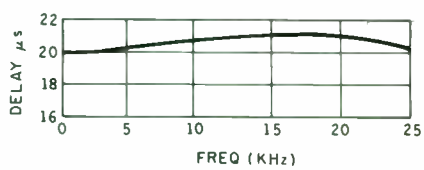


FIGURE 3: Simulation of 20μsec delay: modified Pade approximation (C = 0.85): 5% peaking; cutoff, 27kHz.

IMPLEMENTATION. My first breadboard design used a C = 0.75 value. I chose this response as a good tradeoff between delay peaking (1.5%) and delay bandwidth (20kHz). I used the Steffen all-pass filter topology[3] shown in *Fig. 4*. This circuit exhibits low component sensitivities and provides unity gain and low output impedance for a very reasonable parts count.

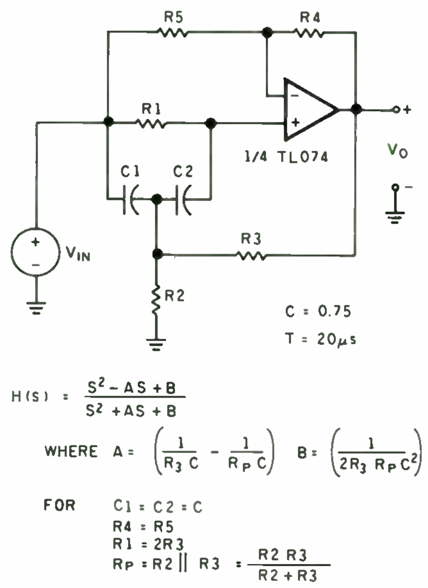


FIGURE 4: Steffen all-pass filter for 20μsec delay (C = 0.75); see parts list for values.

My first attempt was a 1MHz oscillator. The performance emphasized the importance of stray and parasitic capacitance. A couple of applicable rules of thumb:

- Keep the impedance level of circuits low to minimize noise pickup and the effects of stray capacitance;
- Consider what your circuit looks like at *all* frequencies, not just the "range of use."

I reduced the circuit's impedance to a "reasonable minimum," which alleviated stability and noise problems; however the delay rolloff was premature and the amplitude response also began to roll off at 20kHz (theoretically it should remain flat). Experiments showed this behavior was due to stray capacitance and finite op amp gain. Fortunately, the delay peaking provided by the C = 0.75 circuit reduced the objectionability of the delay rolloff.

ACTUAL PERFORMANCE. *Figure 4* shows one stage of the circuit used to realize this response. *Figure 5* shows the measured delay (lower trace: 20μsec/div) and amplitude response (upper trace: 0.5dB/div) of *four* delay sections cascaded.

If *all 320μsec* of delay are used, the amplitude response will have approximately 0.5dB of high frequency accentuation. The delay can be expected to deviate less than 5% from the DC value to 18kHz. I measured the signal-to-noise ratio to be 86dB (1V RMS, 20kHz bandwidth, unweighted). *Table 1* shows distortion for various signal levels.

CONSTRUCTION. I constructed a circuit board for a stereo version of the delay line. *Figure 6* shows the copper pattern and *Fig. 7*, the parts placement. *Photo 1* is the completed board.

I used TL074 quad op amps because of their low noise, wide bandwidth, high input impedance, low distortion and high slew rate. The circuit board provides for 10Ω resistors and 1μF tantalum caps for ± supply decoupling of each IC. Point-to-point wiring interconnects the stages.

3. Daryanani, Gobind, *Principles of Active Network Synthesis and Design*, John Wiley & Sons, 1975, pp. 296-97.

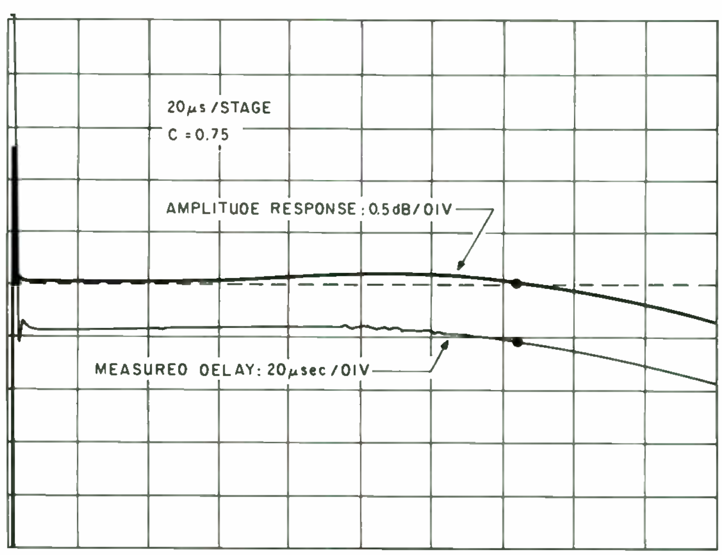


FIGURE 5: Measured response of four cascaded 20μsec-delay sections (C = 0.75).

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 •

1982

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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 •

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 •

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 •

1986

The Edgar Midrange Horn • Sand-Filled Stands • Crossover Networks: Passive and Active • 5-Sided Boxes • A 2×4 Transmission Line • The Free-Volume Subwoofer • Notch Filters • By-Wiring the LS3/5A • A Push-Pull Constant Pressure System • Current and Power in Crossover Components • The Unbox (Egg) • Upgrade Speakerlab's S-6 Crossover • Measure Speakers with Step Response • A Gold Ribbon System • A Visit with Ken Kantor • A Tractrix Horn Design Program • Reviews: Audio Concepts "G;" Seven T.L. Midranges; Focal's Model 280; the Audio Source RTA-ONE.•

1987

A Compact TL Woofer • Frequency Response and Loudspeaker Modeling (three parts) • A Manual Coil Winder • The Model-One Speaker • Designing a Listening Room • A Six-Order Vented Woofer • Tapered Pipe Experiments • Visiting Boston Acoustics • A Vented Compound System • The Octaline • Spreadsheets for Speaker Design • In Memoriam: Richard Heyser (two parts) • Using Non-Optimum Vented Boxes • Building Speaker Stands • Evaluating Driver Impedance Compensation • Tuning Bass-Reflex • Six Woofers Compared • Bullock on Passive Crossovers: Alternate Bandpass Types • Fast, Easy Filter Calculations • A Mobile Speaker • Modifying the Polk 10

TABLE I (Distortion 1kHz)	
Signal Level	**THD**
1V RMS	0.04%
2V RMS	0.07%
3V RMS	0.10%

C1 and C2 are Class I ceramic disks. Ceramic disks in this range could be Class I or Class II. Class I caps are temperature compensated (NPO, N330, and so on), and use a higher quality dielectric than Class II (Z5P, Z5U, and so on). The higher quality dielectric provides significantly higher Q and stability at the expense of cost and size. Class II caps may have Qs as low as 25 while Class I caps are usually higher than 500 (audio frequencies). Class II caps "reform" when soldered. This causes the capacitance to increase as much as 20%, then drift toward the original value over a period of weeks. If you are unsure of Class, use silver mica caps.

The $1\mu F$ tantalum decoupling caps may be replaced with $10\mu F$ electrolytics, in parallel with $0.01\mu F$ ceramic disks, if you find tantalum costs exceed your budget.

The ±15V power supply should be well regulated and capable of ±100mA. LM7815 and LM7915 regulators are well suited to this application.

To avoid blowing up TL074s, check that pin 4 of each IC is connected to the +15V supply and pin 11 to the −15V supply before turn-on. Once the unit is powered-up, check each stage for proper operation. Apply a 25kHz sine wave to each input and check that its output is approximately 180° out of phase (inverted) and about the same amplitude as the input.

COMPONENT SENSITIVITIES. Component tolerances affect the *high-frequency amplitude response* of the delay line. (The Steffen topology guarantees unity gain at DC.) This becomes particularly important for long delay lines where many stages are cascaded. Rather than specify precision components, I recommend the following:

- Sort 470pF caps in 2% (10pF) wide bins. Use two caps from the same bin for each stage (C1 and C2).
- Sort 10k resistors in 1% (100Ω) wide bins. Use two resistors from the same bin for each stage (R4 and R5).
- After assembly, tweak each stage toward unity gain at 18kHz *from the input of the delay line to the output of that stage* (see *Table 2*).

TABLE II		
18kHz Output (dB)	**Add Across R1**	**Add Across R3**
< −.4	—	270k
−.2	—	560k
0	—	—
.2	1.5M	—
.4	560k	—
.6	390k	—
>.8	270k	—

For matched C1 and C2, a 10% increase in capacitance will result in a 10% increase in delay and a 10% decrease in stage bandwidth. This is useful to know for "fine-tuning" the delay of a stage or for redesigning the delay line for wider bandwidth or longer delay per stage.

INSTALLATION. *Figure 8* shows the preferred installation location for the delay line. Delay sections for midrange and tweeter can be shared and is superior from a noise standpoint. For example, AC power line hum picked up by the delay line cables would be filtered out for midrange and tweeter. For some applications (such as subtractive cross-

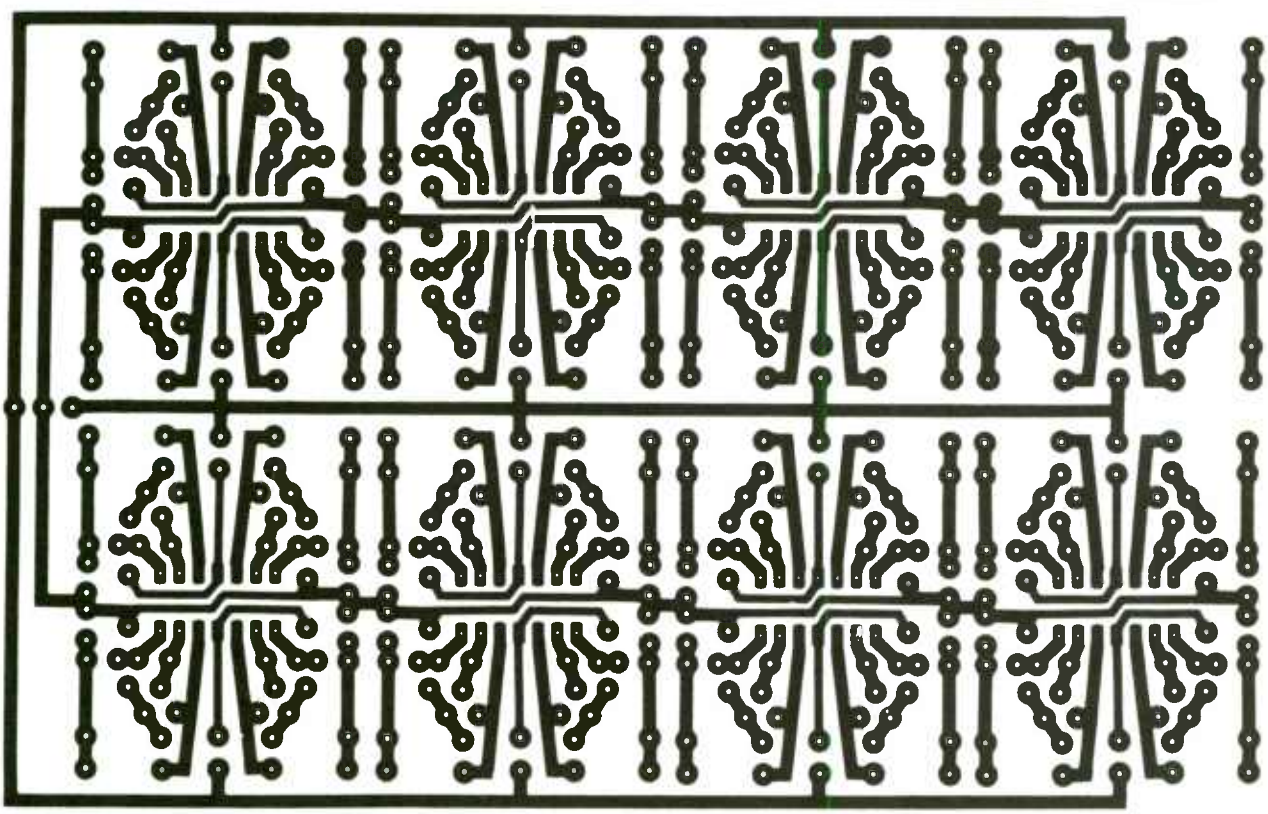

FIGURE 6: Circuit board for two channels of 320μsec maximum delay.

overs) the crossover must precede the delay line. Three-way systems of this type can be accommodated by "splitting up" the line if the total delay does not exceed 320μsec. This approach is shown in *Fig. 9*.

SYSTEM ALIGNMENT. Setting the proper delays to achieve delay alignment is somewhat difficult, particularly without sophisticated test equipment. If acoustic center measurements are available from the manufacturer, take advantage of them.

The method I eventually arrived at to determine acoustic centers is tedious, but simpler methods do not consider errors caused by reflections and phase-response ripples in speaker and/or microphone. In my approach I used on-hand equipment: a lab grade microphone, dual trace scope, audio generator and frequency counter.

The test microphone must have linear phase response over the test range. The oscilloscope requires X-Y capabilities to compare the phase of two signals using Lissajous patterns. I strongly recommend a frequency counter—1% error in frequency could result in a 10% error in acoustic center measurement, due to the multiplication effect of the measurement distance to delay distance.

ACOUSTIC CENTER. Follow this procedure and measure for each driver individually. *Figure 10* shows my test equipment and setup.

1. Place the microphone *exactly* 6″ from the *baffle* front, on-axis with the driver. I selected 6″ to provide useful data points for each driver without introducing large errors due to uncertain mike placement. Any distance in this range is acceptable, but it must be *identical for all drivers tested.*

If the acoustic center of the mike element is not coincident with its tip, the error will be the same for all measurements, and does not invalidate this procedure for taking relative measurements. Make sure the arrangement does not reflect sound from the mike stand to the driver (I stuck the mike element through a funnel). Adjust the audio generator level to provide a clean signal at the microphone output, and be careful not to overdrive the tweeters. Adjust the scope sensitivity (X and Y) to roughly "fill the display."

2. For a range of frequencies covering the useful range of the driver, vary the frequency of the signal generator and record all frequencies where the phase of the signal from the microphone matches that at the driver input. This condition produces a scope display (Lissajous pattern) which is a straight line with positive slope. Also record all frequencies where the phase of the signal from the mike is 180° from that at the driver input. (This produces a Lissajous pattern of a straight line with negative slope.) Taking a large number of points improves the certainty of the result. It is important to have a complete list of these points *in increasing order of frequency*. Precise measurements are also important—try to record frequencies to the nearest hertz.

Table 3 is the data for a midrange driver I tested. The first column is the complete list of in-phase and out-of-phase frequencies.

3. The frequencies recorded satisfy the property that the distance between the acoustic center of the driver and the mike is an exact integer multiple of one-half wavelength. The integer multiple of one-half wavelength which corresponds to a particular frequency is easy to predict, if you know the polarity of the microphone (and speaker) and the distance between the acoustic centers of the driver and microphone to within a half wavelength. I will assume that this may not be the case. (This will require iteratively assigning half integers to each frequency until the results are consistent).

It is now necessary to assign integer multiples of half wavelengths to each frequency in the list. Start by checking that the frequencies in the list are all separated by roughly the same amount (approximately 1kHz for the 6″ mike distance). With the possible exception of the first two points, the frequencies in *Table 3* satisfy this requirement. Delete

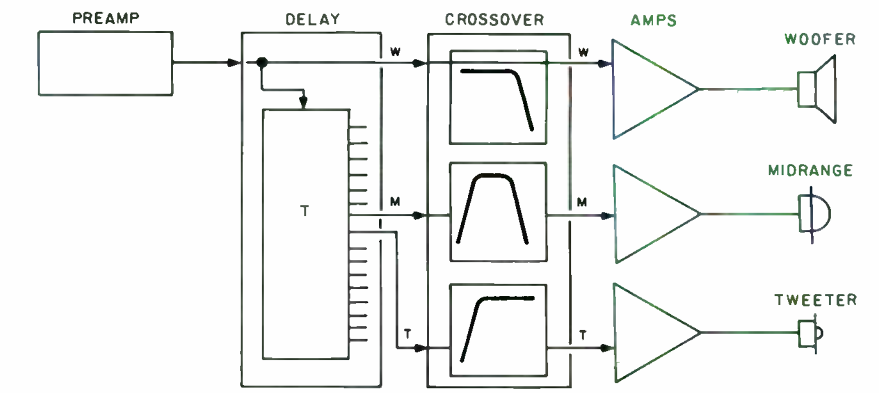

FIGURE 8: Preferred installation location of delay line.

TABLE III

	1st Assign.		2nd Assign.	
Freq. (Hz)	**N**	**Delay (μsec)**	**N**	**Delay (μsec)**
461	.5	1085	0	—
1151	1.0	869	.5	434
1975	1.5	759	1.0	506
2908	2.0	688	1.5	503
3951	2.5	633	2.5	506
4848	3.0	619	2.5	516

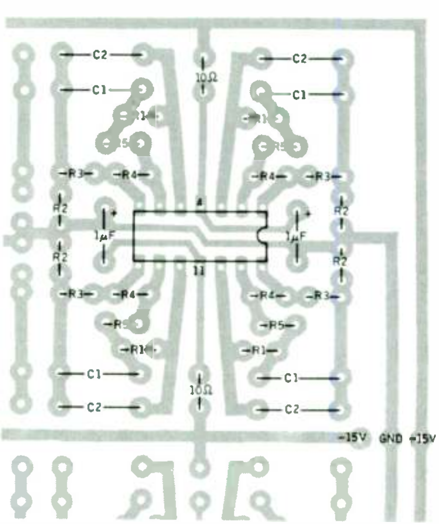

FIGURE 7: Parts placement.

PARTS LIST

The parts required for the fully loaded board (2 channels of 320μsec max delay) are given.

Qty.	Ref.	Desc.
32	R1	33k
32	R2	8.2k
32	R3	16k
64	R4, 5	10k
16	R decoup.	10Ω
64	C1, 2	470pF
16	C decoup.	1μF tant.
8	IC	TL074

All resistors ¼W, 5% (prior to sorting—see text).

entries which don't follow this pattern, and write "missing entry" where there appears to be a gap. (Spurious data can result from phase shifts in the speaker or microphone response.) Assign the lowest entry 0.5, the next entry 1.0, the next 1.5, and so on. (see the second column of *Table 3*.)

4. Assuming the assignments made in step 3 are correct, the relative delay between speaker acoustic center and mike acoustic center can now be calculated. For each entry in the list, calculate N/f where N is the half integer assigned in step 3 (0.5, 1.0, and so on), and f is the frequency (in Hz). N/f is the delay in seconds. The third column of *Table 3* shows the calculated delays in μsec.

After calculating the delay for each entry, look down the list and check that the results are about the same (discard entries which are not within 10%), and take the average as the delay. If the results show an *increasing trend* down the list of more than 5–10% between entries, the assignments made in step 3 are incorrect. Reassign values of N, *adding* 0.5 to each entry, and repeat step 4. If the results show a *decreasing trend* down the list, reassign values of N, *subtracting* 0.5 from each entry, and repeat step 4.

Examine the data in *Table 3*, the first assigned delays showed a strong decreasing tendency, therefore the second assignments of N were 0.5 lower. The resulting delay calculations are consistent. Discarding the first valid entry, I consider the delay to be 506μsec.

To determine the time delays required for delay alignment, you do not need to calculate the location of each driver's acoustic center. If you are interested however, the distance between the acoustic center of the driver and the acoustic center of the microphone is the speed of sound multiplied by the delay. Using 13,500 inches per second as the speed of sound, and the delay in seconds, you can calculate the distance. Assuming the acoustic center of the microphone is at its tip, the acoustic center of the speaker is this distance *minus 6″* (or the distance between the mike and baffle if 6″ was not used) *behind* the baffle. (For the test driver, I calculated 0.84 inches).

Perform this procedure for each driver. The difference between the woofer and midrange results is the midrange relative advance, and is the time delay you should add to the midrange signal for delay alignment with respect to the baffle. The difference between the woofer and tweeter results is the tweeter relative advance, and is the time delay you should add to the tweeter signal for delay alignment with respect to the baffle.

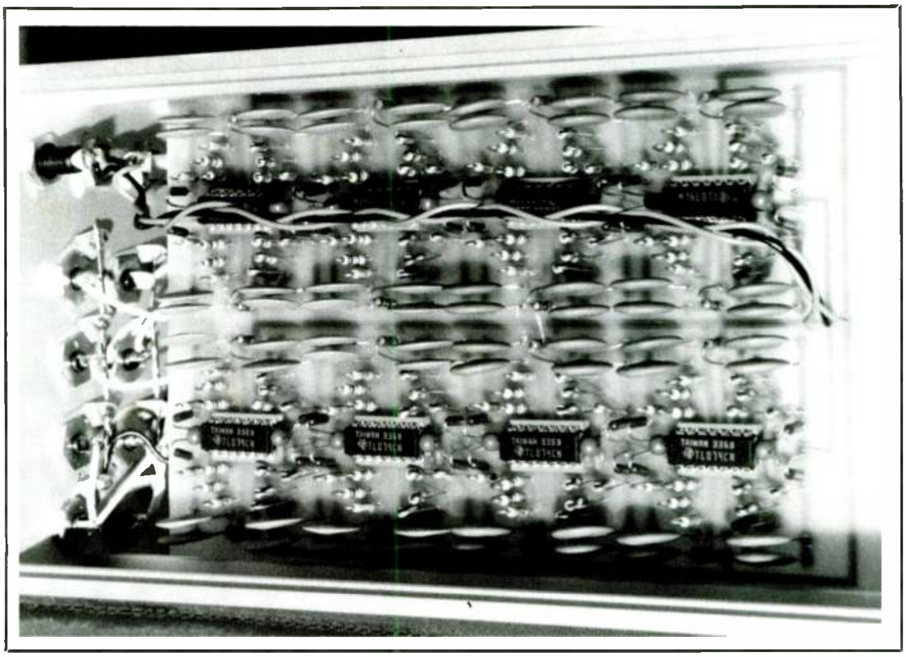

PHOTO 1: Completed circuit board.

CORRECTION FACTOR. When delay aligning a system you should choose the optimum listening position. The standard assumption is delay alignment should be with respect to the plane of the speaker front, which is equivalent to assuming that the listener is infinitely far from the speaker and on-axis. The approach I recommend instead, is to select a point in space which is your typical listening position, and select time delays for the

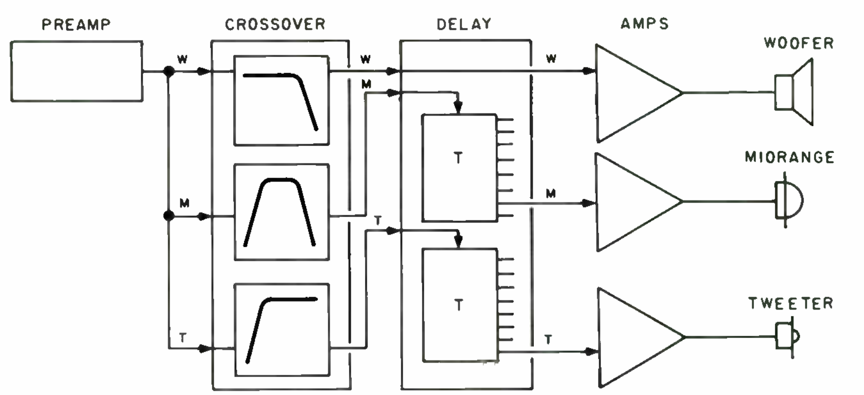

FIGURE 9: Alternate installation location of delay line.

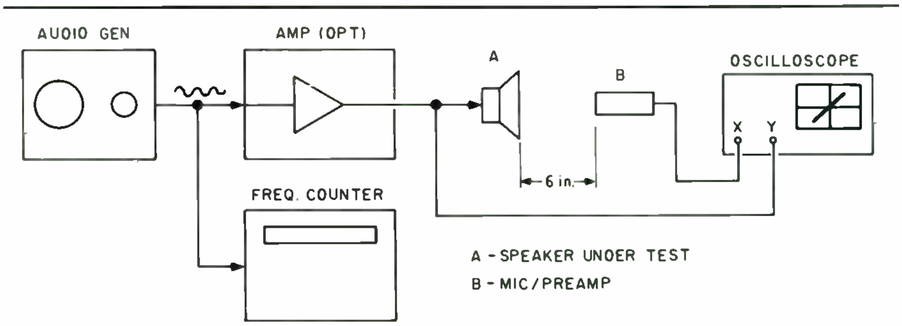

FIGURE 10: Setup for measuring relative acoustic centers of drivers.

midrange and tweeter which provide acoustic coincidence at that point for all three drivers.

To understand the significance of the difference between the two approaches, consider the simple case of drivers with acoustic centers the same distance behind their respective mounting plates. The first approach would say the drivers should be mounted on a flat baffle, while the second would say the drivers should be located on a circle (sphere) with the origin at the listening spot.

To implement the second approach, I use time delays for the midrange and tweeter based on the relative acoustic center measurements, modified by the *spherical correction factor*. You can easily calculate this correction factor using trigonometry.

HEAR THE DIFFERENCE? After the sizable expenditure of time and effort to delay align the system, the obvious question is, "What's the audible difference?" To answer this question, I built a "demo switch." This switch selectively bypasses the delay lines used for delay alignment while I sit at my normal listening position.

I checked the validity of the experiment by listening to each driver individually; there was no audible difference between demo switch positions A and B.

For the experiments, I used only one side (R) of my system. Realizing the difference between the two modes would be most audible for material near the woofer crossover region, I passed pink noise through a 750Hz (half-octave wide) band-pass filter as a test signal. The most obvious difference I heard with the delay line was the actual change in tonal quality of the noise. This makes sense when you consider that time mismatch creates "comb filtering effects," but really says little about audible improvement or degradation in imaging. An interesting effect is the perceived location of the noise source. Using the delay, the sound appeared to come from a point in space. With the delays absent, the sound source elongated into a "tube" from the speaker to behind the speaker. I am surprised this effect is so apparent, particularly with the test signal's random nature.

I replaced the pink noise with music and listened to both speakers. The differences are more subtle. For much material, I could not hear the difference between the two modes, however, for music with dominant horns or vocals the imaging was noticeably improved.

I should add that for the purposes of this experiment, I used a first-order crossover. This is important because for ideal drivers, the on-axis response is "transient perfect." I would expect the audible benefits of delay compensation to diminish as transient response becomes less ideal (through crossover selection and listening position). As most modern crossovers sacrifice transient response for amplitude flatness, cutoff rate and polar response, I suggest you look for designs which do not.

For example, the "delay derived" constant-voltage crossover[4] provides optimum amplitude and transient response as well as steep slopes and exceptional polar response (Bessel alignments). This filter requires a delay line (on the order of 300μsec), making it impractical to implement passively. A drawback of this approach is the high sensitivity of the high-pass response to delay and subtraction errors. I believe post-filtering the high-pass output below approximately $f_c/4$ would be a remedy.

Continued on page 67

4. Lipshitz, Stanley P. and John Vanderkooy, "A Family of Linear-Phase Crossover Networks of High Slope Derived by Time Delay," *JAES*, Jan./Feb. 1983, Vol. 1, pp. 2-20.

ENCLOSURE SHAPES AND VOLUMES

BY LUBOS R. PALOUNEK

If you want to design a naturally sounding speaker, start with an enclosure that has theoretically correct volume and shape.

The proper internal volume depends on selected drivers and alignment[1]. For a given driver and alignment, there is only one correct internal volume.

An enclosure with a specific volume may have many different shapes. Some shapes result in a much better sounding loudspeaker system, compared to other shapes. The enclosure's height, depth and width should be selected so the standing waves inside the cabinet are evenly spread throughout the spectrum. Wall panel resonances' undesirable effects should also be minimized.

You should also consider how tilting the enclosure affects sound. A properly tilted front panel helps achieve phase coherency of the sound coming from two or more drivers, and may help reduce the effect of unwanted reflections—sound waves from the enclosure back wall which reach the listener through the driver cone. The speaker cabinet dimensions must certainly be selected so we can mount the drivers on the front panel, have sufficient depth for the vent (if we are building a vented enclosure), etc. Also, the enclosure must fit its place within the room, physically and aesthetically.

Speaker design is a combination of science and art. Many different shapes might satisfy the requirements mentioned above. This article presents a method which helps you select the best shape for a specific set of circumstances.

I will consider only enclosures for dynamic drivers in closed or vented designs in the following examples. However, the approach may be adapted for other designs, such as transmission line enclosures.

Asymmetrical braces can reduce sound coloration, by dividing the panel into uneven parts. You might want to include in your designs braces placed in optimal positions; these are not shown in the following examples.

HANDBOOK. Your personal handbook of enclosure shapes and dimensions is a quite useful and timesaving tool. My handbook includes the dimensions of elementary enclosure shapes and some more complicated favorites. All enclosures' dimensions have identical (normalized) volume.

The handbook makes the selection of an enclosure shape which best fits specific requirements a relatively easy process. The handbook makes it easier to select a shape which takes advantage of the beneficial effects of narrow cabinets, vertical in-line driver arrangement, and irregular shapes. (An irregular shape lowers sound coloration because of the asymmetry of the internal air space and individual wall panels.)

The selection of the enclosure dimensions becomes a streamlined, two-part process: First, I calculate the required volume and the factor by which the needed enclosure differs from the normalized enclosures in the handbook. Second, I find out which shapes in my handbook best satisfy all other requirements and multiply the dimensions by the "volume factor," to obtain the actual enclosure dimensions.

I like to make small scale models of my favorite shapes in the handbook, all having the same volume; to help visualize the final enclosures. Because these models have normalized volume, they can be reused later, when I design another system.

The speaker design process is often iterative. After going through the design and evaluation of different shapes, you may find the theoretically correct enclosures are unsuitable for one reason or another; in that case, go back to driver catalogs to try different drivers or alignments. The handbook of shapes and dimensions makes this process less time consuming, and more efficient and enjoyable.

The *Tables* and *Figures* shown on the following pages are samples from my handbook. The *Tables* show dimensions of many enclosures having two basic

1. See many articles on Thiele/Small alignment in past *SB* issues, other periodicals, and several books on loudspeaker design. Loudspeaker design programs for personal computers (which greatly reduce the time required to do the Thiele-Small alignment calculations) are available from several sources.

The "LoudSpeaker" program for IBM personal computers created by *Speaker Builder* author Max Knittel has many useful features. We may select different values of the enclosure damping loss Ls. The program not only calculates the theoretically correct volume for different alignments, but also calculates the frequency response of a driver mounted in an enclosure that has a given volume. It lets us compare several designs. Up to six low-end frequency response curves are shown on one graph. This is very helpful when we want, for example, to evaluate the effect of a driver unit which has actual parameters that are at the limit of the tolerances specified by the manufacturer. The program displays or prints the expected frequency response of such a driver mounted in an enclosure having volume designed for the nominal parameters. The programs calculate other needed values, such as the vent dimensions for allowable vent air velocity and comes with a helpful library of driver parameters which we can change and add additional drivers to.

The latest version of the LoudSpeaker program is available for $39.95 from Maximum Effort Software, 2701 Cedarwood Avenue, Bellingham, Washington 98225. When ordering, specify whether your Personal Computer does (or does not) have an 8087 math coprocessor. Both versions of the program are equally powerful; the version which takes advantage of the math coprocessor works noticeably faster.

shapes: a simple rectangular box and an enclosure with tilted front. As stated above, all enclosures have the same, normalized volume, and their shape satisfies at least some of the theoretical requirements.

STANDING WAVES. The fundamental requirement for a rectangular enclosure dictates that standing waves inside the enclosure are evenly distributed through the spectrum; you do not want standing waves created between different walls to have the same frequency and therefore reinforce each other.

To spread standing waves within a rectangular enclosure[2] evenly through the relevant part of the sound spectrum, the ratio[3] of the three primary internal dimensions (width, depth, and height) should be based on $2^{1/3} = 1.2599$.

Rectangular enclosure dimensions which satisfy this requirement have the ratio: 1 to 1.2599 to 1.5874.

Notice[4] that $1.2599 \times 1.2599 = 1.5874$. The dimension ratios of that enclosure may also be expressed as 0.7937 to 1 to 1.2599. Notice that $0.7937 \times 1.2599 = 1$.

A speaker enclosure based on this shape can be built six ways, shown in *Figs. 1* and *2*, depending on the orientation of the three dimensions. The simple rectangular enclosure is shown in *Fig. 1*. The slanted front enclosures, derived from the rectangular shape, are shown in *Fig. 2*. All enclosures in *Figs. 1* and *2* have identical volumes of $1 \times 1.26 \times 1.59 = 2.00$.

To satisfy the requirement of spreading standing wave frequencies evenly, any of the dimensions may be an integral multiple of the recommended dimension. For example, a ratio of 1 to 1.2599 to 3.1748 satisfies the theoretical requirement, because $2 \times 1.5874 = 3.1748$. The shapes which have the width, depth or height multiplied by two are called "double wide," "double deep," or "double high," and abbreviated "dw," "dd," and "dh" in *Table 1*. The single dimensions are abbreviated "sw," "sd," and "sh." Enclosures which incorporate triple dimensions, or higher multiples, are not included in these examples.

You can design enclosures having the elementary rectangular shape using the six configurations from *Fig. 1*, and you can double one or two of the shape's dimensions; this leads to 42 variations of each shape. Each variation can be built as a simple rectangular enclosure, as shown in *Fig. 1*, or with a slanted panel.

I chose an 18.4° slant in these designs for two reasons (the precise value of the angle is 18.43494882°). First, many

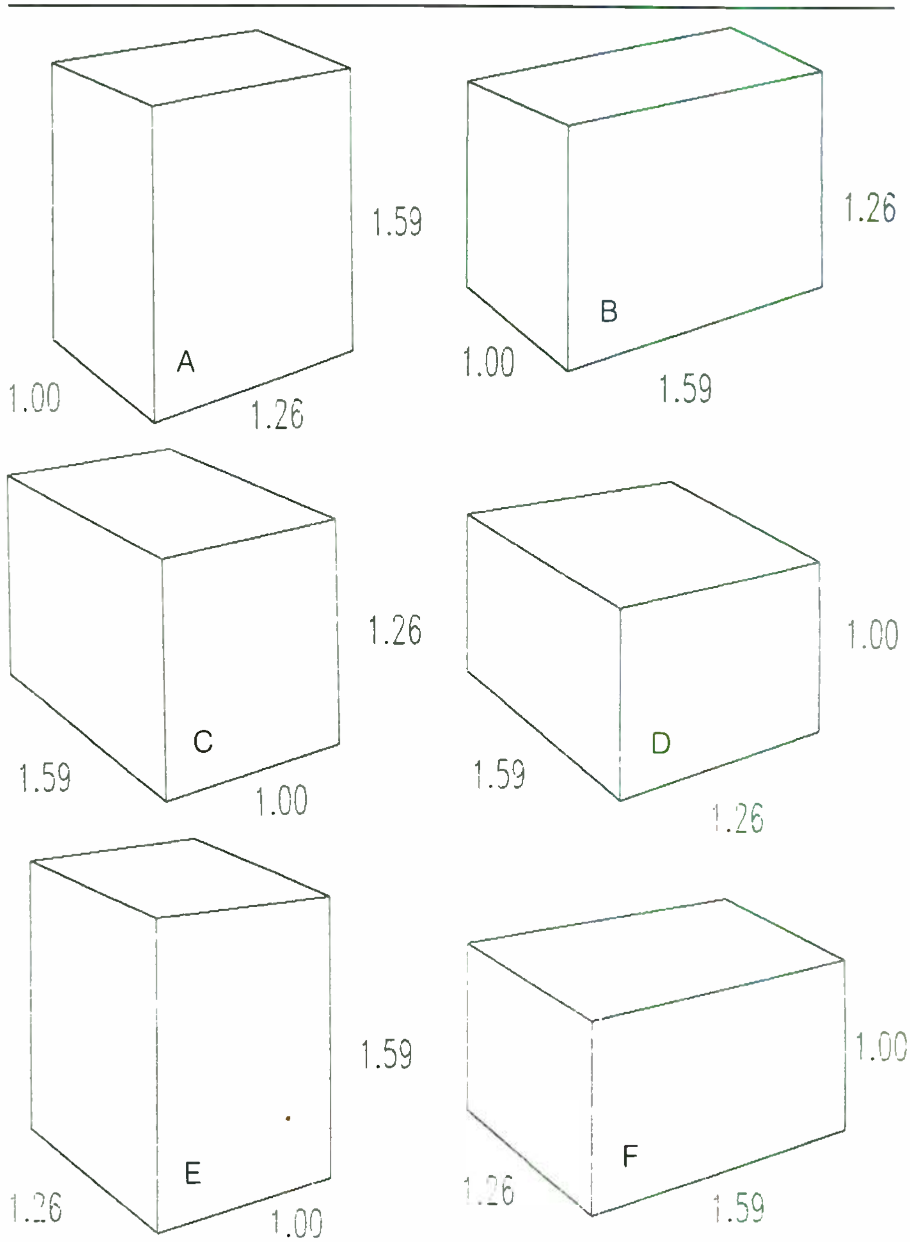

FIGURE 1: Simple rectangular enclosures. The drivers are mounted on the panel shown on the left side of the drawing.

2. Resonant frequencies of a rectangular enclosure with dimensions of W × D × H are based on the Rayleigh equation:

$$f_r = c/2 \sqrt{(n_w/W)^2 + (n_d/D)^2 + (n_h/H)^2}$$

where n_w, n_d, and n_h are integers 0, 1, 2, 3 ... for the corresponding resonance modes and c is the speed of sound. The effect of these resonant frequencies on sound coloration is minimized when they are evenly spread through the relevant part of the spectrum.

Both rectangular speaker enclosures and listening rooms are subjected to the same physical laws; their acoustical behavior is often studied in a very similar way.

Many recommend you base dimensions of rectangular enclosures on the ratio of 1:1.2599. Another approach is described in the article "A New Criterion for the Distribution of Normal Room Modes," by Oscar Juan Bonello, *JAES*, Vol. 29, No. 9, 1981 September, pp. 597-606.

3. This ratio of 1 to 1.2599 to 1.587 must not be confused with the ancient Golden Mean ratio, which states that the most aesthetically pleasing "golden mean" rectangle has the ratio of 1 to 1.618 between its shorter and longer dimensions. The precise value of the Golden Mean ratio is $(1 + \sqrt{5}) \div 2$, and is based on the requirement that the ratio of the shorter side to the longer side is equal to the ratio of the longer side to the sum of the shorter plus longer sides.

This "golden ratio" is often used for overall dimensions of an object and for parts of the total object. As the "acoustical" ratio shown in the main text refers to internal dimensions, and the aesthetically oriented "golden ratio" refers to external dimensions, it is sometimes possible to design an enclosure which satisfies both of these objectives. These ratios, while given to many decimal places for accuracy, are rounded for practical purposes.

driver pairs work best on a panel tilted 18-20°. Second, the tangent of this angle is one-third. This simplifies the layout of panels and joints, as this angle corresponds to the 1:3 slope. Therefore, this angle is a very good starting point for most designs.

Table 1 shows dimensions for the rectangular enclosures from *Fig. 1*, and tilted front panel enclosures from *Fig. 2*. Row 1's first three columns (w, d, and h) show the rounded value[4] of the fundamental width, depth, and height of a rectangular enclosure which satisfies the rules for optimization of standing waves. The next two columns (d top and d bottom) show the top and bottom depth of an enclosure with a front panel slanted at 18.4°. The front panel column shows the length of the slanted front panel.

Enclosures described by Row 1 do not have normalized volume. Row 2 shows the dimensions of the normalized enclosures, which have a volume of one cubic unit. If you multiply the normalized width, depth and height of an enclosure, you get volume of one. For example, look at the enclosure described by the first line: $0.793701 \times 1 \times 1.259921 = 1$.

Now let's find out how we can use a properly selected constant to convert the normalized dimensions shown in *Table 1* to the actual internal dimensions of an enclosure.

ALIGNMENT. The selection of the enclosure type, such as closed box or vented box, or the selection of drivers, is outside the scope of this article (see footnote 1). Let's assume you have already selected the drivers and alignment.

INTERNAL VOLUME. Next, you must establish the correct internal volume of the enclosure. You can pick up the value specified in driver catalogs, or calculate the volume using methods described in past *SB* issues, or published in several speaker design books. Computer programs greatly simplify the design calculations for some cases when the parameters (such as the enclosure damping loss L_s) change.

Remember, add the additional space used by items within the enclosure to the required volume, including:

- Driver(s);
- Vent (if a vented enclosure);
- Crossover (if you plan to mount it internally);
- Bracing, enforcements, subenclosures and supports;
- Damping material (bituminous impregnated felt pads and so on) if any.

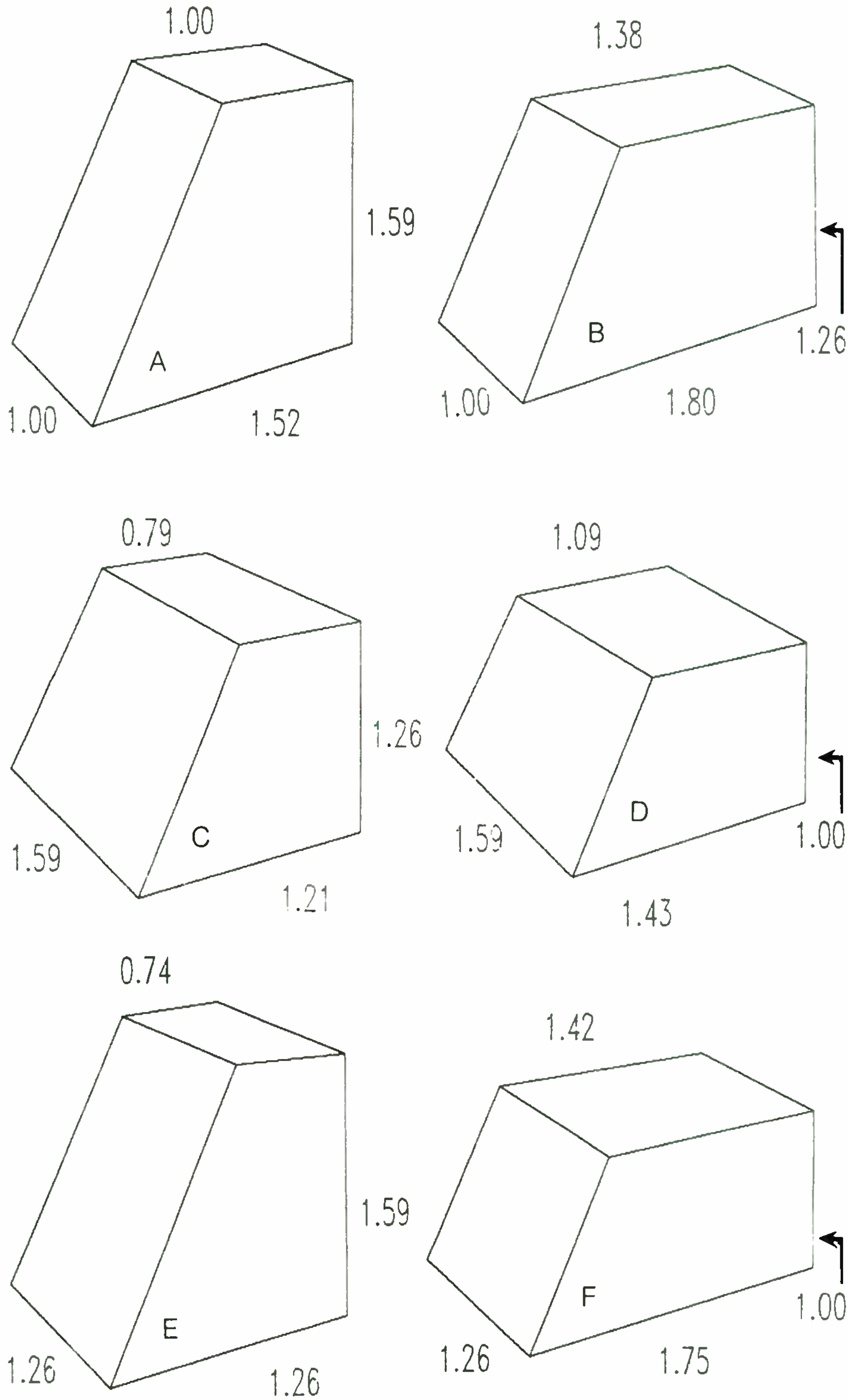

FIGURE 2: Slanted-front enclosures. The drivers are mounted on the panel shown on the left side of the drawing.

Uncompressed, loosely packed wool or similar filling material inside the enclosure does not change the effective internal volume. Dense packing noticeably *increases* the effective volume.

You might want to add to the theoretically established volume to allow for future fine tuning of the sound. You can reduce the effective internal volume by adding stones or other material later (through the vent, or using a removable

Continued on page 26

4. The values in the table were calculated to eight decimal places, and then rounded. Therefore, the values sometimes do not directly "add up".

TABLE 1

ROW 1 **ROW 2**

ENCLOSURE SHAPE	w	d	h	d top	d bot.	front panel	NORMALIZED width	depth center	height	depth top	depth bottom	front panel
A sw-sd-sh	1.00	1.26	1.59	1.00	1.52	1.67	0.793701	1.000000	1.259921	0.790013	1.209987	1.328073
sw-sd-dh	1.00	1.26	3.17	0.73	1.79	3.35	0.629961	0.793701	2.000000	0.460367	1.127034	2.108185
sw-dd-sh	1.00	2.52	1.59	2.26	2.78	1.67	0.629961	1.587401	1.000000	1.420734	1.754068	1.054093
sw-dd-dh	1.00	2.52	3.17	1.99	3.05	3.35	0.500000	1.259921	1.587401	0.995354	1.524488	1.673268
dw-sd-sh	2.00	1.26	1.59	1.00	1.52	1.67	1.259921	0.793701	1.000000	0.627034	0.960367	1.054093
dw-sd-dh	2.00	1.26	3.17	0.73	1.79	3.35	1.000000	0.629961	1.587401	0.365394	0.894527	1.673268
dw-dd-sh	2.00	2.52	1.59	2.26	2.78	1.67	1.000000	1.259921	0.793701	1.127638	1.392205	0.836634
B sw-sd-sh	1.00	1.59	1.26	1.38	1.80	1.33	0.793701	1.259921	1.000000	1.093254	1.426588	1.054093
sw-sd-dh	1.00	1.59	2.52	1.17	2.01	2.66	0.629961	1.000000	1.587401	0.735433	1.264567	1.673268
sw-dd-sh	1.00	3.17	1.26	2.96	3.38	1.33	0.629961	2.000000	0.793701	1.867717	2.132283	0.836634
* sw-dd-dh	1.00	3.17	2.52	2.75	3.59	2.66	0.500000	1.587401	1.259921	1.377414	1.797388	1.328073
dw-sd-sh	2.00	1.59	1.26	1.38	1.80	1.33	1.259921	1.000000	0.793701	0.867717	1.132283	0.836634
dw-sd-dh	2.00	1.59	2.52	1.17	2.01	2.66	1.000000	0.793701	1.259921	0.583714	1.003687	1.328073
dw-dd-sh	2.00	3.17	1.26	2.96	3.38	1.33	1.000000	1.587401	0.629961	1.482408	1.692395	0.664037
C sw-sd-sh	1.59	1.00	1.26	0.79	1.21	1.33	1.259921	0.793701	1.000000	0.627034	0.960367	1.054093
sw-sd-dh	1.59	1.00	2.52	0.58	1.42	2.66	1.000000	0.629961	1.587401	0.365394	0.894527	1.673268
sw-dd-sh	1.59	2.00	1.26	1.79	2.21	1.33	1.000000	1.259921	0.793701	1.127638	1.392204	0.836634
sw-dd-dh	1.59	2.00	2.52	1.58	2.42	2.66	0.793701	1.000000	1.259921	0.790013	1.209987	1.328073
dw-sd-sh	3.17	1.00	1.26	0.79	1.21	1.33	2.000000	0.629961	0.793701	0.497677	0.762244	0.836634
dw-sd-dh	3.17	1.00	2.52	0.58	1.42	2.66	1.587401	0.500000	1.259921	0.290013	0.709987	1.328073
dw-dd-sh	3.17	2.00	1.26	1.79	2.21	1.33	1.587401	1.000000	0.629961	0.895007	1.104993	0.664037
D sw-sd-sh	1.59	1.26	1.00	1.09	1.43	1.05	1.259921	1.000000	0.793701	0.867717	1.132283	0.836634
sw-sd-dh	1.59	1.26	2.00	0.93	1.59	2.11	1.000000	0.793701	1.259921	0.583714	1.003687	1.328073
sw-dd-sh	1.59	2.52	1.00	2.35	2.69	1.05	1.000000	1.587401	0.629961	1.482408	1.692394	0.664037
sw-dd-dh	1.59	2.52	2.00	2.19	2.85	2.11	0.793701	1.259921	1.000000	1.093254	1.426588	1.054093
dw-sd-sh	3.17	1.26	1.00	1.09	1.43	1.05	2.000000	0.793701	0.629961	0.688707	0.898694	0.664037
dw-sd-dh	3.17	1.26	2.00	0.93	1.59	2.11	1.587401	0.629961	1.000000	0.463294	0.796627	1.054093
dw-dd-sh	3.17	2.52	1.00	2.35	2.69	1.05	1.587401	1.259921	0.500000	1.176588	1.343254	0.527046
E sw-sd-sh	1.26	1.00	1.59	0.74	1.26	1.67	1.000000	0.793701	1.259921	0.583714	1.003687	1.328073
sw-sd-dh	1.26	1.00	3.17	0.47	1.53	3.35	0.793701	0.629961	2.000000	0.296627	0.963294	2.108185
sw-dd-sh	1.26	2.00	1.59	1.74	2.26	1.67	0.793701	1.259921	1.000000	1.093254	1.426588	1.054093
sw-dd-dh	1.26	2.00	3.17	1.47	2.53	3.35	0.629961	1.000000	1.587401	0.735433	1.264567	1.673268
dw-sd-sh	2.52	1.00	1.59	0.74	1.26	1.67	1.587401	0.629961	1.000000	0.463294	0.796627	1.054093
dw-sd-dh	2.52	1.00	3.17	0.47	1.53	3.35	1.259921	0.500000	1.587401	0.235433	0.764567	1.673268
dw-dd-sh	2.52	2.00	1.59	1.74	2.26	1.67	1.259921	1.000000	0.793701	0.867717	1.132283	0.836634
F sw-sd-sh	1.26	1.59	1.00	1.42	1.75	1.05	1.000000	1.259921	0.793701	1.127638	1.392205	0.836634
sw-sd-dh	1.26	1.59	2.00	1.25	1.92	2.11	0.793701	1.000000	1.259921	0.790013	1.209987	1.328073
sw-dd-sh	1.26	3.17	1.00	3.01	3.34	1.05	0.793701	2.000000	0.629961	1.895007	2.104993	0.664037
sw-dd-dh	1.26	3.17	2.00	2.84	3.51	2.11	0.629961	1.587401	1.000000	1.420734	1.754068	1.054093
dw-sd-sh	2.52	1.59	1.00	1.42	1.75	1.05	1.587401	1.000000	0.629961	0.895007	1.104993	0.664037
dw-sd-dh	2.52	1.59	2.00	1.25	1.92	2.11	1.259921	0.793701	1.000000	0.627034	0.960367	1.054093
dw-dd-sh	2.52	3.17	1.00	3.01	3.34	1.05	1.259921	1.587401	0.500000	1.504068	1.670734	0.527046

Continued from page 27

panel). To increase the volume 16%, the linear dimensions should be increased 5%; to increase the volume by one-third, the linear dimensions should be increased 10%.

Now, let's assume you selected a bass driver and alignment which needs a total internal volume (including all the "extras") of 108 liters.

LINEAR FACTOR. The linear factor equals (total required volume)$^{1/3}$. It can be calculated on a scientific pocket calculator, or estimated from *Table 2*, Volume and Dimension Ratios.

Table 1 shows internal dimensions of normalized enclosures; each has an internal volume of one cubic unit. For the real system, you need an enclosure which has a volume of n cubic units. To change the volume of the enclosures shown in the tables by a factor of n, we must change the linear dimensions shown in the table by $n^{1/3}$. In other words, to increase the volume from one cubic unit to n cubic units, we must multiply the linear dimensions (length, width and depth) by third root of n.

It does not matter which measurement units are used, but be consistent. If the volume is expressed in cubic feet, then choose linear dimensions in feet. For volume in cubic inches, make the linear dimensions inches; For liters (cubic decimeters), make the linear dimensions decimeters (1dm = 10cm = 100mm).

For example, if the required internal volume, including additional space for drivers, braces, and so on, is 108 liters, the linear factor equals:

$$108^{1/3} = 4.762203155$$

or approximately 4.7622. As the volume is expressed in liters (cubic decimeters), the linear dimensions are in decimeters.

For a detailed discussion of the relationship between changes in linear dimensions and changes in volume, or the *linear factor* concept, read this article's sidebar, "Volume and Dimension Ratios."

FIND DIMENSIONS LIMIT. Some enclosure dimensions might have practical limitations; make a note of these limitations, using the correct unit. Convert the known limitations to the normalized dimension: (normalized dimension) = (actual dimension) ÷ (linear factor).

Let's say, to continue our example, you plan to use a driver which needs a front panel (internally) larger than 210mm. The actual internal width and height of the front panel must be over 2.1dm; otherwise, the driver would not physically fit. Thus, the actual dimension limit equals 2.1dm.

Now calculate the minimum normalized internal width and height of the front panel, using the linear factor calculated above (4.7622). Normalized dimension = 2.1 ÷ 4.7622 = 0.441.

For our example, we must select the height and width of normalized shapes which have a front panel more than 0.441 normalized units.

SELECT A SHAPE. Next, we select a shape from the entries in our handbook of shapes and dimensions. We must pay attention to the limitations established above, and to other design considerations.

Lets assume you want to use a narrow, slanted front panel, to minimize the refraction effects from the cabinet front edges; and want a deep cabinet to reduce the effect of the reflected sound waves from the back wall, which reach the listener through the speaker cone.

Now, we look for an enclosure just slightly wider than 0.441 normalized units which satisfies all our requirements. From *Table 1* we select a normalized enclosure with the internal dimensions shown in *Table 1a*.

TABLE 2

VOLUME	DIMENSION	VOLUME	DIMENSION
1.00	1.000 000	46.42	3.593 812
1.10	1.032 497	51.09	3.710 601
1.21	1.066 050	56.23	3.831 185
1.33	1.100 694	61.90	3.955 688
1.47	1.136 464	68.13	4.084 237
1.62	1.173 395	74.99	4.216 963
1.78	1.211 528	82.54	4.354 003
1.96	1.250 899	90.85	4.495 495
2.15	1.291 550	100.00	4.641 586
2.37	1.333 521	110.07	4.792 425
2.61	1.376 857	121.15	4.948 165
2.87	1.421 601	133.35	5.108 967
3.16	1.467 799	146.78	5.274 994
3.48	1.515 498	161.56	5.446 417
3.83	1.564 748	177.83	5.623 410
4.22	1.615 598	195.73	5.806 155
4.64	1.668 100	215.44	5.994 839
5.11	1.722 309	237.14	6.189 654
5.62	1.778 279	261.02	6.390 801
6.19	1.836 068	287.30	6.598 484
6.81	1.895 735	316.23	6.812 916
7.50	1.957 341	348.07	7.034 317
8.25	2.020 949	383.12	7.262 913
9.09	2.086 625	421.70	7.498 937
10.00	2.154 434	464.16	7.742 631
11.01	2.224 447	510.90	7.994 245
12.12	2.296 735	562.34	8.254 036
13.34	2.371 373	618.96	8.522 269
14.68	2.448 436	681.29	8.799 219
16.16	2.528 003	749.89	9.085 169
17.78	2.610 156	825.40	9.380 412
19.57	2.694 979	908.52	9.685 249
21.54	2.782 558	1000.00	10.000 000
23.71	2.872 984		
26.10	2.966 348		
28.73	3.062 746		
31.62	3.162 276		
34.81	3.265 042		
38.31	3.371 146		
42.17	3.480 699		

TABLE 1a

NORMALIZED DIMENSIONS

internal width	=	0.5000000
internal center depth	=	1.587401
internal height	=	1.259921
internal top depth	=	1.377414
internal bottom depth	=	1.797388
front panel	=	1.328073

This is a single wide, double deep and double high version of *Fig. 2*'s tilted front shape B. The slanted "normalized front panel" is shown in *Fig. 3b*, while the simpler rectangular enclosure is shown in Figure 3a. Note that both enclosures in *Figures 3a* and *3b* have identical volume.

You need to multiply the normalized dimensions shown above by the linear factor 4.7622 to obtain the actual internal dimensions, and by 100 to convert from dm to mm. Therefore, multiply the numbers shown above by 476.22 and obtain the actual internal dimensions in mm (see *Table 1b*).

TABLE 1b

ACTUAL DIMENSIONS

		mm	inches
internal width	=	238.11	9.37
internal center depth	=	755.95	29.76
internal height	=	600.00	23.62
internal top depth	=	655.95	25.82
internal bottom depth	=	855.95	33.70
front panel	=	632.46	24.90

The front of the enclosure shown in *Fig. 3b* (and all tilted front enclosures in *Table 1*) is tilted 18.4°, which is an angle suitable for many pairs of woofer/midrange and tweeter drivers. However, we must verify that this angle is suitable for our drivers.

As a check, calculate the internal volume of the enclosure. In our example, we convert millimeters to decimeters and calculate 2.3811 × 7.5595 times 6 = 107.9996, which is very close to the

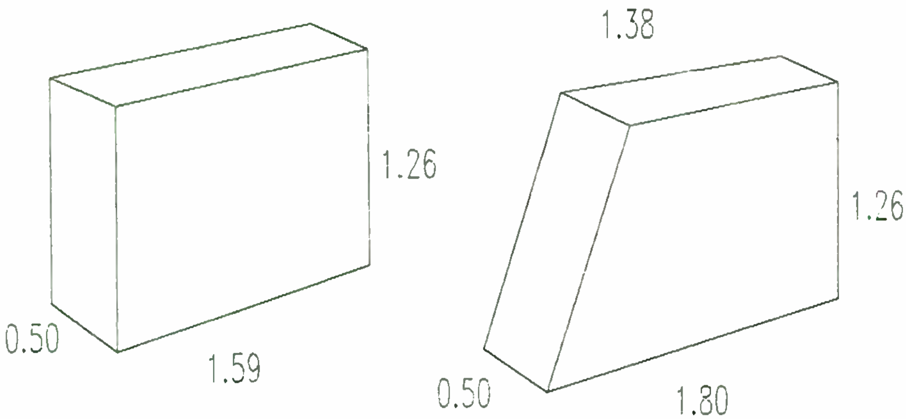

FIGURE 3: Two enclosures of identical volume.

divides the square into two unequal sections might lead to an acceptable solution.

CONCLUSION. This article describes a notebook of normalized dimensions for different enclosure shapes and illustrates its use. Such a notebook helps us select the optimal enclosure dimensions. The tables given for many variations of the two fundamental shapes can be directly used during the enclosure design. Each user can add his or her favorite shapes to the handbook.

Speaker building is a combination of science, art, and craftsmanship. After you design the enclosure, you must build it with utmost attention to good bracing and well-built joints.

correct value of 108 liters. The difference is caused by rounding.

Depending on the thickness of the material and type of joints you plan to use, calculate the external dimensions and the size of all individual panels, and then round the dimensions. You should always do the calculations with precise numbers (many decimal places) and round the final result; this eliminates accumulated rounding errors.

Be sure the design does not violate some design practices. For example, the top panel shown in *Fig. 2a* and in the first line of *Table 1* is almost square, which is the worst shape for an enclosure panel, as second-order panel (plate) resonances occur at simultaneous pairs. This could result in pronounced sound coloration. Such a shape might be acceptable with proper bracing. A diagonal brace or an asymmetrical brace which

VOLUME AND DIMENSION RATIOS

Figure 4 shows a "one unit" cube. If we increase the dimensions to three units, the volume increases

$$3 \times 3 \times 3 = 3^3 = 27 \text{ times.}$$

When sides of a three-dimensional object change by a factor of a, the volume changes by a factor of $a \times a \times a = a^3$.

Similarly, when we want change the volume of an object by a factor of n, we we must change the linear dimensions by $n^{1/3}$. For example, to change the volume from 27 cubic units to 1 cubic unit, or by a factor of 1/27, we must multiply the linear dimensions (length, width and depth) by third root of 1/27, or by the linear factor $(1/27)^{1/3} = 0.33333$.

In other words, if we want to change the volume of the large cube from *Fig. 4*, which has a volume of 27, to a cube of volume 1, (change the volume by a factor of 1/27), we must multiply its linear dimensions by 0.33333.

Table 2 shows the relationship between volume and linear dimensions, as volume is increased by 10% increments. If, for example, we want to

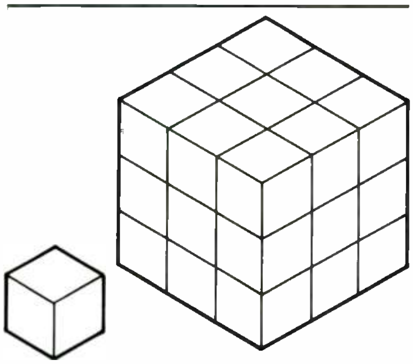

FIGURE 4: Increasing a cube's dimensions three times equals an increase in volume 27 times.

decrease the volume by a factor of 27, we can interpolate:

26.10–2.966358
28.73–3.062746

A volume change by a factor of 27 corresponds to the change of linear dimensions by a factor of 3.

Use column 1 of *Table 2* for the required volume and column 2 shows the corresponding linear factor.

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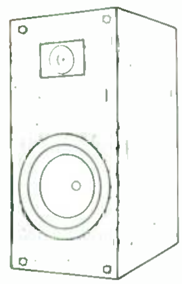
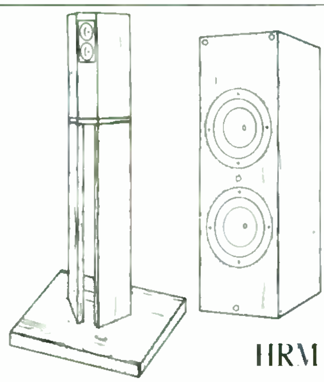


HRM



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David Davenport reporting in SPEAKER BUILDER MAGAZINE issue 2/88.

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Volume 1 / Number 6

SAMPLE ISSUE

April 1988

VOICE COIL

THE NEWSLETTER FOR THE LOUDSPEAKER INDUSTRY

Industry News

James Winey, President of Magnepan, Inc., has announced the settlement of a lawsuit filed by Magnepan against Apogee Acoustics for infringement of several basic patents on planar-magnetic and ribbon transducers held by Magnepan. During a pretrial conference conducted by Judge Mark Wolf, agreement was reached which will allow Apogee to continue selling its product in the present form in exchange for paying an undisclosed license fee to Magnepan for past and future sales.

Oaktron Manufacturing, a division of Oaktron Industries, 704 30th Ave., Monroe WI 53566, is reported to have filed for Chapter 11 protection from its creditors in early April. A company spokesman says this is a "temporary maneuver" and they "expect to be out of Chapter 11 in three to four months."

The headquarters address for Audax Industries of France (formerly Societe Audax) has been changed. The new address is: Audax Industries, Peripole 243, 33 Av. Du Marechal De Lattre De Tassigny, 94127 Fontenay Sous Bois, France. Tel 48-76-6161 Telex 262-830 Fax 48-77-2638.

Polydax Corporation of America, remains the exclusive US distributor for Audax products, and is the company agent for all business.

The Festival du Son (Festival of Sound), the French equivalent to our CES (and maybe the *Stereophile* show also, since it is open to the public part of the time) was held April 7-12. The May issue of *Voice Coil* will carry items of interest.

Precise Acoustical Laboratories, a division of Onkyo USA, will begin shipping its newly introduced Keith Johnson-designed line of loudspeakers, on May 3rd, as scheduled.

Production and Research Test Equipment

Even in this era of computer FFT analyzers, a good sweep generator and a chart recorder are still a viable choice for obtaining loudspeaker frequency magnitude measurements for design and quality control purposes. Dr. Floyd Toole of Canada's National Research Council, and others, make a good argument for averaged on- and off-axis free-field measurements (anechoic or outdoor half-space and ground plane) taken from about two meters as providing the best information for design purposes (JAES, May 1986). At any rate, few chart recorders are now available from which to choose. One of the more interesting ones is made by Neutrik.

US Neutrik product distribution has undergone some changes over the last few years, but Neutrik USA was recently formed to handle all Neutrik products.

The Neutrik Audiograph 3300, a modular chart recorder consists of the 3302 mainframe unit, which is the paper and writing system. Add to it the 3312 input module, with balanced and unbalanced inputs, and the the 3322 output module to form a minimum unit. The output module will do continuous sine sweeps, ⅓-octave stepped sine sweeps, or ⅛-, ⅓-, and ½-octave warble tone sweeps (equivalent to pink noise measurements). Other modules include the 3332 Phase module, with a built-in delay control for doing acoustic phase measurements; the 3314 Tracking Receive Filter module, for selective frequency measurements; the 3323 Compressor module, which is an amplifier capable of maintaining constant level, current, and power for use in doing impedance measurements; the 3324 Noise Generator/Tracking Send Filter module can be used for reverb time measurements; the 3335 Frequency Expanding Display module, which can be used to increase graphic detail of normal chart sections; the 3360 Synchro module, which allows the unit to be automatically engaged by a test tape or record; and the Audiograph Analyzer 3337 which a digitally programmable oscillator and distortion analyzer.

Prices for the 3302, 3312 and 3322 modules are $1,800, $650, and $500 respectively, making the cost of the basic unit about $2,950. The 3332 Phase module is priced at $1,300, and the 3323 Compressor module goes for $1,200. Neutrik also makes two excellent and reasonably priced measurement microphones. The unbalanced version is priced at $180, while the balanced model sells for $215. For more information contact Jim Cowan, Neutrik USA, 1600 Malone St., Millville, NJ 08322, (606) 327-3113.

Technical Papers

The 84th AES Convention took place last month in Paris, France. For those of you who didn't make the trip, a number of interesting loudspeaker papers were presented.

Preprint 2563—*Active Loudspeaker System for Sound Reinforcement Applications.* Describes a high output, low distortion, active speaker system using a ribbon-type driver for mid and high frequencies. The paper also describes a computer controlled method of manipulating directivity.

Preprint 2572—*Distortion Mechanisms in the Electrodynamic Motor System.* Correlates distortion and different types of magnet structures, suggesting improvements.

Preprint 2573—*Sound Radiation from Circular Stretched Membranes in Free Space.* Gives a method of predicting radiation characteristic of electrostatic speakers.

Preprint 2574—*Computation of Diffraction for Loudspeaker Enclosures.* Describes a computer program for predicting edge diffraction as part of a total CAD, multi-way speaker design package.

Preprint 2575—*General Equivalent Electrical Circuits for Acoustic Horns.*

Preprint 2576—*New Magnetic System Designs for Sound Reinforcement Loudspeaker Applications.* Discusses use of non-traditional materials in speaker motors.

Preprint 2577—*Constant Component of the Loudspeaker Diaphragm Displacement Caused by Nonlinearities.*

Preprint 2578—*Compact Ribbon Midrange/Tweeter Loudspeaker.* Discusses a new Philips single driver ribbon having a 800Hz-30kHz range.

Preprint 2600—*Applications of Digital Filters to Loudspeaker*

VOICE COIL, the newsletter for the Loudspeaker Industry is published on the 15th of each month by Audio Amateur Publications, P.O. Box 576, Peterborough, New Hampshire 03458-0176 USA. Edward T. Dell, Publisher; Vance Dickason, Editor; Karen Hebert, General Manager; Katharine Gadwah, Circulation Director.

Crossover Networks. Discusses high slope digital high- and low-pass filters and their subjective evaluation.

Preprint 2612—*Computer Simulation of Loudspeakers*. Discusses simulation techniques.

Preprint 2615—*Straightforward Determination of Loudspeaker Parameters Using Nyquist Interpolation Plots*. Describes a new method of parameter determination based on the interpolation of complex impedance loudspeaker plots derived from a two-channel FFT analyzer.

Preprint 2619—*A Model of Loudspeaker Impedance Incorporating Eddy Currents in the Pole Structure*. Shows impedance variations caused by pole piece eddy currents.

Preprint 2632—*Minimum Requirements for Reference Listening Rooms*. A proposed standard for listening rooms.

Preprint 2642—*Another Approach to the Ideal Crossover: The Energy Filler*. A filler driver technique for Linkwitz filters.

Preprint 2646—*An Electronic Loudspeaker Enhancement and Protection Device*.

CAD Software

LEAP's BOUNDS—USER REPORT PART II

Voice Coil's March issue discussed at length the substance of LEAP (Loudspeaker Enclosure Analysis Program) written by Chris Strahm. As promised, this month's installment includes examples of LEAP's calculations for a real driver.

The woofer example chosen is a 10" sealed-box design type. The woofer, the AC-10, is sold by mail-order retailer and OEM distributor, Audio Concepts. AC are an excellent OEM resource for small to mid-sized manufacturers. They stock component parts like caps, coils, and resistors as well as drivers, competitively priced in relatively small quantities. Contact them at PO Box 212, La Crosse, WI 54603, (608) 781-2110. Their AC-10 is a 40 oz. magnet, foam surround, polypropylene, stamp frame unit.

Parameters entered into LEAP for this woofer were as follows: Nom Z = 8Ω, S_d = .0234 sq. M, B_L = 9.3TM, R_{evc} = 5.86Ω, L_{evc} = 1.22mH, V_{as} = .138M³, $C_{ms} = 926 \times 10^6$ M/N, M_{ms} = 50.5 grams, F_s = 23.27Hz, Q_{ms} = 3.6010, Q_{es} = .5000, Q_{ts} = .4390, X_{max} = 7mm, P_{max} = 150W RMS.

The cabinet model entered was 3.07ft.³ for a calculated box Q of .7. This figure came from straight Thiele/Small calculations taken from the tables of the *Loudspeaker Design Cookbook*. The box represents an empty enclosure with no filling or any series resistance. The first design analysis was carried out at full precision at 1W/1M with a nominal voice coil temperature of 25°C. *Figures 1, 2,* and *3* are six of the ten curves from the LEAP analysis mode.

Looking at the frequency magnitude curve, we see pretty much what the Thiele/Small calculations predicted. The −3dB frequency prediction was 37Hz and the LEAP model shows 36Hz. The phase angle at this point is 92.8°, which is close to the expected −3dB phase angle of 90° for a theoretical Butterworth second-order electrical filter.

So far LEAP gives us a confirmation of our previous calculations, plus such useful information as acoustic phase, group delay, cone excursion, and impedance phase. Now let's see what happens to this woofer when we increase the power level and also increase the voice coil temperature, as it would in a long

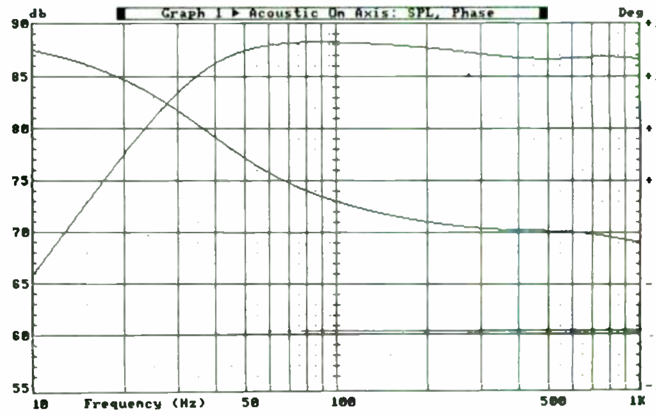

FIGURE 1: AC-10, 1W/88, dB/25, C −3dB phase angle 92.8.

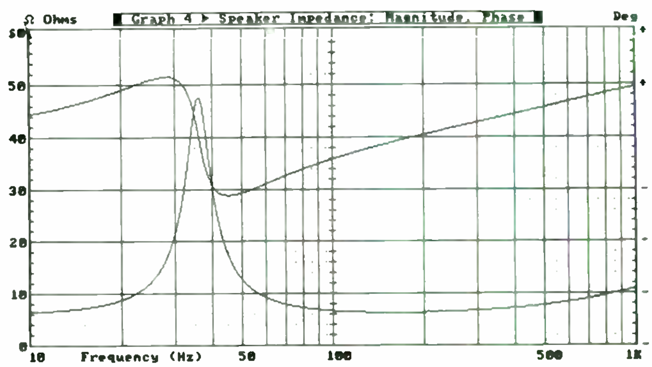

FIGURE 2: AC-10, 1W/88, dB/25, C −3dB phase angle 92.8.

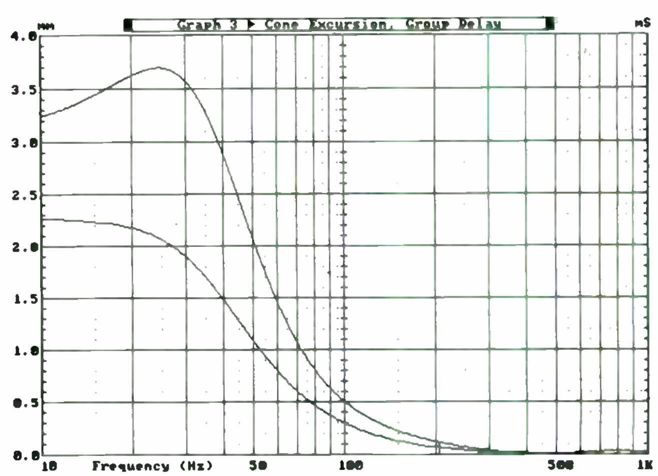

FIGURE 3: AC-10, 1W/88, dB/25, C −3dB phase angle 92.8.

term listening situation. For this I created a "family" of cone excursion curves, each at an increased power level and increased voice coil temperature factor. *Figure 4* gives the results for 1, 10 and 20W input levels coincident with an increase in temperature from 25 to 75°C.

Now it is easy to see that at a 100dB output level, a 30Hz note would require about 8mm of excursion. Since the maximum excursion for this driver is 7mm, we would be entering the area of nonlinear operation, and would expect distortion to increase at this point. Actually, this is pretty good for a single 10" woofer, and given the ear's relative insensitivity to low-frequency distortion, the driver would likely "sound" quite acceptable at this level and higher, but it's clear that we are near the edge of the linear operating "envelope" for this driver.

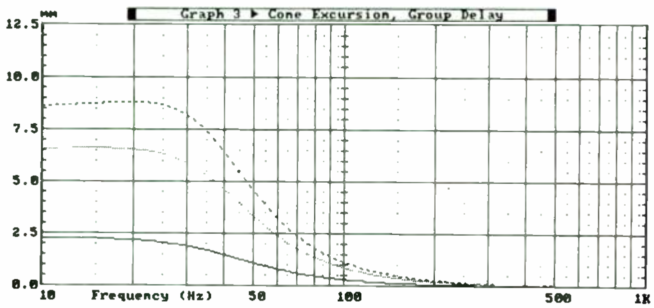

FIGURE 4: AC-10, 1W/88, dB/25, C −3dB at phase angle of 92.8; 10W/98, dB/50, C −3dB phase angle 97.4; and 20W/100, dB/75, C −3dB, phase angle 101.1.

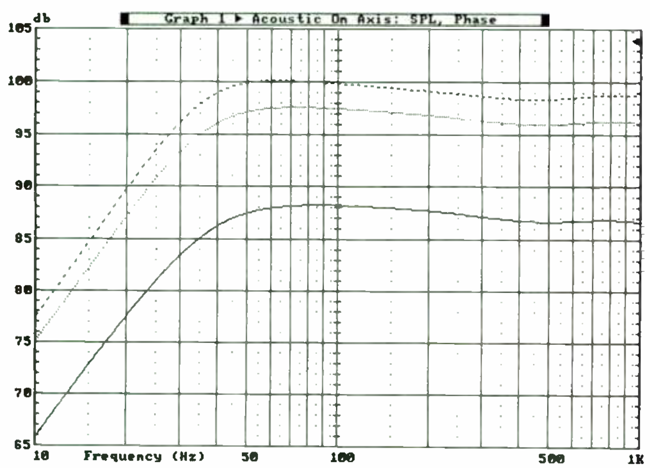

FIGURE 5: AC-10, 1W/88, dB/25, C −3dB at phase angle of 92.8.; 10W/98, dB/50, C −3dB phase angle 97.4; and 20W/100, dB/75, C −3dB, phase angle 101.1.

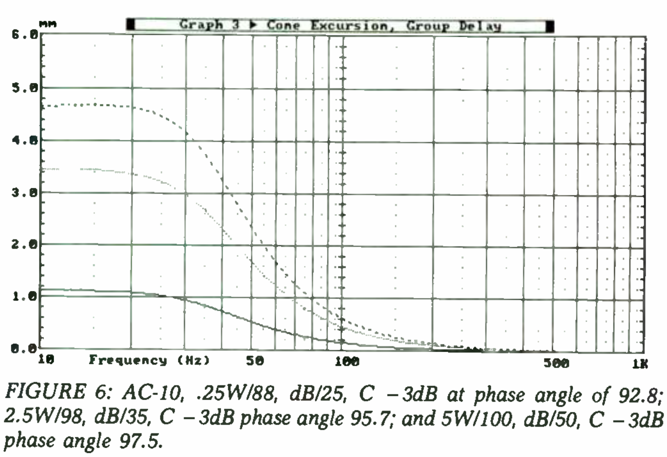

FIGURE 6: AC-10, .25W/88, dB/25, C −3dB at phase angle of 92.8; 2.5W/98, dB/35, C −3dB phase angle 95.7; and 5W/100, dB/50, C −3dB phase angle 97.5.

Notice also that as the output level increased, the woofer damping changes, and the phase angle at the −3dB point is now up to 101°. If you look closely at the composite SPL curves, *Figure 5*, the woofer rolloff has become slightly steeper at 100dB. LEAP simulations, as demonstrated in Chris' AES paper (Preprint #2419) are very close to reality—within 1-2dB.

If we really wished to make a high performance subwoofer with this driver, capable of high output and low distortion, the obvious answer would be to use two of them in a larger enclosure. *Figure 6* gives the cone excursion curves for such an enclosure. To achieve the same output levels as before, input levels are divided by four, since efficiency has increased 6dB over the previous single woofer design.

Things look much better at the 100dB level. Cone excursion required at 30Hz is now only a little over 4mm. The estimated voice coil temperatures would be decreased, and this is reflected by less change in the −3dB phase angle, which is now 97.5°. Incidentally, producing the composite curves shown here is very easy with LEAP's file control mechanisms.

As you can readily see, LEAP is an extremely powerful design tool, and I haven't half enough space to explain its potential for professional design work. At its current price of $149, it's an incredible bargain. In fact, if you don't already own a PC, it would be worth getting one just for this program alone. You can also expect a 4.0 version within the next year (maybe longer) which will include nonlinear modelling (the upgrade charge should be modest). For information contact CNS Electronics, PO Box 42389, Portland, OR 97242, (503) 231-7247.

OEM Driver News

Precision Loudspeakers announces another new woofer. The TX205R is a bass mid 8" woofer featuring a mineral filled polypropylene cone, a butydene-styrene surround, optimized for bass reflex and high efficiency closed-box alignments. Parameters are: F_s – 32Hz; V_{as} – 75 liters; Q_{ts} – .37; 90dB 1W/1M. For further technical information, contact Dick Pierce, Precision Loudspeakers, 2-B Columbia Drive, Amherst, NH 03031, (603) 883-7050, TWX 5106014163 PRECISION SPKR.

Stereophile's High End Hi-Fi Show

Stereophile held its third Hi-Fi show on April 8-10 in Santa Monica, California. While final attendance figures weren't available at the time *Voice Coil* went to press, it was in excess of 4000 enthusiastic audio consumers over the three-day period. All the loudspeaker manufacturers represented at the show were very excited by the response. The scene certainly resembled the kind we saw at the Riviera Hotel at the Winter CES. Judging from exhibitor comments, this type of consumer show can provide effective promotion for the industry, and we hope they will continue.

Loudspeaker manufacturers in attendance included:

Apogee Acoustics debuted their new "Signature" series of full-range ribbon speakers.

Ariston Acoustics displayed its newly released "Image" series two-way loudspeakers at the show.

Audio Products International demonstrated the Energy, Mirage (a dynamic driver dipole radiator) and Image lines of speakers. API, while currently using their own facilities, was one of the first speaker manufacturers to make use of the facilities of the NRC of Canada. They were also involved initially with the creation the Peter Schuck's network optimization software. Watch for a report in *Voice Coil.*

B&W Loudspeakers demonstrated the 801s and displayed the "Matrix" series.

dbx showed their new SoundField series released at the Winter CES.

Duntech displayed their Sovereign as part of a large video projector surround-sound demonstration. Duntech will also be shortly releasing a new speaker, probably at the June CES. The Marquis is a three-way, five-driver unit using two 8" woofers, two 4.5" mids, and one .75" dome driver. Price is $3,500/pair. The drivers will all be Duntech constructed proprietary designs, rather than the Dynaudio drivers Duntech has used in its other highly successful speakers.

Fostex demonstrated their line of loudspeakers which feature ribbon mid- and high-frequency drivers.

GNP Audio Video primarily demonstrated their Valkyrie loudspeaker.

Lantana Ltd. displayed their Tad speaker with Laug subwoofer. The company is also introducing two new units. The Mink uses the Tad upper range drivers and a dual 6" rear-firing subwoofer. The Martin uses the Tad basic driver complement, with an 8" rear-firing subwoofer. The Martin is priced at $1,150/pair and the Mink at $950/pair.

Madisound promoted their Audio Projects Computer Bulletin Board. The company also had literature available on their OEM woofer line.

While **Marantz** did not display any of their new loudspeakers shown at the Winter CES, they shared their space with an unusual new loudspeaker from **Sound Pipes**, constructed from large-diameter plastic ABS, the kind used for drain pipes in your home.

Mitek demonstrated their new ZTE enclosureless design. Introduced at the Winter CES, it received considerable attention. Mitek is now shipping units to its dealers.

Monitor Audio displayed several of the models in their line.

NAD showed the new Stratus loudspeaker by PSB (now owned by NAD). The speaker, designed by Canadian Paul Barton, who also used NRC facilities and was among the first to do so. As I noted in an earlier *VC*, NRC has an automated computerized measuring setup which is able to average many on- and off-axis response points to assemble a composite response suitable for network design. Paul also used Peter Schuck's XOPT for network design.

Naim Audio demonstrated their SBL speaker.

Nelson-Reed's Ron Nelson was on hand to demonstrate the latest of his designs for the LA crowd. (When I tried to get in to see Ron, it was more like a mob).

Rogersound Labs—although not a national manufacturer, RSL has been a strong force in the California market with its own line of speakers sold both in their own stores and other California dealers.

SOTA—although not the focus of the SOTA booth, the Panorama loudspeaker was on display. A companion subwoofer is due out fairly soon.

Speaker Builder. As one of *Speaker Builder*'s Contributing Editors, as well as the author of the *Loudspeaker Design Cookbook*, published and distributed by the same organization, I was extremely pleased to see the enthusiastic response of Californians to the idea of professional home audio equipment building. Put it this way, we distributed 1000 complimentary copies of the latest *Speaker Builder* and several hundred of *Audio Amateur* in about four hours. Publisher Ed Dell flew in from New Hampshire to enjoy the affair, which included a highly original all-horn loudspeaker designed by *SB* Contributing Editor Bruce Edgar. Amplification was from an all-MOSFET design by Bill Chater.

Vandersteen Audio demonstrated the highly successful 4A speaker.

Recent Product Announcements

Linn has announced the release of the two-way Nexus speaker. This speaker utilizes an 8" woofer and .75" tweeter dome mounted on an expanded foam polymer baffle board. Price is $1,095 per pair.

MB Quart Electronics has recently introduced the entire line of MB finished loudspeakers from Germany. The line has five models including a two-way 6.5", two-way 8", three-way 10" with a cone mid, three-way 10" with dome mid, three-way 8" with dome mid, and a dual 8" three-way with a dome mid. Prices range from $389 to $1,999 per pair. MB hopes to get into the US OEM market within a year or so.

Martin-Logan has introduced its new Sequel speaker system. The Sequel is a 4-foot curvilinear electrostatic with a built-in 10" woofer system. Price for the Sequel is $2,250 pair.

Advent has released its Baby II, an improved version of the Baby Advent. This 6.5" two-way is $250 per pair.

Recent Patents

These recent, and past patents, all relating to loudspeakers and found primarily under the Office of Patents and Trademarks classification 181 and 179, Acoustical Devices, are available for $150 each from: The Commissioner of Patents and Trademarks, Washington, DC 20231.

DOMESTIC LOUDSPEAKERS

Patent No. 4,489,432-Method and Apparatus for Reproducing Sound Having a Realistic Ambient Field and Acoustic Image, Matthew S. Polk, assigned to Polk Audio. Describes the technology applied in the highly successful Polk Audio SDA series of loudspeakers. The design improves upon sonic image by a unique method of canceling interaural crosstalk.

Patent No. 4,475,233-Resistively Damped Loudspeaker System, William H. Watkins, Kingsport, TN. Watkins, who sold one of his dual voice coil patents to Infinity some years back, describes a method of paralleling resistors with all reactive drivers in the crossover. This is a rather indiscriminate way to deal with the reactive impedance of a dynamic driver. Conjugate networks are usually the method of choice. However, computer optimizing network programs can sometimes accomplish the desired goal without any compensation.

Patent No. 4,160,882-Double Diaphragm Electrostatic Transducer Each Diaphragm Comprising Two Plastic Sheets Having Different Charge Carrying Characteristics, Michael Driver, Altadena, CA. A "sandwich" electrostatic assembly using polyester film, a metalized layer, and an optional vinylidene chloride layer is said to increase efficiency, and be relatively low cost in production.

Patent No. 4,154,979-Woofer Efficiency, assigned to Bose Corporation. Similar to Thiele's sixth-order augmented alignments, this patent describes a particular vented tuning arrangement with the addition of active equalization to obtain a particular response. In this case, the response appears to favor bass emphasis.

MINIMUM PHASE CROSSOVERS

BY RALPH GONZALEZ

The SPICA TC-50 loudspeaker is highly regarded for its subjective qualities (*SB* 4/87, p. 56). Bert Whyte briefly described this speaker in *Audio*,[1] and reported it had a phase response of 0° (±15°) over the 350Hz to 4.2kHz range. This amounts to a (nearly) *minimum-phase* response.

For a speaker with a flat magnitude response, various frequencies, as they are reproduced by the speaker, experience little change in their sine waves' relative magnitude. If a speaker also has a *phase response*[2] which is near zero over a wide frequency range (minimum phase), then the relative *phase angle* of frequencies reproduced in this range will change little.

Since we can think of a square wave as the sum of many sine waves of different (harmonically related) frequencies, a *non-minimum phase* speaker may offset sine waves of one frequency with respect to those of another, altering the *shape* of the square wave as viewed on an oscilloscope.

Likewise, such a speaker would change the shape of a music waveform. The audible effect of this on listening to music has not yet been firmly established, although some claim it affects clarity, image depth, and so on. (Curiously, identical parameters are said to be affected by such unrelated audio components as amps, cables, tonearms, power cords, and so on.) A recent study by S. Lipshitz, M. Pocock, and J. Vanderkooy[3] confirms that interest in the phase issue is more than simply academic.

The only component in the audio chain which is ever clearly non-minimum phase is the multi-way loudspeaker. The individual drivers are often minimum phase, but the crossover requirement usually spoils this. Many speaker companies exploit the public's mystification over matters of phase, and advertise their speakers as "linear phase," "phase coherent," "phase perfect," "time correct," and so on. While some such speakers are minimum phase designs (or at least attempts), in practice, these terms do not necessarily indicate a minimum phase speaker. There are currently very few minimum phase speakers on the market.

PRACTICAL CROSSOVER CONSIDERATIONS. There are several approaches to obtaining a minimum phase crossover.[4-8] Most of these approaches are impractical for passive crossover design, or have other drawbacks such as severe lobing.

Given drivers with sufficiently wide bandwidths, three useful minimum phase crossover approaches that I am aware of are worth considering:

- A true first-order crossover,
- A three-way "filler driver" approach, and
- A crossover employing a high-order Bessel low-pass filter and deliberate interdriver depth displacement, as pioneered by SPICA.

The first-order approach seems easy to design at first glance, but requires drivers with extremely broad bandwidths, including a tweeter which can handle a first-order crossover without damage. If a first-order crossover is applied to drivers which are operated near the limits of their natural response ranges (as is usually the case), then the actual slopes will be closer to second- or even third-order. A three-way approach using a wide-band cone midrange is more feasible to obtain a well integrated minimum phase system with true first-order slopes.

The "filler driver" technique[9] for three-way systems also looks promising. In this case, only the midrange needs first-order slopes and the woofer and tweeter are allowed second-order slopes. (Due to driver rolloffs, you are still likely to be limited to first-order filters.)

In practice, a trial and error approach is usually required, using software such as LMP, for modelling crossover and driver configurations. The resulting slopes may share aspects of both the first-order and the filler driver ideals. Use notch filters to equalize the woofer's and/or midrange's natural high frequency rolloffs, to appear first-order over part of their respective frequency ranges.[10]

The SPICA TC-50 woofer, on the other hand, employs a Bessel low-pass filter,

1. Whyte, B., "Behind the Scenes," *Audio*, March 1984, p. 26.
2. Gonzalez, R., "An Introduction to Frequency Response and LMP," *SB* 1, 2/87.
3. Lipshitz, S.P., M. Pocock and J. Vanderkooy, "On the Audibility of Midrange Phase Distortion in Audio Systems," *JAES*, Vol. 30, 1982, p. 580.
4. Clarkson, P.M., J. Mourjopoulos and J.K. Hammond, "Spectral, Phase, and Transient Equalization for Audio Systems," *JAES*, Vol. 33, 1985, p. 127.
5. Lipshitz, S.P. and J. Vanderkooy, "A Family of Linear-Phase Crossover Networks of High Slope Derived by Time Delay," *JAES*, Vol. 31, 1983, p. 2.
6. Lipshitz, S.P. and J. Vanderkooy, "Use of Frequency Overlap and Equalization to Produce High-Slope Linear-Phase Loudspeaker Crossover Networks," *JAES*, Vol. 33, 1985, p. 114.
7. Preis, D., "Phase Distortion and Phase Equalization in Audio Signal Processing—A Tutorial Review," *JAES*, Vol. 30, 1982, p. 774.
8. Small, R.H., "Constant-Voltage Crossover Network Design," *JAES*, Vol. 19, 1971, p. 12.
9. Baekgaard, E. (Bang & Olufsen), "A Novel Approach to Linear Phase Loudspeakers Using Passive Crossover Networks," *JAES*, Vol. 25, 1977, p. 284.
10. Gonzalez, R., "An Introduction to Frequency Response and LMP," *Part II, SB* 2/87, p. 47.

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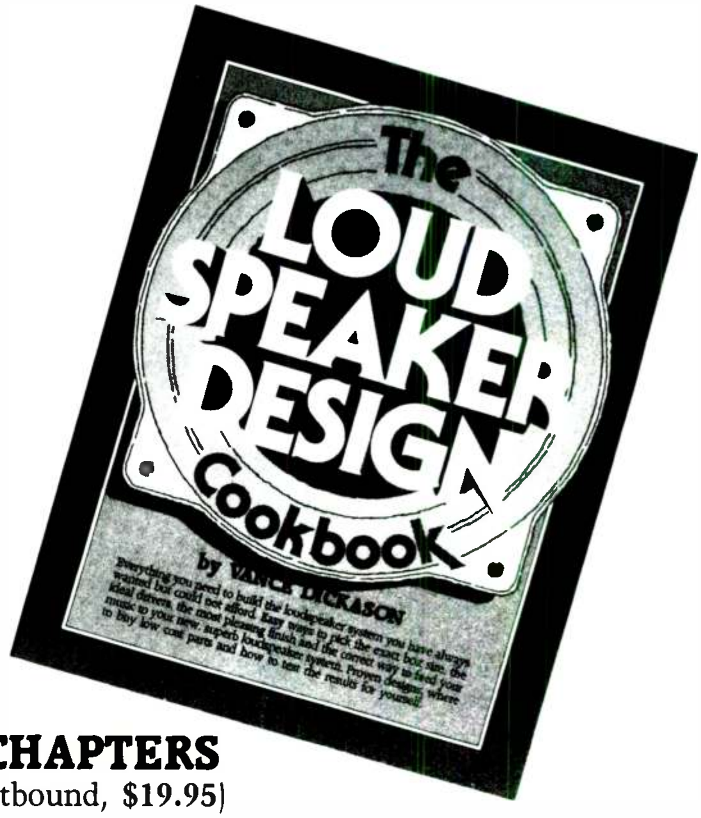

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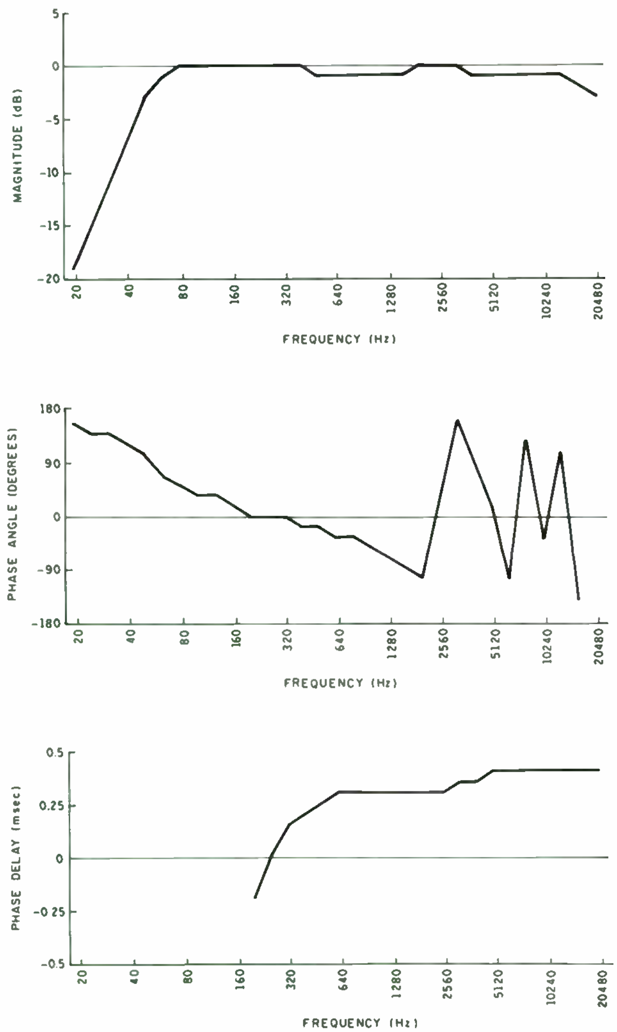

FIGURE 1: Bessel crossover, zero delay at woofer location.

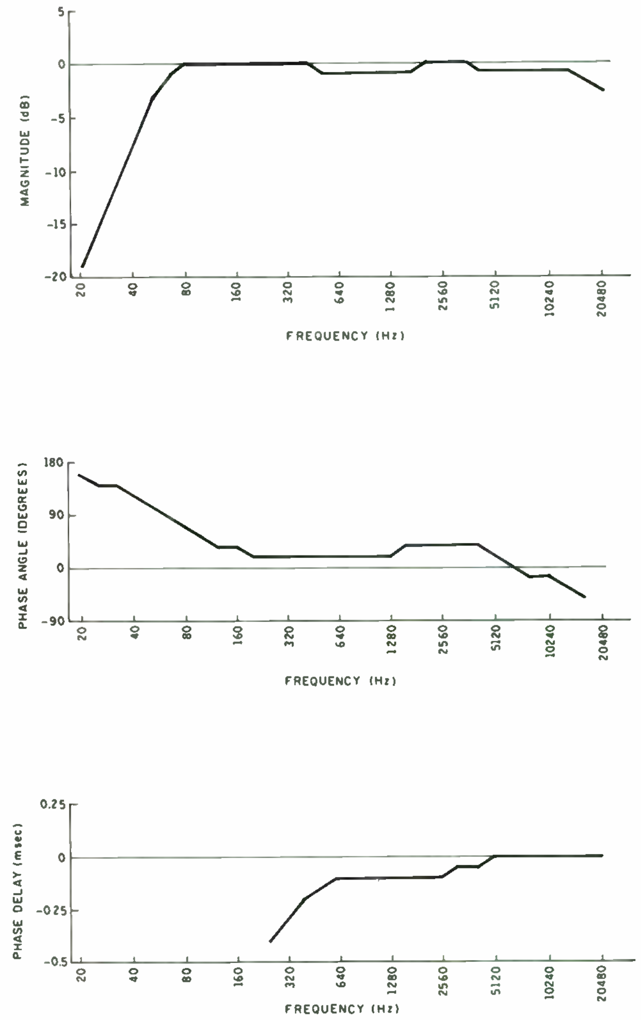

FIGURE 2: Bessel crossover, zero delay at tweeter location.

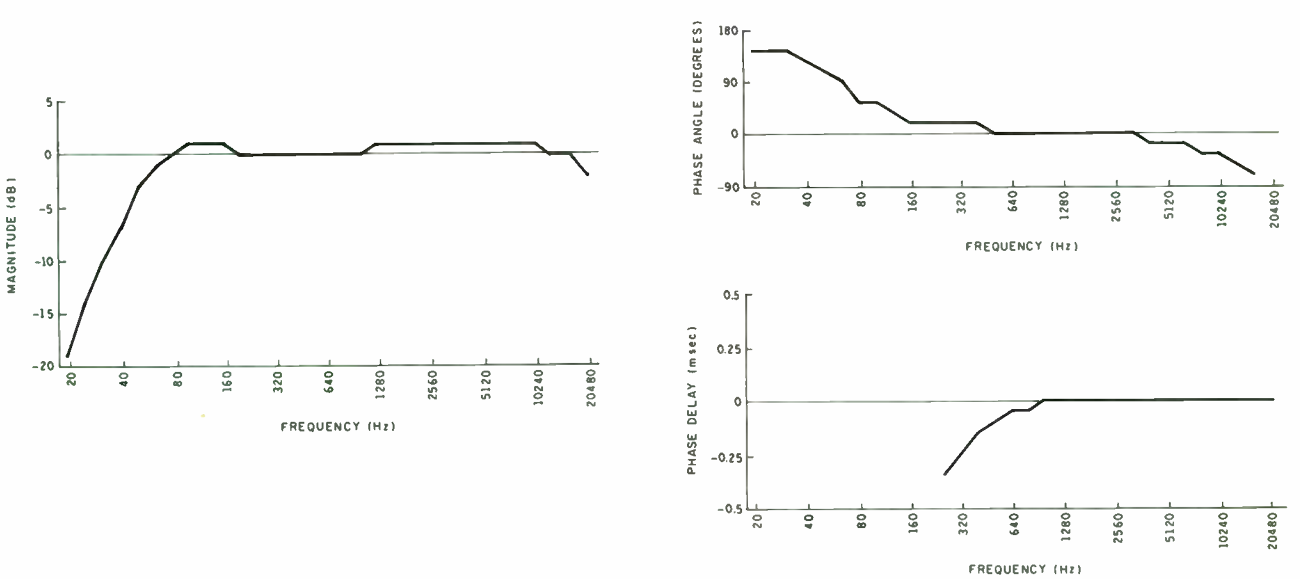

FIGURE 3: First-order crossover, zero delay at woofer/tweeter location.

whose phase angle approximates that of an ordinary time delay. By physically displacing the tweeter (via the unusual slanted front baffle), its own time delay relative to the woofer matches that approximated by the Bessel filter. Thus, both drivers are approximately equivalent to a single driver with a time delay.

A time delay's phase shift is proportional to frequency, and does not distort the musical waveform's shape. By compensating for this time delay in the speaker measurement, any such system can be made (nearly) minimum phase. The only problem is the phase shift from the tweeter's high-pass crossover and the tweeter's inherent phase shift.

Apparently the designers of the SPICA TC-50 were able to design a high-pass filter for the tweeter which combined with the woofer to give a smooth magnitude response as well as the aforementioned near-minimum phase characteristic over the 350Hz to 4.2kHz range. (Beware of other manufacturers who limit their discussion, for example, to frequencies above 2kHz.) According to John Bau, the founder of SPICA, design tools used to produce the TC-50 were given incorrectly in Bert Whyte's *Audio* article. The primary design tool was Deane Jensen's COMTRAN (Computerized Optimization and Transient Analysis) program, written for HP desktops. John Bau used an FFT measurement system to verify the design.

If you believe that phase distortion is relevant, it is still unclear whether the relative phase angle of different frequencies in a musical signal is audible, or whether the time delay corresponding to such a phase shift is more significant.

MEASUREMENTS. For example, a phase shift of 90° at 1kHz corresponds to a time delay of 0.25msec, while the same phase shift at 10kHz corresponds to only 0.025msec. Divide the phase angle by the frequency at each point on the phase angle plot and take the negative, to obtain a plot of the *phase delay*.[11] This plot indicates the amount of time delay required to produce the measured phase angle at each frequency.

I believe a good measure of how well a speaker maintains the shape of the musical waveform is the *amount of variation in the phase delay plot* over the *frequency range of interest*. This is accurate only if the frequency amplitude response is flat; the phase delay can be ambiguous unless the system is already near-minimum phase.

Since loudspeakers are measured by a microphone at some distance, a time delay must exist in the phase-angle measurement and it is useful to compensate for this measurement, to eliminate the huge (albeit unimportant) resulting variation in the phase-angle graph.

Figures 1 and *2* show the LMP simulation of a system employing a Bessel crossover, including the woofer's low frequency cutoff and the tweeter's high frequency limit. The woofer employs a fourth-order, low-pass Bessel filter and the tweeter, a second-order, overdamped, high-pass filter. These slopes *include* the driver rolloffs; that is, the electrical filters were designed to provide these slopes only after summing with the drivers' natural responses.

The woofer diaphragm is 2.7 inches closer to the listener than the tweeter diaphragm, and the crossover occurs around 2kHz. The phase-angle plots are compensated so there is no time delay at the physical location of the woofer diaphragm (*Fig. 1b*) or the tweeter's location (*Fig. 2b*). The variation in phase angle in *Fig. 2b* is admirably low, and supports SPICA's claims for the TC-50. Note that the amount of *variation* in the phase delay plots (*Figs. 1c* and *2c*) is unaffected by the time-delay compensation. (I modified LMP to produce these plots.)

Figure 3 shows the LMP simulation of

11. Kaufman, R.J., "Phase Filter for Digital," *Audio*, July 1984, p. 34.

a speaker with an ideal first-order crossover at 2kHz, with the diaphragms of both drivers physically aligned, equidistant from the listener. (Again, by "ideal" I mean the first-order slopes must result from the summation of the electrical crossover response with the driver responses. This is virtually impossible with real-world drivers.) I could have inserted a small time "lead" compensation into the measurement, to make the phase-angle plot even better above 5kHz.

CONCLUSIONS AND QUESTIONS. The *Figures* show the Bessel crossover has a great deal of merit in designing a near-minimum phase, two-way speaker. The bandwidth requirements of the drivers are much less stringent than those of the ideal first-order crossover, particularly if the woofer's and tweeter's natural rolloffs are incorporated into the crossover design. However, since the low-pass Bessel filter is only an approximation of a true time delay, this approach cannot quite match the phase integrity of an ideal first-order crossover.

Of course, SPICA's designers may have used a slightly different implementation. We can conclude, then, that the ideal first-order crossover, while requiring very special drivers and/or a three-way implementation, has the potential of providing less variation in phase delay over a broader frequency range.

Two questions now arise: Do the SPICA TC-50's special subjective qualities arise from its Bessel crossover or from its unusual enclosure geometry? And, if the crossover is significant, what frequency range and variation in phase delay are sufficient to make a speaker sound identical to one with ideal minimum phase?

Finally, I'm not sure whether *group delay* (the negative of the rate of change of phase angle with respect to frequency) is a viable means of evaluating a speaker's ability to preserve the musical waveform, since this measurement cannot distinguish a constant phase shift (which distorts the shape of the waveform) from zero phase shift. Group delay is a good measure of whether phase is a linear function of frequency; however, a linear function can include a constant term. That is, phase may be linear with frequency yet not be proportional to frequency; and such a phase response would not indicate a simple time delay.

I'd be very interested in hearing the opinions of other readers about any of these matters.

Using Bessel Crossovers

You may experiment with Bessel crossovers using LMP. First, enter your woofer model (alone) into LMP, as detailed in my article[2] A good approximation to a Bessel filter may be obtained by using a first-order, low-pass filter with the woofer, one or two octaves below the woofer's natural second- or fourth-order high-frequency cutoff.

Next, try different values for the woofer's *Depth Displacement* (usually +1 to +3 inches). Choose the value which makes the phase response flat, to the highest frequency possible.

Now you can specify *Number of Drivers*=2, and add your tweeter's model. Enter *Depth Displacement*=0, and *Polarity Inversion*=N. Try a first-order, high-pass filter at the same frequency as the woofer's low-pass filter.

Experiment with the crossover frequencies and the woofer's *Depth Displacement*, until you've arrived at a smooth overall response.

Though you can experiment further, you may have difficulty obtaining a smooth response with a higher order high-pass crossover. The Bessel approach generally requires relatively high-power, wide-band drivers.

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FILTERS & SPEAKER SAVER

KH-2: SPEAKER SAVER AND OUTPUT FAULT DETECTOR [3:77]. This basic two-channel kit includes board and all board-mounted components for control circuitry and power supply. It features turn-on and off protection and fast opto-coupler circuitry that prevents transients from damaging your system. The output fault detector has additional board-mounted components for speaker protection in case of amplifier failure. Each **$62**

KF-6: 30Hz RUMBLE FILTER. [4:75] This kit implements Walt Jung's 1975 design for a low frequency garbage filter. The filter knee is set to 30Hz. Roll-off below that knee is the 18dB/octave characteristic of its three pole design. Gain for the filter is unity (0dB) but can be simply adjusted for up to 12dB of gain. The reprint of Jung's article explores the use of the filter with other components in crossovers (see kits SBK-C1A, C1B, C1C). He shows how to obtain slopes of 6, 12 or 18dB in high and low pass filters. The kit contains all parts for building a two channel HPF including a board (3" x 3"), quad op amp IC, precision resistors and capacitors. Requires a bipolar supply of ± 15V, the KE-5 is suitable. Each **$28**

AIDS & TEST EQUIPMENT

KK-3: THE WARBLER OSCILLATOR [1:79]. This unit will produce a swept signal covering any ⅓-octave between 16Hz and 20kHz. The total harmonic distortion at the output is less than 1.5%. The output voltage is adjustable from 0 to 1V. When used with a microphone it is as effective as a pink noise source in evaluating speaker system performance. It also reveals the listening environment's effect on sound through reflection and absorption. The sweep rate is set at about 5Hz. The kit includes 3¼" x 3⅜" circuit board, transformer, all parts and article reprint. Each **$65**

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 **$62**

KC-5: GLOECKLER 23-POSITION LEVEL CONTROL. [2:72] All metal film resistors, shorting rotary switch and two boards for a two-channel, 2dB per step attenuator. Choose 10k or 250kΩ. Each **$42**

KL-6: MASTEL TIMERLESS TONE BURST GENERATOR. [2:80] All parts with circuit board. No power supply. Each **$24**

KP-2: TWO TONE INTERMODULATION TEST FILTER. [1:82]. This filter is designed to isolate the two high frequency tones at an amplifier's input from low frequency intermodulation products present at the output. The high pass filter corners at 2kHz and rolls off at 24dB/octave. A 5kHz signal at the low pass input will be down at the output by 80dB. An article reprint detailing design and use is included with the kit. All parts are supplied including quad op amp IC, circuit board and precision resistors and capacitors. Each **$26**

SBK-D2 WITTENBREDER AUDIO PULSE GENERATOR. [SB 2:83] All parts, board, pots, power cord, switches and power supply included. Each **$80**

SBK-E4: MULLER PINK NOISE GENERATOR. [SB 4:84] All parts, board, 1% MF resistors, capacitors, ICs, and toggle switches included. No battery or enclosure. Each **$32**

CROSSOVERS

KC-4A: ELECTRONIC CROSSOVER, KIT A. [2:72] Single channel, two-way. All parts including C-4 board and LF351 ICs. Choose frequency of 60, 120, 240, 480, 960, 1920, 5k or 10k. KE-5 or KF-3 supplies are suitable. Each **$12**

KC-4B: ELECTRONIC CROSSOVER, KIT B. [2:72] Single channel, three-way. All parts including C-4 board & LF351 ICs. Choose two frequencies of 60, 120, 240, 480, 960, 1920, 5k or 10k. Each **$15**

KK-6L: WALDRON TUBE CROSSOVER LOW PASS: Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot. Choose 1: 19-210; 43-465; 88-960; 190-2100; 430-4650; 880-9600; 1900-21,000 hertz. Each **$58**

KK-6H: WALDRON TUBE CROSSOVER HIGH PASS: Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot. Please specify 1 of the frequencies in KK- 6L. No other can be supplied. Each **$60**

KK-7: WALDRON TUBE CROSSOVER POWER SUPPLY. [3:79] Includes board, transformer, fuse, semiconductors, line cord, capacitors to power four tube crossover boards (8 tubes), 1 stereo bi-amped circuit. Each **$100**

SBK-A1: LINKWITZ CROSSOVER/FILTER. [SB 4:80] Three-way crossover/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 **$72** Two channels **$132** SBK Board only **$14**

SBK-C1A: JUNG ELECTRONIC TWO-WAY CROSSOVER. [SB 3:82] 30Hz filter with WJ-3 board & 4136 IC adapted as one channel crossover. Can be 6, 12 or 18dB/octave. Choose frequency of 60, 120, 250, 500, 1k, 2k, 5k or 10k. The KL-4A/KL-4B or KW-3 are suitable supplies. Each **$30**

SBK-C1B: THREE WAY, SINGLE CHANNEL CROSSOVER. [SB 3:82] Contains 2 each SBK-C1A. Choose high & low frequency. Each **$58**

SBK-C1C: TWO CHANNEL, COMMON BASS CROSSOVER. [SB 3:82] Contains two each SBK-C1A. Choose 1 frequency. Each **$62**

SBK-C2: BALLARD ACTIVE CROSSOVER. [SB 3,4:82] three-way crossover with variable phase correction for precise alignment. Kit includes PC board (5³/₈ x 9¹/₂"), precision resistors, polystyrene & polypropylene caps. Requires ± 15V DC power supply—not included. Can use KL-4A/KL-4B or KW-3. Two channel **$145**

SYSTEM ACCESSORIES

KW-3 BORBELY IMPROVED POWER SUPPLY [1:87] This single channel, low impedance supply was designed for the exacting requirements of Erno Borbely's moving-coil preamp [2:86, 1:87]. The design utilizes polypropylene caps and 1% metal film resistors. LM317/337s are used in the preregulator and Signetics NE5534 in the op amp regulator. The kit includes a low profile 24V toroidal transformer, 4¼" x 5½" circuit board and all board mounted components. Chassis and heatsink are not included. Each **$130** Two or more **$122**

KE-5: OLD COLONY POWER SUPPLY. Unregulated, ± 18V @ 55mA. Each **$20**

KF-3: GATELY REGULATED SUPPLY. ± 18V or ± 15V @ 100mA. Each **$48**

KL-4A: SULZER POWER SUPPLY REGULATOR. Each **$40**

KL-4B: SULZER DC RAW SUPPLY. ± 20V @ 300mA. Each **$42**

KH-8: MORREY SUPER BUFFER. [4:77] All parts, 1% metal film resistors, NE531 ICs, and PC board for two-channel output buffer. Each **$20**

SBK-E2: NEWCOMB NEW PEAK POWER INDICATOR. [SB 2:84] All parts & board, new multicolor bar graph display; red, green & yellow LEDs for one channel. No power supply needed. Each **$14** Two for **$22**

KL-2: WHITE DYNAMIC RANGE & CLIPPING INDICATOR. [1:80] One channel, including board, with 12 indicators for preamp or crossover output indicators. Requires ± 15V power supply @ 63 mils. Single channel. Each **$58**
Two channels. **$110** Four channels. **$198**

KW-1: MAGNAVOX CD PLAYER MODIFICATION. Improves frequency response. Includes two Signetics NE5535s, two Panasonic HF series 330μF capacitors and four 3.92k, 1% metal film resistors. Each **$12**

KW-2: MODIFICATION. As above, but with two AD-712 op amps in addition to the NE5535s. Each **$16**

KX-1: CD ACCESSORIES. Set of 3 Sorbothane feet, 3 Tiptoes and Mod Squad's Disc Damper with 15 centering rings. Each Set **$75**

KX-2: POOGE-4 CD PLAYER MOD. Jung/Childress extensive rework of the Magnavox CDB 650. Call or write for details.

HDHFT: HI-FI TIPS. Imported for Old Colony. Solid brass, ⅞"H conical feet for components and loudspeakers. Includes self-adhesive pad. Each **$3.00**
10 or more Each **$2.50**

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, faceplate, 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 ***TAA*** and ***SB*** 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.

SPOT SOUND ABSORPTION

BY THOMAS E. COX

Ideally, if you designed the ultimate speaker enclosure—your listening room, it should have proportions that distribute each room dimension's resonant frequencies, wall and ceiling contours which aid reflected sound diffusion, and selective sound absorption.

Few speaker builders have the opportunity to custom design their listening room for ideal acoustics and even fewer could redesign a room for a new speaker radiation pattern, for example, changing from front firing cone driver enclosures to flat panel drivers with dipole patterns. However, you can adapt the acoustics of an existing room to the speaker radiation pattern using selective absorption and diffusion. For this application, I designed spot sound absorbers which can be hung from any convenient wall, and which I built using picture framing techniques.

I constructed a frame to enclose the sound absorbing material as shown in *Fig. 1*. Frame dimensions, materials and construction methods are not critical, but my design is intended to blend aesthetically with my other, more usual wall hangings.

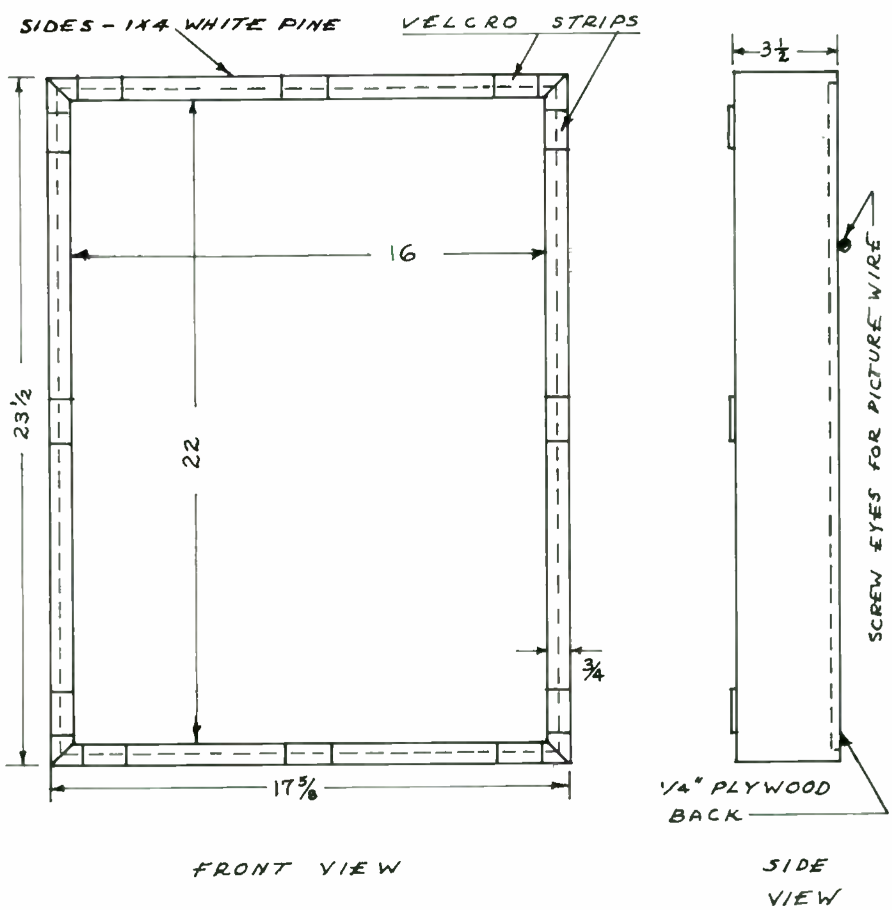


FIGURE 1: Absorber frame construction.

TEST SAMPLES. I assembled several frames and experimented with sound absorbent filler materials inside the frame, including fiberglass, polyester batting, and polyurethane in both sculptured and block forms. I also tried two front covers with each filler material, one of acoustically transparent material, and one absorbent type.

After some experimentation with different combinations of fillers and covers, I selected five combinations for testing. Cross sections of my five test absorbers are shown in *Fig. 2*. For the tests, I suspended the frames from a simulated wall section on an ordinary picture hook.

ACOUSTIC MATERIAL. Your local building materials supplier is unlikely to have an extensive stock of acoustical raw materials but some persistence will usually yield a supplier (see list of sources at the end of this article).

To locate a source for block polyurethane foam I inquired at a furniture store for their supplier of special foam "rubber" cushions for custom interior decorating jobs. They recommended a very helpful local firm which cut small blocks for me from huge (3 by 6 by 9 feet) slabs of raw stock called buns. The special purpose cutting machines held tolerances of $\frac{1}{16}$-inch. I purchased an industrial grade, polyester urethane foam; a single 16 by 22 by 3-inch block.

Sculptured polyester urethane is available directly from the manufacturer under the tradename Sonex. I ordered the smallest quantity available, Sonex' Junior

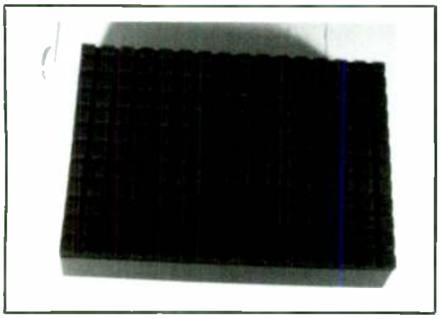


PHOTO 1: Photo 1 shows a completed spot sound absorber mounted on the wall section for testing.

package of four, two-inch thick, 24-inch squares (2 convex and 2 mating concave sculptured patterns).

I compared high density polyester batting (often referred to a "tailor's fleece") at several fabric stores, and I selected a material with a density of approximately 3 pounds per cubic foot. I used fifteen layers of this batting to fill the frame.

For the fiberglass test material, I also used 3 pounds per cubic foot density. While this is a popular commercial sound absorbing product, a minimum order for local delivery was $250. Fortunately, the wholesale distributor agreed to sell me a smaller quantity (six 24 by 48 by 3-inch batts) if I picked up the package at his warehouse. Even the single package was far more material than I needed but I intend to use the remainder for area sound absorption. This fiberglass is much denser than wall insulation types, cuts easily with a sharp knife and its stiffness eliminates any density changes through settling.

Fiberglass is the lowest cost filler I used, at roughly three dollars per frame, if you have use for all 48 square feet in the package. High density polyester at $13, and block polyurethane at $10 are the least expensive fillers if you need only a small area of absorbent material.

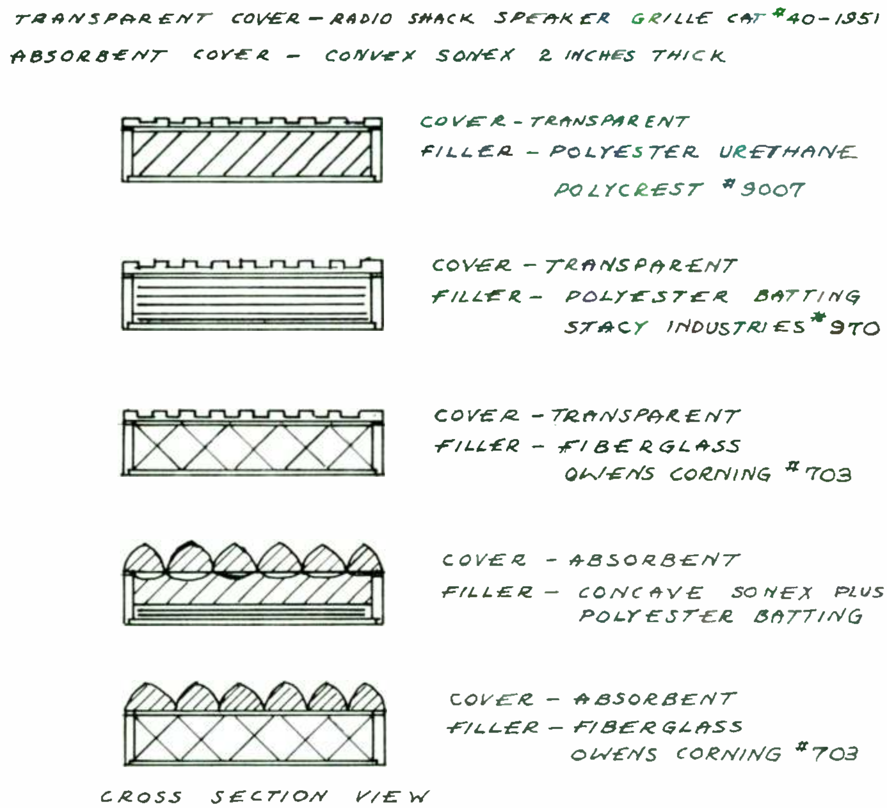


FIGURE 2: Fillers and covers.

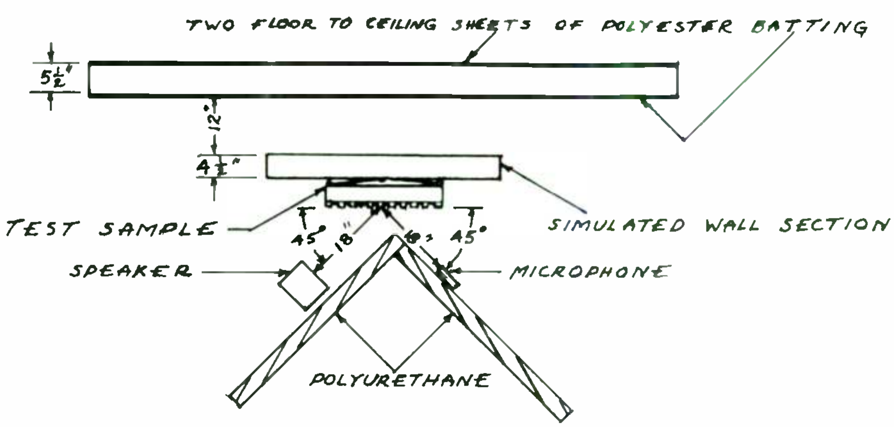


FIGURE 3: Spot sound absorption test setup.

ABSORBER TESTS. Commercially available products, like Sonex, are tested for sound absorptivity using methods specified by the American Society for Testing Materials (ASTM). This test essentially measures the change in decay time for a standardized reverberation room with and without the test materials. Decay rate frequency dependence is measured by performing the test at 125, 250, 500, 1k and 4k hertz.

The ASTM test provides an objective measure of relative sound absorption for general purpose applications. However, I was unable to find any published information on the behavior of sound absorbing materials with waves arriving from specific off-axis directions.

For my application, spot absorption of musical sounds on the wall of a frame house, I devised the test setup shown in *Fig. 3* to simulate these special conditions. I mounted test absorbers on a simulated wall section; 2 by 4 studs at 16 inches center-to-center, 3½-inch foil-backed fiberglass insulation between the studs, and ½-inch gypsum wall board covering front and back.

I generated test sound waves with a ⅓-octave sweep generator (Dick Crawford's "The Warbler," *TAA* 1/79, p. 22), driving an NAD 3020A amplifier, with Boston Acoustics' A40 bookshelf speaker. I measured sound levels after reflection with a microphone and microphone amplifier (Numark STD272 and PX2626) feeding a 200kHz, true RMS, digital voltmeter (Fluke Model 8060A). Two 24 by 48 by 3-inch acoustically absorbent polyurethane foam sections (Crest-Foam #9007 polyester urethane), between speaker and microphone, limited direct sound transmission and promoted the test signal's arrival at a 45° angle to the test sample axis.

First I determined frequency response of 1/3-octave warble signals reflected from the wall section alone. Then I hung each test absorber on the wall section and determined changes in frequency response relative to the bare wall section response.

RESULTS. Test data for the five test sample frames is shown in *Table 1*. Entries show the decibel change for each frequency response tested, compared to the blank wall section response.

My samples are not particularly effective at low frequencies but the polyester batting (B) and fiberglass (C) fillers with transparent covers are more effective at 400Hz. In midrange tests, polyester filler/transparent cover (B), fiberglass filler/transparent cover (C), and the Sonex-polyester filler/absorbent cover (D) are the most effective absorbers. At high frequencies, all test absorbers are effective, with the block polyurethane filler (A) somewhat weaker at 6.3kHz. Units based on the data in this article have been used in my listening room for some time and I am very happy with the results.

TABLE 1

The table shows the decibel change for each frequency response tested, compared to the blank wall section response.

Freq.	A	B	C	D	E
Low					
200Hz	−2dB	−1dB	−1dB	−2dB	−2dB
250Hz	−2	−2	−1	−4	−3
400Hz	−2	−6	−5	−2	−2
Mid					
1kHz	−5	−9	−8	−9	−3
1.6kHz	−2	−6	−6	−5	−4
3.2kHz	−5	−11	−10	−8	−5
High					
6.3kHz	−6	−18	−15	−18	−15
12kHz	−18	−19	−17	−22	−19
15kHz	−20dB	−19dB	−18dB	−22dB	−20dB

A. Polyurethane Block/Transparent Cover
B. Polyester Batting/Transparent Cover
C. Fiberglass #703/Transparent Cover
D. Sonex Polyester Batting/Absorbent Cover
E. Fiberglass #703/Absorbent Cover

SOURCES
Crest-Foam Corp.
100 Carol Place
Moonachie, NJ, 07074
Polyester urethane foam (Polycrest model #9007), 16 x 22 x 3-inch block cost about $10 including cutting.

Illbruck
3800 Washington Avenue North
Minneapolis, MN, 55412
Sculptured polyester urethane under the tradename Sonex; the Junior package, $52.

Stacy Industries
Wood Ridge, NJ
High density polyester batting (model #970 Thermolan Plus); fifteen layers at a total cost of $13.

Owens-Corning Fiberglass Corp.
Fiberglass Tower
Toledo, OH 43659
Commercial fiberglass (model #703); six 24 by 48 by 3-inch batts, $48.

Audio Concepts
1631 Caledonia St.
La Crosse, WI 54602
Acoustical foam, $1.90/sq. ft.

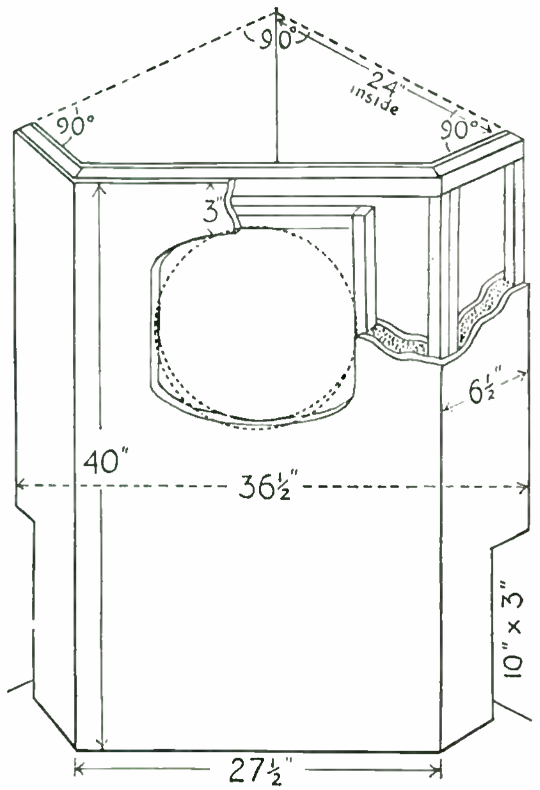
FURTHER READING

1. Everest, F. Alton, *Acoustic Techniques for Home and Studio*, Second Edition, 1984, Tab Books Inc., Blue Ridge Summit, PA, 17214.
2. Hoffman, William R., "Build a Live End/Dead End Listening Room," *Audio*, December, 1986 p. 62.
3. Davis, D., "The LEDE Concept," *Audio*, August 1987, p. 48.
4. Pisha, B.V. and C. Bilello, "Designing a Home Listening Room," *Audio*, September 1987, p. 56.

Vintage Designs

Sand-Filled Corner Panel

From G.A. Briggs, Wharfedale Construction Sheets (circa 1957). *Courtesy of Paul Penk, Santa Monica, CA.*



Distance along wall from corner to front of lid is 26¼" Weight of front panel 124 lb.

Recommended Units

W15/FS (bass only), Super 12/FS/AL or 12" coaxial full range units

This enclosure gives excellent results with the W12/FS in a budget three-speaker system, with the addition of a tweeter. The free surround 10" units also perform well here, but with 8" units the new concrete column gives a better balance.

Materials

Solid wood frame 1" thick, faced on both sides with sheets of ½" plywood. Space between plywood filled with tightly-packed dry sand. Top in 1" plywood or blockboard. For maximum bass response an air-tight fit to walls must be ensured.

Sub-baffle about 16" x 16" in ⅜" plywood should be fitted up to the rear side of the front plywood panel, inside the frame shown in drawings.

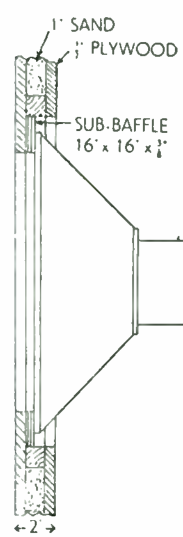


Diagram to show method of fitting sub-baffle to sand-filled panel.

TWO-SPEAKER SYSTEM

Where two speakers are used with crossover network, the treble unit should be mounted on an open baffle placed above the reflex cabinet. Plywood ⅜" thick is satisfactory and a crossover at 1,000 cycles is correct with the following sizes:

8" Treble unit baffle 14" x 12" approx.
10" " " " 16" x 14" "

Under these conditions a quarter section series network gives excellent results.

If a 500 cycle crossover is used the baffle should be 3 or 4 inches bigger.

With a 3" treble unit, the baffle size may be reduced to about 6" x 4" and the crossover frequency should not be lower than 3,000 c/s.

BEAM EFFECT

To reduce directional effect and spread the H.F. beam, the treble unit in all the above cases may be mounted horizontally with the cone facing upwards, or at an angle of 45° facing into the corner of the room for good reflection effects.

MATCHING A SUBWOOFER (OR TWO) TO YOUR SYSTEM

BY JAMES T. FRANE

"You need only one subwoofer because there is no stereo information below 100Hz."

"Place a subwoofer anywhere because low frequencies are nondirectional."

"Use it as a coffee table."

So say some self-proclaimed subwoofer authorities. Are these statements true? Do they have any basis in fact? Using my experiences with subwoofers, let's explore these "axioms" and set some guidelines about subwoofer integration in a home stereo system.

Most rules of thumb, home remedies, and old wives' tales have some origin in fact; through individual experience and/or observations. But often the expert presents a plan, a formula or procedure without identifying all of the essential parameters. The caveats are missing: limitations affecting the procedure's applicability and performance, and the parameters that must be satisfied for the recommendation to be valid.

For example, let's examine the opening paragraphs. The music we enjoy at home most often is from an LP record, tape or compact disc. Generally, records have decreasing channel separation at the lowest frequencies (below about 100Hz). Channel separation in tapes may be greater than that of records at lower frequencies. CDs have the capability, and usually the presence, of a greater amount of channel separation over a wider frequency range than either tapes or records. If you listen mostly to records, one subwoofer may be an acceptable compromise. More about this later.

It may be possible to locate a subwoofer remote from the satellite speakers and have the sound source not be apparent. This can occur because the bass frequencies tend to radiate in all directions from a speaker. There is a problem of alignment by time, though, when you locate the subwoofer and satellite speakers in different parts of the room. The electrical signal from the amplifier will reach the voice coils of all of the drivers at about the same time, causing them to move. The speed of sound is independent of the frequency of the sound. If each driver can physically respond equally well to the electrical signal, the respective speakers will move air at the same time.

Any article placed on the subwoofer used as a coffee table may resonate at some frequency that the subwoofer produces.

For the sound from the various drivers to reach the listener's ears at the same time, the voice coils of the satellites and subwoofers should be in the same physical plane.

It is possible to locate a subwoofer at a distance equal to that of the satellite, but in a different direction. In such a situation, the arrival time to each of the listener's ears will not be the same unless the subwoofer is located directly behind the listener. Thus, the subwoofer location would probably be discernible unless it was reproducing only the lowest octave.

As for coffee table use, there may be problems, in addition to differing sound arrival times. Any article placed on the subwoofer used as a coffee table may resonate at some frequency that the subwoofer produces, thus adding its own sound to that of the speaker.

CHOOSING YOUR SUBWOOFER. You can purchase satellite/subwoofer systems from some manufacturers, or you may already own speakers to which you wish to add one or two subwoofers. It's even possible that you own a subwoofer to which you want to add satellite speakers. In any case, you will only be satisfied with the end result where the satellite and subwoofer act as a coherent and integrated sound source. This will require a seamless blending of the subwoofer output with that of the satellites.

Whether you choose to go with one or two subwoofers depends on your personal preference after auditioning a system. The amount of output from the satellites below 100Hz will be a significant factor in determining the acceptability of just one subwoofer. For example, if the −3dB point of your present speakers (satellites) is 38Hz and the subwoofer will primarily serve to augment the lowest octave with the satellite run full range (not cut off by the crossover below some frequency), it's very possible that one subwoofer will serve you well. The stereo information on the recording will be reproduced down to the high 30Hz range.

However, if the satellite's −3dB point is around 80Hz and you use one subwoofer, you may be missing a lot of channel separation in the lowest two octaves. For each satellite and subwoofer, there will be published manufacturers' specifications and perhaps measurement data from product reviews as well. This information will give you a starting basis for matching satellites and subwoofer.

SOUND TESTS. As with all audio equipment, measurements provide only limited information to predict how a system will sound. The most important test is listening to the system. If at all possible, cart your satellites to the subwoofer store and audition the satellites and subwoofer(s) together, using familiar music. It will also be helpful if the electronics are at least similar, if not identical, to what you are using at home.

If the satellite's –3dB point is around 80Hz and you use one subwoofer, you may be missing a lot of channel separation in the lowest two octaves.

If your satellites are too large to move, arrange for a subwoofer audition in your home, with return privileges. You need to have the option to back out of the deal if the improvement is not worth the money, or if the system doesn't sound right to you.

Listen with one and two subwoofers and decide which sounds best. Try different types of music, some with extremely low bass. Play it at your usual listening volume level and also quite loud. Perform this test both with and without the subwoofer(s). Also listen to male voices. If they have a congested or "chesty" sound there is probably too much emphasis in the 100Hz range.

If the satellites can be placed on top of the subwoofer(s) and the voice coil planes of the satellites and subwoofer(s) line up, I believe you will have the best chance for seamless integration. This setup's added advantage is the subwoofers replace speaker stands (when two subwoofers are used) for small satellites. Alternately, placing the subwoofer(s) beside the satellites may work well.

ATTENUATION AND BALANCE. If the crossover network on the subwoofer has a level adjustment for the satellite level, experiment with it. Each attenuator (level control) may be slightly different, so I recommend the following procedure to ensure channel balance is maintained when two subwoofers are used:

1. Connect a volt-ohmmeter (VOM) across the subwoofer crossover output terminals (the terminals to which you will connect the satellites). Set the meter to read AC voltage at a low level; i.e., if a 3V reading deflects the needle nearly full scale, then ½V increments will be easy to read.
2. Connect your amplifier and a white or pink noise source to the subwoofer crossover input terminals (FM interstation hiss is a handy white noise source).
3. Adjust the crossover attenuator to the maximum setting.
4. Adjust the amplifier volume control to give a reading of 3V on the VOM scale and mark the attenuator position.
5. Decrease the attenuator to obtain a 2.5V reading on the VOM and mark that position.
6 Continue this process and mark each ½V increment down to a reading of 0V on the VOM.
7. Repeat this process for the other subwoofer.

The crossover attenuators will give exactly the same amount of attenuation to each of the satellites when they are set at the same mark for each channel. Now you can experiment to obtain the best subwoofer/satellite balance and still maintain channel balance.

An optional method is to connect a power level meter to the satellite terminals and use the same method, reading watts from the power level meter.

MORE PARAMETERS. Much has been written about the speed of attack of different types of speakers, with panel speakers (ribbon, electrostatic and planar) generally being identified as faster than most cone/dome type speakers. A faster transient response can occur when the moving mass of the driver is less and/or the magnet for the voice coil is more powerful. Some subwoofers will have a faster transient response than others, and thus may more readily blend with a panel speaker.

Panel speakers are dipole radiators, meaning that there is equal sound output from front and back. Their performance is influenced by room boundaries because of reflected back radiation and its effects. This back radiation lends a distinct quality to the sound of panel speakers. There are some dipole radiator subwoofers now on the market. Trying one may make the task of mating a subwoofer to a panel speaker easier. Again, careful listening is the key to making the best choice. There are no laws that predict what combinations may work well together.

Changing the listening room environment and speaker placement can create so many differences that accurate predictions may not be possible.

If your subwoofer has a passive radiator, you can alter the bass quality: add mass to the passive radiator to tighten the bass or remove mass for a warmer bass. Also, as with any speaker, you can alter a subwoofer's performance by changing its position relative to room boundaries. In-phase reflected sound will reinforce the output. Out-of-phase reflections cancel the sound—when the distance from the driver to the room boundary is equal to one-fourth the frequency's wavelength; the reflected sound will be 180° out of phase with the direct sound.

Low frequency reflection/absorption will be affected more by the room boundaries' mass than the surface covering. Concrete, stone, or brick reflects more bass than wood frame walls.

If the above are not an adequate number of variable parameters, the use of an electronic crossover, with its attendant second power amplifier will allow you to readily adjust the crossover frequency between satellite and subwoofer. The crossover frequency, and whether the satellites are permitted to operate full range will play a part in the sound. If the satellites can operate full range at your loudest listening levels, without the bass being distorted, they probably will work well with the subwoofer(s) in this configuration. You can alter the low frequency cutoff of the satellites by adding capacitors in series with the crossover output to the satellite. Capacitors act as high-pass filters, limiting the low frequencies to the satellites.

The subwoofer manufacturer can probably make initial set-up recommendations to give you a start in the right direction. You can then experiment with placement and level control variations.

Tools, Tips & Techniques

Taking A Stand

By now many readers are aware of the benefits of using carpet piercing spikes underneath their loudspeaker or supporting base. The improved rigidity enhances imaging, bass impact and overall clarity. Another benefit is cosmetic, as the carpet piling is not flattened and previous loudspeaker positions are less obtrusive.

I've discovered a further improvement over (or should I say under?) spikes. I screwed Robertson (square) head wood screws directly through the carpet into my wooden floor, using the corresponding spike positions of my loudspeaker base. Then I simply placed the spike tips into the screw heads. Make sure you do this only after you have experimented and are sure of the optimum loudspeaker position.

The result is an exceptionally rigid mounting for your loudspeaker, even more than spikes alone. The wood screws may be removed later and leave no tell-tale marks on the carpet.

If you have empty columns or chambers in your loudspeaker base, fill them with lead shot or sand. The added weight lowers the center of gravity, which also adds to the overall rigidity of your system.

The coupling between the base and the loudspeaker is equally important. Ideally, you should rigidly couple the enclosure to the base using wood screws or bolts into t-nuts; or, install a second set of spikes and install them pointed up into the enclosure. Don't use rubber feet or isolation pads on the enclosure/base interface. This decouples the loudspeaker from the base and nullifies any rigidity which is the desired goal.

The end result of your labor is better sound. That speaks for itself.

Carmine Gitto
Thornhill, Ontario

"Blind" Testing

I began connecting my CD player directly, bypassing my preamp, because I thought it sounded cleaner. Still, I could not verify this distinction because of the elapsed time interval required for connection changes.

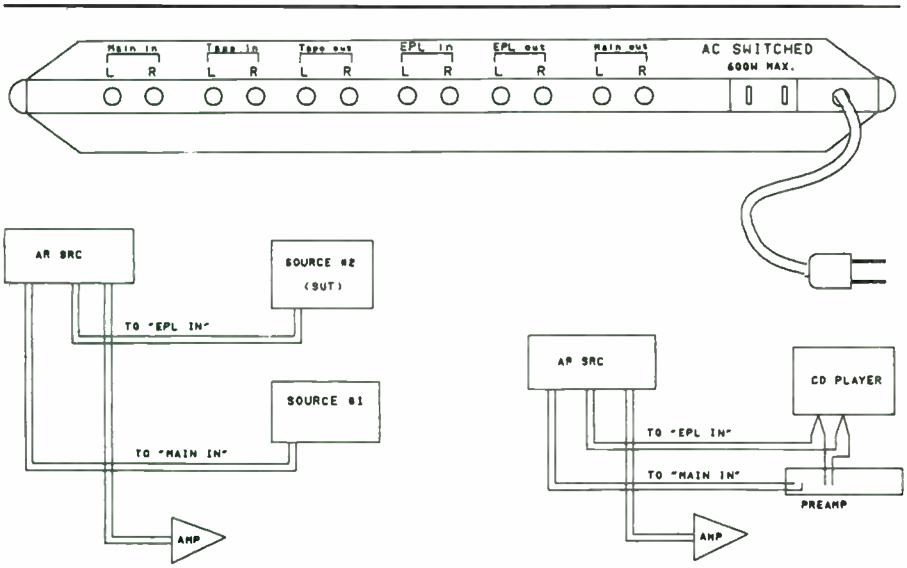


FIGURE 1: The back of the AR-SRC and typical SUT/reference comparison connections.

I often make A/B comparisons of equipment, but I think the interruption to switch cables or buttons makes my conclusions less than valid.

The AR-SRC is an add-on, infrared remote control with off/on switch, level, mute, balance and other functions. It has a precisely matched 1½dB step attenuator, claims a THD less than 0.01%, and includes a remote switchable external processor loop and a manual switchable tape loop.

By connecting the subject under test (SUT) to the "EPL In" and the reference to the "Main In," I can switch back and forth from my listening seat. The levels of the SUT and reference must be matched with surprisingly great precision to avoid unfair "clues." Differences of ½dB can cause audibly discerned "preferences." A sound pressure meter is very useful, particularly when FM interstation noise is used as a source for level balancing. Radio Shack sells one for about $30.

Once the levels are matched exactly, the attenuator of the SRC can be used to vary the level of listening without affecting the relative levels of the reference and the SUT. The EPL button on the remote toggles between the two. A few rapid taps and you lose track of the connections. Listen with closed eyes as you toggle and when you think you hear a difference, a red light on the SRC will advise you which source is connected. I repeat this several times to establish whether there is a valid audible difference.

At reasonably loud listening levels, using my blind test method (see *Fig. 1*), I was not able to confirm that the preamp detracted from the sound quality of my CD. At louder than comfortable levels, I was able to verify my direct connection preference.

After you are through with your testing, you may find the remote control is a useful addition to your system in its intended modes of operation and well worth the price. It is available from Crutchfield: 1-800-446-1640, $99 plus $2 processing charge.

James Bock
Swan's Island, ME 04685

Useful Board

In my never-ending search for the perfect enclosure material, I found some interesting stuff in a building supply store: a very dense particle board, 1⅛" thick, that is intended for interior stair treads. Its only drawback is the width. When its rounded edge is ripped off, the width is reduced to 10¾". Although not wide enough for some applications, it is perfect for smaller cabinets such as my pair of small two-ways. It has a very smooth finish which works well even with my quick-and-dirty black paint style.

Dave Schneider
Marion, IA 52302

Sizing Those Clearance Holes

Average builders of prototype or hobby equipment rarely have access to the variety of tools available to the production facility. It often means having to "make do" with whatever is available from Sears or the hardware store.

When selecting a drill size for a machine screw clearance hole, however, you can take advantage of the regularity in screw body diameters to select a fractional size drill bit.

Here is how to find a fractional inch drill size for number size machine screws. The body diameters of US machine screws are described with a number. For example, a screw could be described as "6/32 x ¾ BHMS". This means the screw has a Binder Head (BH), is ¾" long, has 32 threads per inch, and has a body diameter number 6. There is a simple linear relationship between body diameter and number size:

$$\text{Diameter} = 0.060 + 0.013 \times N$$

(N is the screw size number, and the number diameter is given in inches.)

The table below gives the diameter for even-numbered screw sizes. (Screw sizes with odd numbers do exist, but are rarely used.)

Screw Size	Diameter
0	0.060
2	0.086
4	0.112
6	0.138
8	0.164
10	0.190

(I continue to be amazed at the number of machine designers who have never encountered this simple formula. If you are a young engineer and are desperately in need of credentials with designers and the guys in the model shop, *memorize this rule*. Your co-workers will be dumbfounded at your apparent total recall of this mass of data.)

A clearance hole for a screw must be slightly larger than the body diameter, and the clearance (difference between body size and hole diameter) should increase with each increasing screw size.

The following rule will help you select a fractional drill bit for a machine screw clearance hole:

$$\text{Drill Size in 64ths} = N + 4.$$

That's all there is to it. Here are the results:

Screw size	Dia.	Drill size			Clearance
0	.060	4/64	1/16	.0625	.0025
2	.086	6/64	3/32	.09375	.00775
4	.112	8/64	1/8	.125	.013
6	.138	10/64	5/32	.15625	.01825
8	.164	12/64	3/16	.1875	.0235
10	.190	14/64	7/32	.21875	.02875

By following this rule, you'll have a clearance of several thousandths for the tiny #0 screw. The clearance will be just over 1/64 of an inch for a #6, and will be nearly 1/32 of an inch for a #10.

I have used this rule in my own work for some time, and it has served me well.

Arthur Hansen
Oak Park, IL 60304

dmp COMPACT DISCS

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DMP recordings are made with careful consideration to all elements in the recording chain. The recent acquisition of custom-designed, Class-A recording electronics (from Mark Levinson's Cello Ltd.) has taken us even further in providing our listeners with CDs characterized by unique musical accuracy, steep wavefronts of sound and high frequency transients. In short, DMP recordings extend the limits of the CD listening experience. However, there is more to the DMP philosophy than simply advanced technology. In fact, Compact disc sound quality often suffers from too much technology—the "more is better" syndrome of the recording industry today (i.e., more mics, more tracks, more processing). At DMP we have returned to the basics, limiting all elements in the recording chain to the critical essentials and recording live to digital two-track.

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CD-449 *Liquid Silver* Andy LaVerne. Andy LaVerne (piano), John Abercrombie (guitar), Eddie Gomez (bass), Peter Erskine (drums), the Essex Quartet (strings). *"...subtle, quiet, inventive jazz compositions and a crystal-clear sound."* —*Audio*

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CD-453 *NY Cats Direct* John Tropea. John Tropea, Jack Cavari, David Spinozza (guitars), Warren Bernhardt, Don Grolnick, Tom McFaul, Richard Tee (keyboards), Anthony Jackson, Neil Jason (bass), Steve Gadd, Alan Schwartzberg, Lou Marini, George Young (reeds). *"What they put on CD during their sessions is one of the best jazz CDs we've got in our library."* —*Trax*

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CD-455 *Lighthouse* Bill Barber. Billy Barber (keyboards, MIDI computers). *"The sound of this disc is nothing short of immaculate...one of the best recordings of a piano I've heard to date. But the beauty of this disc is that the man behind the electronics is talented, thoughtful, and very, very real."* —*Digital Audio*

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CD-457 *Warren Bernhardt* Warren Bernhardt (piano), Peter Erskine (drums), Marc Johnson (acoustic bass), Anthony Jackson (electric bass), John Tropea (electric guitar), Kenny Ascher, Robbie Kondon, Rick Tuttobene (synthesizers). *"...a stunning piano recording, flawless playing, exceptionally well recorded."* —*Hi Fi News*

CD-458 *Neon* Flim & the BBs. Billy Barber (keyboards), Bill Berg (drums), Flim Johnson (basses), Dick Oatts (sax, flute). *"Another exciting triumph from Flim & the BBs! There's a marvelous blending of energy and gentle shimmer to this CD music. Highly recommended.* —*High Performance Review."*

CD-459 *Braziliana* Manfredo Fest. Manfredo Fest (keyboards), Portinho (drums), Cyro Baptista (percussion), Paul Socolow (bass), Roberta Davis (vocals). *"...immaculately clean, dynamically exciting sound...."* —*Audio*

CD-460 *Thom Rotella Band* Thom Rotella. Rotella (guitars), Wayne Pedzwater (bass), Mark Minchello (keyboards), Clint de Ganon (drums). *"Emotional acoustic and electronic music, combining influences of ethereal sounds, rock, jazz-fusion and pop...."* —*The New York Post*

CD-461 *Spectrum* Bob Mintzer Big Band.

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Craftsman's Corner

Office System With Style

Since I discovered *Speaker Builder*, I have built numerous speaker cabinets, normally relying on manufacturers' recommendations for driver alignment. I often change dimensions, construction details, or materials to suit my sensibilities as to attractive design or proper construction, but only to the extent that the overall design adheres to recommended parameters.

Still, I am a cabinet maker first and a speaker builder second. I am often amazed that many builder projects pay so little attention to the overall appearance of the finished system. At any rate, I thought I would share my endeavor to create a private office audio system.

I wanted extended bass response, but I definitely did not want the speaker cabinets to be large and obtrusive. This was, after all, intended for a work setting. My design uses a subwoofer and satellite arrangement utilizing a credenza to house the subwoofer, and two small satellite cabinets mounted on matching stands. The credenza, a common furniture component in the office environment, offers adequate space to conceal a large speaker enclosure. It can also double as a component rack and bookcase.

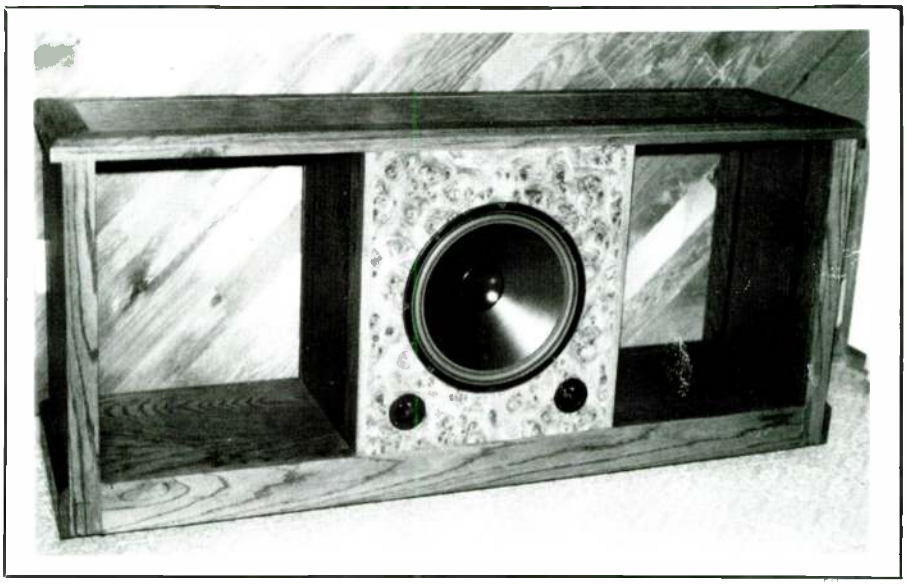

PHOTO: Back of the credenza and rear-firing subwoofer.

My existing office furniture style is early 1920s, Federal oak and I wanted my system to match. The photo shows my credenza borrows design elements from several furniture styles to form a rather traditional composite, yet it also exudes a contemporary feeling due to my choice of oak and the hand-rubbed walnut finish, which brings out the oak grain prominently. Plastic laminate counter tops with a walnut burl pattern provide a contrasting visual element, through the center pod face frame, rear panel speaker surround and top surface.

While all exposed surfaces are oak or laminate burl, the cabinet is constructed of solid oak, veneered plywood, laminate, and high density particle board. For example, the speaker pod has an inner particle board shell and an outer shell of the finish materials. The walls of the pod have a minimum cross section of 1½". I despise flimsy, commercially made furniture, and my cabinets are certainly stout. A little extra mass in a speaker cabinet doesn't impair clean sound reproduction.

The completed credenza weighs 165 pounds and I mounted piano casters underneath, hidden behind the base trim, for easier movement. An additional caster under the center pod and a Douglas fir ladder frame base reinforces the cabinet, to support the considerable weight of the center pod. The satellite speakers and sand-filled stands are not exactly lightweight, either.

This type of construction is expensive and very time consuming, but I think the result is worth the cost.

One tip: I ensured against buzzing of the adjustable shelf tracks in the two side bays by lining the gap between the track and the wood with rubber weatherstripping and attaching the shelves to the cabinet frame with both screws and two narrow beads of silicon rubber along the tracks' sides and groove. The weatherstripping exerts pressure against the repositionable shelf supports.

The satellite and subwoofer enclosures are acoustic suspension designs: a 50% filled, second-order, 0.5 cu. ft. satellite and a 3.5 cu. ft. subwoofer. Each satellite uses a Dynaudio 17W-75 woofer and Morel MDT-28 tweeter. The subwoofer is a "Sledgehammer" (Madisound 15258DVC) mounted in a rear firing position on the back side of the credenza. I chose this placement to form a Helmholtz resonator with the cabinet positioned close to a wall.

The system is biamped—a 200W per channel unit drives the subwoofer, with a 24dB/octave electronic crossover set at 120Hz and an Audio Control Richter Scale bass equalizer (the normal and inverted mono output jacks are very handy). A 35W Sony receiver drives the satellites with a passive 6/12dB, 2.4kHz crossover design, (Madisound 212-24).

I routed the preamp output signal to the processor for frequency division, and then brought the high-pass signal back into the receiver for amplification; the low-pass signal goes on to the separate power amp. My 13 year old receiver features a normal/separate switch for looping line level signals in and out ahead of the power amp section. I think more current receivers should include processor loops. My single unit controls volume, tone and signal selection. I dislike commercially available subwoofers with separate volume controls that require constant adjustment.

My costs totalled $799 for cabinet materials, drivers and passive crossovers, and the Audio Control unit. I built the power amp from a kit, with modifications, which turned into a construction adventure in itself (over $500 and three months extra labor).

How does it sound? It compares favorably to my four-way, five-driver, floor model monitors in my home system. My office system sound suits my needs, did not cost a fortune and I have the satisfaction of knowing it was my skill and hard work that produced the design.

Cliff Sommers
Coeur d'Alene, ID 83814

Technology Watch

Plasma Loudspeaker System

By Peter Muxlow

No discussion to date has appeared in *SB* on a do-it-yourself "Plasma" loudspeaker, so this may be enlightening for most readers.

A plasma loudspeaker produces sound by means of an ionized gas channel. The current flow through the ionized gas—air, is modulated. This modulates the plasma temperature, which in turn causes pressure changes within the air inside the plasma, creating sound waves.

Since there are no moving parts, the plasma system doesn't suffer the normal limitations of other types of speakers.

There are three methods used to create the ionized gas channel:

- RF corona
- DC corona
- DC glow.

The RF corona is the basis of the Magnat Tweeter[1] invented by Klein (*Fig. 1*). It has two electrodes, mounted in a quartz tube to withstand the 1000°C temperature. An RF oscillator operating at 27MHz is applied between the electrodes, which produces a corona discharge. The oscillator is amplitude modulated to create the sound. The RF corona has two disadvantages[2] the creation of ozone and the difficulty of reducing the amount of RF leakage from the oscillator.

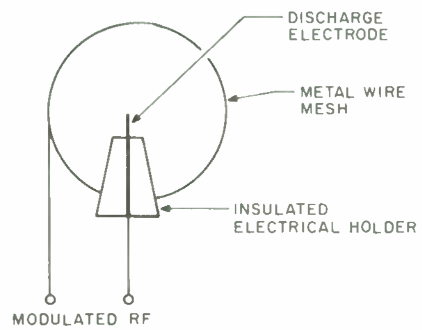

FIGURE 1: The RF corona.

The DC corona loudspeaker[3] applies a high DC voltage, 35kV, to a number of needle cathodes which points at a mesh anode (*Fig. 2*). This creates a corona which is modulated to create sound. No commercial loudspeaker is made using this principle.

1. *Acustica* 4, 77-79 1954; US patent #4464544, 4316120; British patent #756546.
2. *HiFi Answers* August 1984.
3. *Journal* Acoustical Soc. 54 494-498 1973.

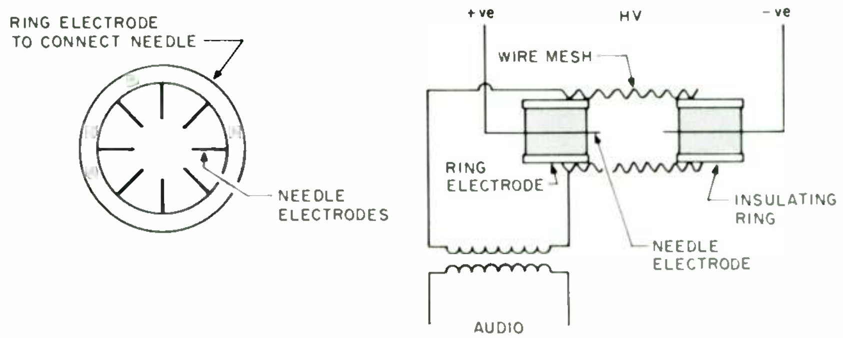

FIGURE 2: The DC corona.

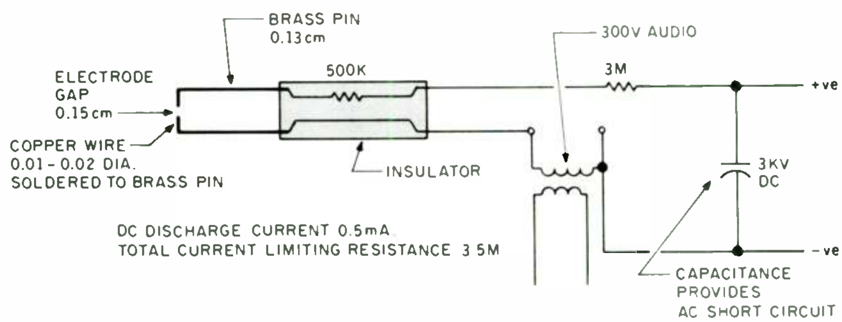

FIGURE 3: STL Ionophone; DC discharge current 0.5mA, total current limiting resistance 3.5M.

The DC glow discharge uses (current limited) air breakdown to create the ionized channel. The current limiting is to prevent the channel from arcing. It differs from the nornmal RF or DC corona[4] by having a consistent ionization spread between the electrodes and a more homogeneous electric field. More importantly, the glow discharge operates at a lower voltage, produces less ozone and is quieter.

The Hill Plasmatronic[5] operates on this principle.

A simple DC glow source can easily be made as described in *Fig. 3.* This is the STL Ionophone[6] originally developed as a simple point source by Fransson, for the measurements of acoustics inside a flute. The sound output is very low but it is a good experimental introduction to plasma loudspeakers.

4. *Journal* Acoustical, Soc. 81, 1972-78, 1987.
5. US patent #4219705; British patent # 2023373.
6. *Journal* Acoustical, Soc. 58, 910-915, 1975.

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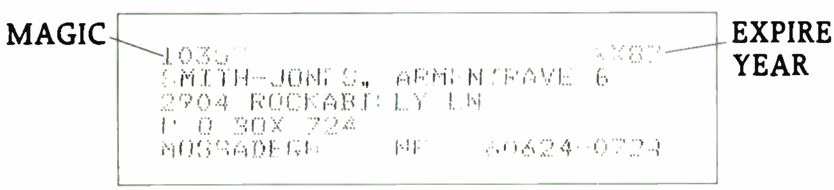

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Book Report

Reviewed by Gary A. Galo

Modular Map

The Brady Guide To Microcomputer Troubleshooting and Maintenance, *by Henry F. Beechhold.* Brady, div. Prentice Hall Press, New York, NY; First Edition, 1987. Softbound, 324pp., $17.95.

Computer users in need of nontechnical troubleshooting information will find Brady's guide extremely useful. Henry Beechhold also authored *The Plain English Repair and Maintenance Guide for Home Computers.* Plain English may seem uncommon in the computer world, but Beechhold appears to have gained his technical background without sacrificing his formal language skills. Beechhold is an English professor and chairman of the linguistics program at Trenton State College. He communicates his ideas clearly and succinctly, a rarity in the field of computer documentation. I wonder if he would consider writing manuals for Zenith and Microsoft. They could certainly use his help.

The graphics, on the other hand, are not particularly impressive. All of the illustrations are hand drawn and many are rather sloppy. Tab's illustrations appear high-tech compared to these. I am surprised at this approach, especially for a computer book.

Beechhold's refreshing introductory chapter, "Where We're Going," gives a few brief comments about his background both in electronics and his "official" field, stating we should not accept our professional categories, or formal education, as limitations. Beechhold then outlines his book's purposes, the manner in which the material will be covered, and concludes with a few questions and answers the newcomer to computer repair will find helpful. He believes the best insurance against potential computer troubles is protection from power line problems and static electricity. I could not agree more. He gives sources for equipment designed to deal with such problems.

Beechhold's Troubleshooting Road Map consists of two, large, removable cards which list common symptoms and suggested places to look for the cause of the problem. You may then refer to specific sections for more detailed information.

His second chapter gives a more thorough introduction to the troubleshooting process. Beechhold points out the value of knowing the limits of your expertise, and recommends professional help when you reach those limits. He attempts to take readers from the "black box" level into knowledge of actual components of the computer.

Beechhold believes in preventive computer hardware maintenance and devotes a chapter to care and cleaning, including disk drives and printers. He also covers storage media and care of software. Throughout, he emphasizes prevention, discussing causes for common problems, and recommending cures.

If you have never used a soldering iron or a voltmeter, his chapter, "Tools and Techniques," is especially useful. The author discusses proper soldering and desoldering techniques, and introduces the use of test equipment for measuring voltage, resistance and current. Beechhold also discusses the usefulness of technical reference manuals, and provides an introduction to reading diagrams, both the "block" and "schematic" types, with illustrations of common schematic symbols.

For readers with no electronics background, the chapter, "How Computers Work," covers the obvious, and gives a clear and understandable introduction to basic electricity as well. Beechhold explains the essential elements of a computer system, communication between the various parts of the computer system, including serial and parallel interfacing, protocols, and data conversion, and introduces basic terminology.

Beechhold devotes a chapter called "Making Changes" to upgrading or enhancing your computer system's performance. He discusses everything from adding disk drives and expanded memory, to CPUs and coprocessors. He recommends the NEC V-20 and V-30 chips, and shows the proper techniques for removing and inserting socketed ICs. CD ROM is also discussed. Five pages of dealers who sell peripherals and enhancements is contained at the end of this chapter, but there are several notable ommissions, including JDR Microdevices and Jameco. I recommend regularly consulting *Computer Shopper* for this information.

Beechhold's technical reference listings include ASCII codes, microprocessor pinouts and numeric conversion tables. A glossary of computer terminology and abbreviations, such as RAM, ROM, LAN, and so on, concludes the book. The book is well indexed, so you can quickly find information on specific topics.

The Beechhold book is most useful if you need to troubleshoot computers at the modular level. Although the author discusses removing components from circuit boards, he does not give enough background to educate the beginner in troubleshooting at the component level.

Beechhold's clear and well-organized writing style is a great asset. His coverage is generic, that is, not oriented toward any specific computer standard. Since the discussion concentrates on isolating the defective module, as opposed to the defective component, it will be most valuable to users of open architecture systems. I recommend the book for this type of information. Computer users at a more advanced level should consider the following Margolis book.

Chips and Charts

Troubleshooting and Repairing the New Personal Computers, *by Art Margolis.* Tab Books, Blue Ridge Summit, PA; First Edition, First Printing, 1987. Softbound, 400pp., $17.95.

Art Margolis is well known to electronics professionals and enthusiasts for his many books and articles on repair and maintenance of electronic equipment. This Margolis book is technical in nature and, in many ways, picks up where the Beechhold book leaves off.

Margolis does not limit his analysis to IBM computers and compatibles; he also covers the Macintosh, Apple II, Amiga, Commodore 64 and Commodore 128. IBM's PS/2 line is not included since the book was published prior to their introduction.

Margolis assumes you have a working knowledge of basic electronics and computer terminology, and uses numerous illustrations; nearly every page contains at least one diagram, photo or chart. Margolis surveys the most common computer ailments, gives a detailed introduction into the workings of the video display, and shows how the display itself can be used as a diagnostic tool.

Getting inside a microcomputer for the first time can be a challenge. Margolis illustrates several popular computer case styles, and give instructions for proper disassembly. The simplest repairs, such as blown fuses, loose connections, and so on, often require little knowledge of how the computer operates. The first seven chapters cover easy repairs, while the remainder concentrate on problems requiring advanced troubleshooting techniques.

IBM PC users will find a useful chart listing the seven major subassemblies found in the PC and a list of problems each board can cause. From there, he brings the discussion to the chip level. Motherboard layouts for four of the most popular personal computers are particularly useful. All chips are clearly labeled for the Apple IIe, IBM PC, Tandy TRS-80 Color Computer and the Commodore 64. He devotes an entire chapter to explain pin connections for large chips.

Margolis gives an excellent introduction to the internal operation of static and dynamic memory chips. He also discusses ROM chips. Pin-outs are given for the most popular chips, and the author explains how these chips interface with the rest of the computer. He gives similar coverage to the vast array of support types, which he calls "primitive" chips.

Margolis introduces the various pieces of test equipment you need to service computers at the component level, and explains how to make a Manual Fault Dictionary covering each chip in your computer. I think he may be a bit optimistic that this will only take an hour, even if there are a couple hundred chips on the board. He also discusses how to use diagnostic programs to isolate problems, including memory, I-O and video tests.

If the prospect of desoldering and replacing the new surface mount chips intimidates you, Margolis' suggestions on the correct methods for changing chips are helpful. His block diagram and explanation of the typical microcomputer is a good introduction to its major functions. He discusses the computer clock in depth, including how it affects the rest of the system.

Margolis examines several common 8- and 16-bit microprocessors in detail, including the 6502 used in the Commodore 64, the Motorola 68000, and the Intel 8088 and 8086, an excellent introduction to the workings of microprocessors. The chapter on the address and data buses clarifies how the microprocessor communicates with the rest of the computer. Unfortunately, Margolis does not cover the Intel 80286, but I was especially disappointed because the summary on the book jacket's back cover states the book covers the IBM AT, which in fact, it does not.

The chapter on memory mapping is fairly generic. I would like to see more specific information on 8088-based systems. The author provides exceptionally detailed explanations of serial and parallel interfacing, D/A and A/D converters, and cassette and joystick interfacing. He explains the IBM Monochrome Adapter and Color Graphics Adaptor in detail in his chapter on video interfacing. He does not discuss the Hercules or EGA standards. I am also surprised Margolis does not discuss hard disks in the disk drive chapter; only floppy disks are covered.

I am a bit disappointed with his power supply information. He confines his discussion to conventional regulated supplies, and does not cover the switching supplies used in the IBM computers and compatibles. He states "power supply troubles are fairly easy to diagnose," which is certainly true for conventional supplies, but the switching supplies can be more difficult. A short chapter on safety, warning technicians about the potential dangers of high voltage, X-radiation, and CRT implosion, concludes the book.

Despite my reservations regarding a few items I wish Margolis had included, I found this book worthwhile, and I recommend it. It is well-written and logically organized. The generous illustrations are a great help. If you have electronics experience, but are new to microcomputers, this book offers a tremendous amount of knowledge about the internal workings of your computer, even if you have no intention of servicing it yourself. I suspect most *SB* readers will find this book more in line with their level of understanding than Beechhold's book.

SB Mailbox

CORRECTION

I'd like to correct an error in a formula from my article "Loudspeaker Cabinets," (*SB* 2/88, p. 26). The formula should be:

$$F_r = \frac{12}{2\pi}\left(\frac{D}{P}\right)^{1/2}\left(\frac{7}{2}\right)\left|\frac{1}{a^4} + \frac{4}{7}\times\frac{1}{a^2b^2} + \frac{1}{b^4}\right|^{1/2}$$

$$D = \frac{Et^3}{12(1-c^2)}$$

where

D = bending stiffness
t = panel thickness
P = density kg/cubic meters
E = Young's Modulus
v = Poisson's Ratio
a, b = panel length and width
c = velocity of sound

Peter Muxlow
Wellington, New Zealand

MINIMUS CORRECTION

Regarding William Hoffman's article, "Modifying Radio Shack's Minimus-7," (*SB* 1/88): *Fig. 5* shows R1 is 82Ω, 1W; it sounds like too much. *Figure 7* shows only an R2; is 82Ω correct?

Tom Wheeler
Alma, AR 72921

William Hoffman replies:

Yes, the value of R1 in Fig. 5 *is 82Ω,* and R2 in Fig. 7 *is this resistor, incorrectly labeled. Thanks for spotting the error.*

"INCREDIBLE" UPDATE

Our recent article "Incredible Bass Performance from a Two-Way Design" (*SB* 1/88) identified an anomalous "third" impedance peak at about 125Hz (see *Fig. 6* and *Appendix 4*). To isolate the source of this peak, I mounted an extra woofer backwards, outside an unfinished fat box cabinet. I used no crossover, tweeter or stuffing. This allowed me to sweep the frequences in the vicinity of the third peak and observe the effects as I performed modifications, without having to demount the woofer (see *Fig. 3*: f_s Measurement Setup).

We've decided at this time that bracing the back of the woofer, as described in the article, is too cumbersome and labor-intensive, especially inside a finished cabinet.

Initially we thought the source of the peak was woofer basket ringing, but tests showed instead the magnet assembly was responsible; the 125Hz impedance decreased by up to 1Ω just by touching the assembly's rear plate with a finger. I firmly grasped the ceramic magnet about its circumference with both hands and impedance dropped from 6.6Ω to 4.4Ω.

To eliminate the ringing I stuck a glob of plumber's putty on the back plate of the woofer. Recall that we used this same material to add mass to the tweeter plate. Voila, total nullification of the resonance.

To stabilize the putty over time, I found a plastic plumbing cap at a building supply store, 3⅜" inner diameter by ⅛" thick by 1⅝" deep. I kneaded a ⅞" thick dollop of putty into the bottom of the cap, placed it over the magnet assembly and tapped gently to ensure good coupling with the magnet. The cap sides cover all but ⅜" of the magnet. Finally, I ran a ¼" bead of white silicon tub caulk around the free edge of the cap to seal it.

We tested this arrangement to our satisfaction, then compared the sound. The modified system showed a dramatic improvement in clarity, not only in the 125Hz region, but across the band. Bass is cleaner and voice is more focused. There is no more slight chestiness and fogginess in the lower registers of Neil Diamond's voice. Remarkably, the transient attack and decay of cymbals and triangles is noticeable. Perhaps intermodulation distortion is reduced when magnet assembly ringing is nullified. The overall neutrality of sound clearly justifies this effort.

Note: Of the four varieties of the Radio Shack #40-8053 woofer we are aware of, only two are suitable for this design. The first version, which appeared in 1983, has a 3⁄16" diameter vent from the back magnet plate to the area behind the porous dustcap. The woofer's Thiele parameters are unsuitable in our design and it will produce loose mushy bass and lacks clarity across the range.

We used the second and third versions (rated at 50W and 75W RMS) for our design. Specs are as given in the data sheets: Q_{ts}, 0.38; V_{as}, 4.7 cu. ft.; and f_s, 27Hz.

The 1988 Radio Shack version's data sheet lists Q_{ts}, 0.41 and V_{as}, 4.87 cu. ft. This requires an additional 0.6 cu. ft. to realize similar theoretical bass performance, and there may be other characteristics that make the latest woofer version unsuitable. I wrote Radio Shack (Canada) for an explanation, but received no response. I am also unsure why this driver is not available in the US.

So, use caution when you buy your woofers. Make sure there is no vent; some models have a recess that looks like a vent—use a toothpick to test for patency, and check that your woofers' parameters are as required for "incredible" results.

S. Wayne Cox
Campbell River, British Columbia

RCA THEATER HORNS

I have been searching for design information on the old RCA Bathtub low-frequency horns. These were used in the many theaters across the nation during the forties and fifties, as the low-end of a two-way system. The LF horns each contained two 15" drivers, model number PL304. There was also a single speaker version, model LC-9. This is all the information I have, but I'm hoping some reader might help me locate the designs, or better yet, the genuine system.

Todd Wilson
Canoga Park, CA 91306

Contributing Editor Bruce Edgar replies:

I do not have any data on the RCA horns you mention. Perhaps a reader who worked in theater sound during the forties might be able to supply some details.

WOOFER MISCONCEPTIONS

I'm trying to understand what makes a given woofer a good candidate for a particular box design. The EBP ratio (F_s/Q_{es}) in *Loudspeaker Design Cookbook* caught my attention and prompts this letter. I'd like a qualitative explanation why this ratio is useful in deciding between closed- and vented-box designs. For EBP less than 50, why should I infer a closed box; while greater than 100, a vented box? Also, what if the value comes *between* 50 and 100?

Vance Dickason points out the alpha must be three or greater for a closed-box design to be considered acoustic suspension, rather than infinite baffle. Is the distinction between these categories simply a matter of the size of the restoring force of the air trapped inside the box?

Should I try to stay in the acoustic suspension region when designing closed box systems? If so, then Q_{tc} will have to be at least twice Q_{ts}. A look at numerous woofers whose EBP is well under 50 reveals Q_{ts} values of 0.4 or greater. How then, do I go about constructing an acoustic suspension system that is transient perfect (Q_{tc} = 0.5) or even maximally flat (Q_{tc} = 0.707)?

How should I use relatively high Q_{ts} woofers, say 0.6 or 0.8? According to Mr. Bullock, they should not be considered for vented systems (Q_{ts} lower than 0.4 is suggested); Mr. Galo suggests Q_{ts} lower than 0.5 for transmission lines. If the only choice is acoustic suspension, then I end up with Q_{tc} of 1.2 and 1.6, not a great state of affairs. By the way, these Q-parameters belong to the Dynaudio 30W100 and 24W75, two highly touted woofers. I'm puzzled that the mathematics do not seem to be favorable.

Thanks to Mr. Dickason for any help he can offer, and a very pleasurable and valuable book.

Richard Halpern
New Paltz, NY 12561

Contributing Editor Dickason replies:

Mr. Halpern asks some very good questions; the answers should help clear up some general misconceptions about closed-box loudspeakers.

Efficiency Bandwidth Product (EBP) was a sort of rule of thumb Richard Small used in a paper he delivered to an AES convention (AES Preprint 1251). Basically, he was saying drivers with Q_{ts} lower than 0.4 are generally likely to produce usable designs utilizing vented enclosures (and by "usable" we mean something with desirable bass extension in an enclosure an average-size living room can tolerate). For usable vented designs, f_s is around 100 times greater than the driver Q_{ts}.

Likewise, drivers with Q_{ts} higher than 0.4 are generally likely to produce usable designs utilizing sealed enclosures. For usable sealed box designs, f_s is somewhere near 50 times greater than driver Q_{ts}.

*Of course, given release from the constraint of "usable" designs, we can take any woofer and usually get it to work in either type enclosure. The best way to confirm this is to look at the parameters of several different designs and compare the differences between those we know work well in sealed boxes and those which we know work well in vented boxes (*Table 1*).*

Group 1 woofers are obviously designed expressly for vented enclosures. Both applications yield more or less usable box sizes, but the closed box alignments all give f_3s high

TABLE 1

			Sealed Q = .7		Vented Qb3	
Driver	**EBP**	**Qts**	**F3**	**Vb**	**F3**	**Vb**
Group 1						
Focal 8N511	147	.20	108	.27	58	.4
Precision TD205R	86	.33	69	.6	39	1
Eclipse 1038R	71	.30	57	1.3	38	2.1
Eton 11-580-	83	.30	61	1.8	33.9	2.8
Group 2						
AC-10	46	.44	37	3	21	7
SEAS P25REX	48	.49	42	4	21	13
Morel MW164	53	.54	48	1	24	7.2
KEF B200	44	.51	35*	3	17	19

*Qtc = .8

enough to be unattractive, compared to the vented alignments. Note all woofers have a EBP near our "rule of thumb" figure of 100.

Group 2 woofers are obviously intended for sealed enclosures. In this case, the vented enclosure alignments call for refrigerator-sized boxes, which makes them an unlikely choice for these woofers. EBP for this group all fall close to the "rule of thumb" figure of 50.

Your question concerning the distinction between infinite baffle and air suspension designs is indeed a matter of the size of the restoring force of the air in the box. By definition, "air suspension" means the compliance of the air in the box has to be three to ten times greater than the compliance of the woofer suspension. Should you try to design air suspension systems rather than infinite baffle systems? To answer that question requires a little historical perspective.

Although originally patented by Harry Olsen, it was Edgar Villchur who popularized the air suspension design and turned it into a desirable consumer product. Before AR and Villchur, speakers with extended bass tended to be behemoth designs which wouldn't fit into the average person's decorating scheme. A method was needed for detailed (well damped), respectable bass extension from a small package.

We enter the era of the "bookshelf" speaker. Using carefully designed woofers, low, well-damped bass became possible in box sizes of two cubic feet or less. These woofer designs (compared to the usual vented speaker) are characterized by extremely compliant (floppy) surrounds, thick, heavy cones, large magnet structures, and long voice coils. By manipulating parameters, you could design a woofer with a fairly low Q_{ts}, low f_s, and small enough V_{as} to fall into the air suspension category in the $Q_{tc} = 0.7$ range. The tradeoff is one of efficiency. Heavy cones, and long voice coils, with large portions of the coil operating outside of the magnetic gap, contribute substantially to making a woofer inefficient.

As the consumer trend moved toward more efficient designs which produced higher SPLs from medium-power equipment, without excessive amounts of clipping distortion; air suspension designs lost their popularity. This phenomenon was undoubtedly influenced by the increased popularity of loud electric music.

With Thiele/Small's "scientific" method for vented enclosures, designers jumped on the higher efficiency bandwagon, and the AS design began to disappear. With the advent of CDs, efficiency and headroom became even more important, so the inefficent, compact enclosure, became less popular. At this point, most sealed-box designs on the market are the infinite baffle variety—utilizing larger enclosures, or limiting well-damped extension to not much below 50Hz.

Let's get back to the question, "Should an amateur builder attempt AS designs?" Sure, but be aware that virtually all of the available closed-box drivers on the market fit air suspension criteria only for a Q_{tc} of 0.8 and

Continued on page 60

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higher, with correspondingly higher cutoffs, but do fit into fairly compact enclosures. To get Q_{tc} in the 0.5–0.7 range, they will all be infinite baffle systems requiring moderately large enclosures. Frankly, I don't know of any OEM manufacturer who sells a woofer which will produce a compact, damped, good low-bass extension, acoustic suspension design. They, like the dinosaur and silver certificates, have disappeared.

As to the high Q_{ts} Dynaudio woofers, the math works just fine for them. Dynaudio suggests sealed, infinite baffle boxes for both the 30W100 and the 24W75, and they perform quite well in that capacity. However, neither will perform as well in compact, well-damped enclosures.

ESL RESPONSE

From the beginning, I have been a subscriber to *Speaker Builder*. During this time I have enjoyed reading the articles and the lively discussion in the *Mailbox* department. The one exception to this has been the series of articles by that self-styled authority on electrostatic speakers, Roger Sanders.

Eight years have passed since his article first appeared in your periodical (*SB* 2, 3/80). Anyone who has been involved in a project, over this length of time, should have learned something about it. At least they should understand the fundamentals. Unfortunately, his review of my book, *Electrostatic Loudspeaker Design and Construction* (*SB* 4/87), makes it clear this is not the case. Not only are many of his statements extremely biased, they contradict material contained in his own article.

As an example, consider his statement about insulating paint, "I have found insulating paint to be unnecessary." In *SB* (3/80, p. 22) he not only states that his speaker plates must be coated with an insulating paint (on both sides) but there is an entire subsection titled "Insulating Paint." He devotes several paragraphs to a discussion of its correct application and then he states "its prime purpose is to prevent a voltage breakdown in the airgap between the stators."

Although Mr. Sanders did not mention it in his review, the high voltage on his speakers is only 2kV. By using 80 mil spacers the field strength between the plates and the diaphragm is set to 25V/mil. My speakers have a 125 mil spacing and a polarizing voltage of 5kV. This combination produces a field strength of 40V/mil. This larger value makes it even more important to have the plates adequately covered by an insulated coating. Perhaps he should have spent more time reviewing his text before criticizing that of someone else.

In the paragraph about the plastic plates he continues to demonstrate his inconsistency and lack of understanding. For instance, consider the statement "the plastic stators are not very strong mechanically, requiring hard-to-build wood ribbing to support them." Before discussing this statement in some detail, it should be noted the plastic plates have a depth of 125 mils. This is six times the thickness of the perforated aluminum he used. How thick do the plastic plates have to be before Mr. Sanders considers them to be mechanically adequate?

Although he does not use a rib to support his speaker's aluminum plates, he has an acrylic insulator glued to them. He further states "these strips are necessary because the diaphragm must be supported every six inches for adequate stability."

This later statement is simply not true. In Chapter Four of my book I indicate the high voltage between the plates and the diaphragm will produce a force of attraction. As indicated, stability is a function of equal plate-to-diaphragm spacing and an adequate value for the spring rate. If all the surfaces were perfectly parallel the two plates would exert an equal and opposite force. Under this condition the diaphragm would remain centered.

If, as is often the case, one of the plates has a bow in it (either in or out) or if it is not parallel to the diaphragm, there will be a difference in the applied force. The diaphragm will, therefore, pull toward the plate with the greatest force.

From a construction standpoint it is difficult to obtain a system where all the spacing values are exactly equal. To overcome this problem a tension must be applied to the diaphragm. The applied force needs to be only a value that is adequate for maintaining the diaphragm in its centered position. In effect the tension on the diaphragm produces a result similar to the properties of a spring. A movement in a spring requires the use of some additional force. When the force is removed the spring rate returns the spring to its normal position. This is also true of the diaphragm. The additional force is supplied by the audio signal. When this signal is removed, the correct spring rate will force the diaphragm to return to its centered position.

Mr. Sanders' stability problem is not related to the support for the diaphragm. In fact, the insulators are a stop-gap solution created by someone who doesn't understand the basic problem. What he should do is to first make sure both plates are flat and parallel to the diaphragm and then determine how much tension must be applied so the diaphragm is stable under the static condition. When this is completed and the correct spring rate has been achieved, the diaphragm will stay centered and his stability problems will be gone. This will also eliminate the need for making adjustments after the speaker has been assembled.

Unfortunately, the reviewer seems to

think only in terms of his application. Because his speakers are limited to midrange and tweeter applications, placing the insulators on the inside does not seriously affect acoustical performance. However, when the response is extended to the lower regions of the audio spectrum, the diaphragm must be allowed to move the maximum amount. My speakers have a peak-to-peak amplitude of 0.25 inches.

In addition, when the ribs are placed on the outside of the plate they become an integral part of the support structure. This strengthens the entire assembly. With a little care, during the cutting and assembly operation, it also provides a rigid and flat surface for mounting the plates. This technique also allows the plates to be checked for flatness before they are assembled into a speaker.

While my book describes the use of a stretching frame, to tension the diaphragm, it should not be construed as the only possible method. In fact, even though Sanders makes a big issue of it, he uses both a stretching frame and a heat gun. If he had carried his technique one step further and measured the diaphragm tension while applying the heat he would have a very viable solution for obtaining a stable diaphragm.

He also says my speakers will have low sound pressure levels. He states many readers will be disappointed in this regard, and attributes this loss to the use of large perforations, electrical segmentation, stray capacitance and a large diaphragm-to-stator spacing.

Such a statement only serves to exemplify his lack of knowledge concerning the parameters that affect the performance of an electrostatic speaker. For instance, large perforations are supposed to lower the output. In actual practice the size of the holes has nothing to do with the acoustic level. What is important is the percentage of open area. In this case the value is 50% only slightly more than the 40% used by Mr. Sanders.

If the length and width of the hole was a factor then the speakers described by both Malme and Matthys (see Chapter 15 in my book) would not work. Both use wire grids for the plate. In each instance the slot length and the spacing between the wires is greater than the holes in the plastic plates. For those readers that are interested, the Malme article also gives some guidelines for the spacing.

Besides the large perforations, Sanders would have us believe segmentation will also reduce the output level. This is another of his many opinions that has no basis in fact. Early researchers in the sound field established a correlation between the size of the speaker and the maximum wavelength of the sound that it must produce. To prevent beaming, a speaker's size must decrease as the frequency is increased. As indicated in Olson's *Modern Sound Reproduction* (p. 31) the required speaker cone or diaphragm movement also decreases with increasing frequency. In this way a multi-way system is able to maintain the same acoustic power level over the entire audio spectrum. It is also the reason why the speakers are segmented.

The next item on his list was stray capacity. Many articles on electrostatic speakers have covered this subject in some detail. They usually indicate a speaker will have some additional, unusable capacity. In most electrostatic speakers the parallel structure created by the plates and the diaphragm is extended over the spacers. Since the diaphragm in this area cannot move, the capacity between it and the plate is wasted. The speakers described in my book do not have this problem because the plates are plastic and no conducting material is applied to the spacer area. Without this there is no additional capacity. They will, therefore, have a lower value of stray capacity than the speakers described by Sanders.

This brings us to the last item, the large diaphragm-to-plate spacing. My speakers were specifically designed for a large diaphragm movement. This is a requirement for obtaining an adequate bass response. If Mr. Sanders had done his homework he would have realized these speakers have a high polarizing voltage and an audio transformer that has a greater step-up ratio. This combination is capable of producing sound pressure levels above 100dB. Although it was inadvertently left out, the frequency response photo shown in my book had a level of 100dB and was measured at one meter. This does not necessarily represent the maximum acoustic level. It is a value that was selected as a maximum listening level.

One other factor that needs some clarification is the use of Ivory soap. Mr. Sanders mentions that in time the coating will come off. Originally this was true. The speaker project that led to the book was started in 1968. I used a copper strip in the first speakers as a contact for the diaphragm. Unfortunately, the soap corrodes the copper which eventually looses contact with the diaphragm. After this was discovered, I changed the contact strip to a carbon material. In this case the same paint used for the conductive coating is also used to make a contact strip for the diaphragm. Over the past ten years there have been no difficulties with the operation of a dozen or so speakers built using this method.

I could continue, but this is enough about the negative aspects of his review. In some areas Mr. Sanders raises a valid concern. For instance, he mentions there is no indication of the speakers' directivity or its acoustic sound pressure level. This latter item has already been covered. Although I no longer have the polar plots that show directivity, the speakers were tested in an anechoic chamber. The results indicated the dispersion angle was 90° at 10kHz. This wide angle is a direct result of the segmentation.

In addition, Mr. Sanders makes a very good point about the missing bibliography and that none of the noteworthy articles (contained in Chapter 15) were from *Audio Amateur* or *Speaker Builder*. The bibliography was removed, by the publisher, because of editorial constraints. I have suggested it should be included in the second printing. If any reader wants a copy, I will be more than happy to send one. Please enclose a dollar (in U.S. currency) to cover the printing and handling cost. Include a self-addressed stamped envelope with your request.

Because *Speaker Builder* is a current publication, I did not deem it necessary to reprint any articles. The material I included contains useful information that has been out of print for many years.

As a closing point, I do not consider my book to be the last word on electrostatic speakers. Instead I offer it only as a beginning. I urge *SB* readers to use it or the material by Sanders as the starting point for their own electrostatic speaker projects. If they have any questions about my book or about the speakers described in it they can write to me (be sure to include a SASE).

One *SB* reader suggested starting a clearingh use for information on electrostatic speakers. [See the "Clubs" listing in the Classified Section.] This is a good idea and I believe the magazine could carry this idea one step further. Why not create a forum specifically focused on electrostatic speakers. This section could include articles on the subject and have its own letters section. I believe that this would benefit readers as well as *Speaker Builder*. This is a good time to demonstrate that an electrostatic speaker is a worthy alternative to the electromagnetic radiator.

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Roger Sanders replies:

Mr. Wagner refers to me as "that self-styled authority on ESLs." I am not a physics professor at a nationally esteemed university, but I have an extensive formal education in the sciences, have taken postgraduate courses in physics, and read the engineering texts in physics, mechanics, and acoustics. I have researched, at the California Institute of Technology Library, all the periodical literature on ESLs, obtained and studied most of the ESL patents, and spoken with and questioned most of the ESL manufacturers, prior to building my first ESL in 1972.

Since then, I have built numerous ESL designs, using my research of ESL design theory and in fact finding some of it to be incorrect, which I will point out shortly.

Through my published work, which resulted in contacts with hundreds of engineers, designers, and individual builders, I incorporated much useful information into later, advanced designs. I also invented and patented (1980) the first method of obtaining a free-standing curved ESL diaphragm, which is apparently being used commercially by both Martin Logan and Xstatic. I am not suggesting to Mr. Wagner that I am the world's leading authority on ESLs, but I have indeed done my homework and know what I am talking about, contrary to his assertions. Perhaps he cannot "enjoy" my articles because I write in layman's language rather than engineering jargon.

I view published materials with skepticism, even if found in engineering texts. I believe the most accurate data is found in actual experiments that test textbook "facts." I have found several errors in published material with regards to ESLs. The most famous example is the necessity of having a high impedance diaphragm so that the speaker operates in a "constant charge" mode. Originally published by Peter Walker in his famous Wireless World *articles in 1955, it is now taken as gospel.*

While I respect Mr. Walker's work, it is easy to demonstrate an ESL operates well with a low impedance diaphragm, even at low frequencies. Not only have my tests proved it, several readers have confirmed it. Furthermore, the Beveridge ESL (patent #3,668,335, June 6, 1972) uses an aluminized Mylar diaphragm with only a 4Ω/inch resistance, is sold commercially and can be heard by anyone who is still skeptical.

I have found other problems with low impedance diaphragms which preclude their use however. First, they are more massive than graphite coated ¼ mil Mylar. Second, they are capable of discharging their entire capacity across an arc; usually the speakers burst into flames if this occurs. Still, the point is diaphragm impedance is not a great concern, and published material on this is simply erroneous from a practical point of view. Fortunately this makes the job of the amateur much easier since the process of diaphragm coating is no longer a sacred art, but works just fine with a rather haphazard application of conducting material.

Mr. Wagner's letter contains errors that I am compelled to comment on. I said that insulating paint is unnecessary, inconsistent with my article "An Electrostatic Speaker System," (Part II, SB 3/80). This is another example of accepting previous publications as "fact." I read about insulating requirements, built my early ESLs with insulation and ad-

vised readers to do likewise. Since that time, tests demonstrate that uninsulated speakers work flawlessly, and I published a follow-up to the original article.

Mr. Wagner also comments that he uses larger field strengths in his speakers that makes insulation necessary. If he had read accurately he would have noted I have a range of 60-80 mils for the diaphragm-to-stator spacing with a bias voltage of 2-3kV. I use 60 mil spacing and 3kV for my speakers, therefore I could use 50V/mil, which is higher than his, and still find insulation to be unnecessary. Those who doubt me should feel free to insulate their speakers, but those readers who wish to build equally well performing speakers at reduced cost and complexity may omit insulation.

Mr. Wagner's claims for his plastic stators are contradictory. After all, if his stators were strong enough he would not have to bother with wood ribbing in the first place. Also, thick stators are highly undesirable, for efficiency, as I will point out. In any event, aluminum or steel stators are stronger than plastic.

I repeat that a diaphragm must be supported every six inches. There are many references in texts to a ratio of 100:1 between the unsupported diaphragm area and the diaphragm-to-stator spacing. While it is true that this ratio could be considerably higher if absolutely flat uniform speakers were built, for amateur builders, the 100:1 ratio remains an excellent compromise, with no detectable influence on speaker efficiency. Mr. Wagner's designs use a ratio of 200:1, even though he does not seem to recognize it.

He also mentions my speakers are limited to midrange and tweeter applications which is not true. I pointed out that they can be used full range and many readers/builders use them that way, as I have myself. Using stroboscopic equipment, I studied the motion of the diaphragms and found they do not move in an arc from insulator-to-insulator, but rather in a piston-like manner, except within about ¼" of the restraining structure. Therefore the bass performance is not significantly impaired by the insulators.

When stretching the diaphragm, Mr. Wagner again has not done his homework as he states I use both a stretcher and a heat gun. Had he read more carefully he would realize that I originally used a stretcher, but now only use a heat gun. This greatly simplifies building the speaker for the average amateur. I continue to use a stretcher for curved diaphragms as they must be stretched in only one direction and a heat gun shrinks both directions. I have found diaphragm tension is noncritical, as long as the speaker is used above fundamental resonance. Although it takes more force to displace a tight diaphragm, one can use higher polarizing voltage which supplies that force, and the net result is no change in efficiency, but a great improvement in the stability of the system using high diaphragm tensions.

I stand by my statement regarding the relatively poor SPLs of Mr. Wagner's design as this has since been confirmed by two individuals who have built his speakers and were disappointed with the SPLs. He states that large perforations do not lower output and states that in actual practice the size of the holes has nothing to do with the acoustic level, but in fact the only thing that matters is the open area (in this case 50%).

To demonstrate how absurdly inaccurate that statement really is, let's take his statement to its logical extreme: Take his 2' ESL stator and make the slots larger, much larger...but always keep the 50% open area he specifies. You would end up at the extreme with a stator with a single slot in it one foot wide. Electrostatic forces only operate over very short distances and there would be essentially no force applied to the 1 foot wide area of diaphragm under the slot. The remaining stator would be two 6" wide stripes on either side of the slot and as they would be solid, no air, and hence no sound would pass through them. The only sound produced by this speaker would be at the edges of the solid stator where the diaphragm sees both stator and free air. Needless to say, it doesn't take a rocket scientist to realize that the speaker would have abysmal SPLs and be extremely inefficient.

My experiments have demonstrated that smaller perforations are better if all else is equal. However, all else is not equal, particularly with respect to the thickness of the stator. When the size of the perforations approaches the stator's thickness, the air sees the holes as "tunnels" and this forms the basis of a Helmholtz resonator with distressing effects on the sound quality. Therefore a thin stator with tiny perforations is necessary for maximum efficiency. Diaphragm-to-stator spacing powerfully reduces efficiency. Electrostatic forces travel only very short distances and the force decreases by the square of the distance. Doubling the spacing from 60-120 mils requires that we square the drive voltage to compensate.

In reality this is extremely difficult to do. In theory all one needs is a transformer with a very high turns ratio. Transformer design is a real quagmire that I will not get into here other than to say that increasing the turns ratio adversely affects virtually every other aspect of transformer design. The ideal ESL transformer has yet to be built. Even the major ESL manufacturers have not been able to solve this one. For example, Dayton-Wright used enormous turns ratios because their speakers had 250 mil spacing. But the transformer couldn't operate above 6kHz, so D-W had to add a dynamic tweeter. Acoustat started with a direct-coupled amp as did Beveridge to avoid the transformer problem. Acoustat now has a dual transformer to try to deal with the problem.

Note that increasing the polarizing voltage does not compensate for the decreased effi-

ciency of large spacing. Larger spacing does allow higher voltages, but the spacing reduces its effectiveness so the net result is no change.

I find it interesting that Mr. Wagner confirms reports that I have heard regarding failures of his Ivory soap diaphragm coating. Apparently he has found a cure for this, although it seriously limits what can be used for the diaphragm contact, which is not a problem with graphite.

I believe Mr. Wagner fails to note the fundamental difference in our design philosophies. He is attempting to build the ideal electrostatic speaker system, and is working within the confines of published engineering constraints. I am trying to build the best performing speaker system using whatever technology is required to do so. At the same time it must also be practical for amateur construction. Within the constraints he has placed upon himself, Mr. Wagner has developed a nice speaker. Fundamentally it breaks no new ground as it is essentially a refined version of the original QUAD or KLH-9. All these designs are basically the same as they use segmented diaphragms, multiple crossovers, dipole radiation, and electrostatic bass. Such designs cannot produce truly deep bass nor Row A concert hall sound pressure levels.

Mr. Wagner will surely disagree and say his are different, but this is only true in subtle ways as the basic design compromises are similar. His main contribution is that it is possible for an amateur to build them. He could have made them vastly easier to build. I stand by my statement that in their present form they are the most difficult, complicated, expensive, and impractical way to build ESLs that I have ever seen. This does not mean that they don't work—I'm sure they do, but the design compromises he chose are, in my opinion, not the best possible.

I had to take a different approach. I record live symphony orchestras and pipe organs. Such an uncompressed master tape requires prodigious amounts of bass and midrange energy for Row A concert hall realism. Totally electrostatic designs that are capable of such performance are not adaptable to the typical home or amateur construction. Fortunately there are other ways to do the job without compromising electrostatic details. I'm flexible enough to experiment, as well as throw away the textbooks when it can be proved experimentally that the problems they define are nonexistent or insignificant. The result can be exceptionally good performance with relative ease of construction.

I think it regrettable that Mr. Wagner's attitude is so defensive. It is obvious that no matter what I might say, he will dispute it which precludes a constructive exchange of ideas. At the same time I think he has much to offer if only he would take his head out of the engineering texts and incorporate some practical experimentation. I once heard that "An ounce of application is worth a pound of abstraction."

KEELE FORMULAS

In Mr. Bullock's comments about reflex design formulas in my books ("Mailbox," *SB* 2/88, p. 47), he attributes the formulas to Hoge. I received the formulas privately from D.B. Keele, Jr.

In fact, Hoge, in his "Confessions of a Loudspeaker Engineer" (*Audio*, Aug. 1978), shows the same design flow chart I used in several books. It appears in Figure 2 of his article and Hoge's caption reads, "Keele's method of approximate vented-box system design."

In that article Hoge also describes the method used in the flow chart, that "it was originally developed by Keele."

David Weems
Newtonia, MO 64853

PUSH-PULL WOOFERS

I am presently building a subwoofer system. I would like to use two drivers per cabinet. Is there any simple formula or way to design a push-pull subwoofer system using two identical drivers?

Also, I am aware the size of the driver determines up to what frequency it will disperse widely without beaming. For example, a midrange with a four-inch radiating area would disperse up to the wavelength of its diameter, which is 16,440/4, or 4110Hz. When I design the crossover, is it better to have wide dispersion using lower crossover points, or less dispersion using higher crossover points?

Franklin Kang
Austin, TX 78729

UK ENGINEERS

I received the *SB* 4/87 issue and its contents so gripped me, I stopped all my other work and spent a long time reading all the fascinating material.

So many of us concerned with audio and sound reproduction over the years have made contacts and friends in our "sound world." One of my early articles, recently unearthed for a lecture, dates back to 1934: "Gramaphone Recording, Record Manufacturer, and Record Defects."

Researching this material, I listed all the British engineers concerned with loudspeaker design and manufacture. Some are dead, of course, but the names may interest our American friends: Hugh Brittain, G.A. Briggs, Rex Baldock, Raymond Cooke, Martin Colloms, Donald Chave, M.L. Gayford, H.A. Hartley, H.D. Harwood, Stanley Kelly, Harold Leak, Laurie Fincham, Norman McLachlan, James Moir, Norman Mordaunt, Capt. H.J. Round, J. Rogers, Arnold Sugden, D.E.L. Shorter, A.F. Sykes, P.K. Turner, P.G.A.H. Voigt, Cecil Watts and B.J. Webb.

Donald Aldous
Consultant Technical Editor
High-Fi News and Record Review
Plymouth, England

STEP RESPONSE CONCLUSIONS

Apparently, my reply to Mr. Brown's letter in *SB* 1/88 didn't manage to communicate my reasons for disagreeing with his interpretations of driver step response measurements. Perhaps a few illustrations will help.

Figure 1 shows two step responses: (a) an ideal driver with a third-order high pass filter at 2kHz and a second-order low-pass at 20kHz; (b) an ideal driver with a second-order high-pass filter at 200Hz and a third-order low-pass at 2kHz. There is no time delay in the simulation. This figure models the step responses of a tweeter and a midrange driver, with their diaphragms physically aligned and employing an ideal third-order crossover.

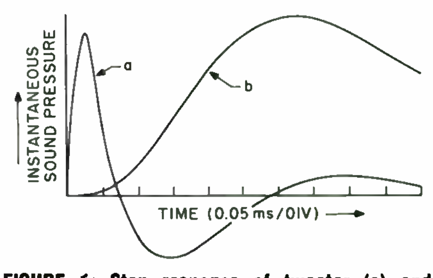

FIGURE 1: Step response of tweeter (a) and midrange (b) models with ideal third-order crossover.

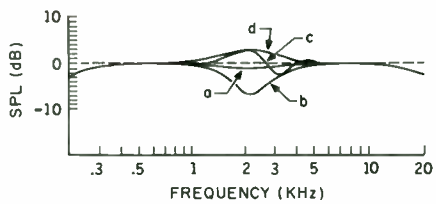

FIGURE 2: Net frequency response with interdriver time delay: (a) 0mSec, (b) −0.05mSec, (c) −0.375mSec, (d) 0.125mSec.

Figure 2 shows the frequency response of the sum of the outputs of the same drivers, with several values of interdriver depth displacement. Curve (a) corresponds directly to *Fig. 1*, with the tweeter and midrange physically aligned. This curve would be flat except for the effect of mild phase shifts from the midrange's low-frequency limit and the tweeter's high-frequency limit. Curve (b) is the net response when the tweeter is delayed by 0.05mSec with respect to the midrange. As seen in *Fig. 1*, such a delay will cause the initial rise of the midrange and tweeter to approximately coincide, as you suggested in your letter. Curves (c) and (d) show the net response when the tweeter has been set behind or in front of the midrange by the distance required to bring the drivers in phase at the crossover frequency (−0.375mSec and +0.125mSec, respectively). The hump appears because ideal odd-order high pass and low pass filters *should* be 90° out of phase with each other at crossover. (Of course, these results apply only to this particular driver/crossover combination.)

As can be seen, when you try to alter the step response by inserting more time delay than is needed to physically align the drivers, you may be adversely affecting the frequency response. As I mentioned in my previous reply, and as is illustrated in my article, "An Introduction to Frequency Response and LMP" (*SB* 1/87), the phase response of a time delay does not equal the phase response produced by driver rolloffs and crossover filters. The discrepancies between driver phase response and the phase shift introduced by the time delay cause the net response to go through a series of reinforcements and cancellations. This can lead to the "comb filter" effect in *Fig. 2c*. Such irregularities make it difficult to fairly assess the effects of interdriver time delays on factors like imaging or overall "sound reproduction."

Your approach is to delay the tweeter so the initial rise in its step response coincides with that of the midrange or woofer. Since the phase and amplitude response of drivers and crossovers do not form a simple time delay, the step response measurements are difficult to interpret. In particular, you are incorrect to conclude that adding a crossover filter to your midrange will effectively introduce more time delay. As seen in *Fig. 1*, the initial rise of the step response of a low-pass filter does not clearly indicate when the actual response starts, even in the absence of any time delays, and particularly in the presence of measurement noise.

Max Knittel makes a somewhat different interpretation of step response measurements in his article, "Step Response of Loudspeakers" (*SB* 4/86). Here, he delays the tweeter with respect to the woofer to try to make the overall step response show a single peak. This is somewhat similar to trying to make the peak in the tweeter's step response coincide with the peak in the woofer's step response. As I mentioned in my previous letter, whether the new step response is truly an "improvement" is unclear. A good step response is related to a good phase response, and no amount of time delay can make a non-minimum phase speaker (such as you and Mr. Knittel discuss) become minimum phase.

As an alternative, Max Knittel mentioned adding interdriver time delay so the tweeter's phase shift at the crossover frequency equals that of the woofer. As shown in *Fig. 2c*, this may have adverse effects in the case where a large time delay is required; also, symmetrical lobing is only realized at the crossover frequency, and not immediately above and below this frequency. I'm not sure of Mr. Knittel's claim that the two alternative methods usually "agree very closely" with each other (*SB* 4/86, p. 12). For example, advancing the tweeter in *Fig. 1* by +0.125mSec will bring the drivers in phase, but will actually move the peaks in the step response farther apart.

In reference to your comments about LMP, judge its performance by obtaining frequency-response measurements of your completed system and compare them to LMP's predictions. If you see discrepancies, it is likely the LMP models you entered for your drivers need revision. Due to unpredictable driver phase shifts, I believe a trial-and-error approach using frequency response measurements and/or modelling (as with LMP) to judge the effects of interdriver time delay and crossover filters is most effective.

Ralph Gonzalez
Philadelphia, PA 19143

FORMULA REVISIONS

As of last year, the revised speed of sound is 331.29m/Sec (1086.9624 ft/Sec). This can have serious implications for speaker builders.

For example, Ralph Gonzalez used the formula (*SB* 2/87, p. 42):

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

It should now correctly be

$$f = 13{,}043.548/w$$

Also Duke LeJeune's formula (*SB* 4/86, p. 16), 1,130 ft/Sec should be revised to 1086.9624 ft/Sec.

Also, in *SB* 3/86, page 29, Peter Crosby's quarter wavelengths should also be revised. For example, 20Hz, quarter wavelength would be 163.04″ instead of 169.05″.

Finally, LeJeune's formula for air density, 1.29kg/m^3, varies with altitude and should be compensated accordingly. Transmission lines would sound different in New Orleans and in Denver, although I don't know to what degree. In Martin Colloms' *High Performance Loudspeakers*, third edition, the same formula is 1.2kg/m^3. Which is correct?

Dennis Wood
Raleigh, NC 27606

Ralph Gonzalez replies:

According to my old physics book, the speed of sound in air at Standard Conditions of Temperature and Pressure (STP) is about 13,044 in/sec (rounded to the whole number). The revision you mention is most likely insignificant to most audio applications. STP is referenced to a temperature of 0 degrees Celsius and a pressure of 1 atmosphere (atm). The figure I gave of 13,500 in/Sec is the speed of sound in air at 20° Celsius, to within 1%. Similarly, Duke LeJeune's and Peter Crosby's figures are correct for 20°C.

The speed of sound waves in a gas is proportional to the square root of p/d, where p is the ambient pressure and d is the density of the gas. If the atmospheric pressure decreases, the density will decrease proportionately, so the speed of sound remains unaffected.

However, it would be interesting to investigate the effect of changes in air density on the tuning of sealed or vented speaker enclosures. At 0°C and 1 atm, the density of dry air is 1.293 kg/m^3. I believe that Martin Colloms' figure of 1.20 kg/m^3 applies to the density of air at 20°C and 1 atm.

The density of air is proportional to the atmospheric pressure, which varies with elevation. At elevations (e feet) below about 6000 feet, a good approximation for the atmospheric pressure is:

$$p = 1\text{ atm} - 3.45 \times 10^{-5}\text{ atm/ft} \times e$$

This will vary somewhat with the current barometer reading. You can calculate the density of air (at 20°C) from the atmospheric pressure via:

$$d = \frac{1.2\text{ kg/m}^3 \times p}{1\text{ atm}}$$

Now, I believe the compliance of a given volume of air to a given woofer is inversely proportional to the density of air. Speakers which are most sensitive to a change in air density are those for which the acoustic suspension dominates the woofer's mechanical one (V_{as} V_b). The resonant frequency and Q of such a woofer in a closed box of a given volume, are inversely proportional to the square root of the air's compliance. In this case, a change in elevation from 0 to 6,000 feet will simultaneously reduce the resonant frequency and Q of the closed box speaker by about 10%. The effect on vented speakers is a little more complicated, but I don't think it is more serious.

I think you can correct the effects of high elevations on enclosure tuning by reducing the enclosure volume proportionately to the reduction in air density.

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ELECTRONIC DELAY

continued from page 21

CONCLUSION. An electronic delay line provides the most practical means of delay aligning active loudspeaker systems. A tapped line using a modified Pade delay approximation with the Steffen topology provides a convenient implementation.

Delay aligning a system is a tedious process when you don't know the acoustic centers of the drivers. Obtain this information from the manufacturer when possible. Be sure to consider the actual listening position when delay aligning—don't automatically align with respect to the baffle.

Most importantly, delay alignment is audible and provides improved imaging.

If readers express enough interest Old Colony will offer the electronic delay circuit as a kit, or a printed board only. If you are interested enter #GC72 on the Fast Reply card.

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INTERACTIVE AUDIO DESIGN SOFTWARE for PC and Macintosh. **SpeakerCAD** draws predicted performance graphs, calculates Thiele/Small parameters, vents. **Filters** calculates first through sixth-order, Bessel or Butterworth electronic crossovers, quotes nearest standard values. $29 for both. ***ATRIUM ELECTRONICS***, 2302 5th St., NE, Salem, OR 97303, (503) 363-5143. T4/88

CARDED WOOL: 100 percent clean and carded wool from the long-wooled sheep for the speaker builder. $9 per lb. plus shipping. Samples, $2 plus SASE. Send to ***BABYLON FARM***, 95 Day St., Granby, CT 06035. T4/88

PYLE DRIVERS AND PIONEER OEM SERIES subwoofers, woofers, mids, tweeters, passives, crossovers, grilles for home, car or guitar. Over 200 models stocked for prompt shipment. Always discounted. VISA/MC/DISC. Dealers inquiries? ***SPEAKER SHOP,*** 318 S. Wahsatch, Colorado Springs, CO 80903, (719) 475-2545. T4/88

ESL BUILDERS—Hard to find ½ mil polyester film 20″ wide, 50¢ per foot. Triad S142A transformers available, $50 each plus UPS. ***SOUND SUPPLIES,*** 419 Appleton, Vernon Hills, IL 60061. T3/88

HARMAN KARDON DEALER QUITTING. Final clearance-factory sealed stock: 690i receiver, $420; 590i receiver, $299; 870 amplifier, $325; 91 tape deck, $150. Display model stock: 590i receiver, $289; 395i receiver, $199; 91 tape deck, $140; 825 preamp, $255; 870 amplifier, $315. Also available: (1) Shure V-15 V-MR, sealed, $75. Prices are firm—all are champagne finish, full factory warranty. MasterCard/Visa. Freight extra. Larry, 625 Claremont, Ashland, OH 44805, (419) 289-6481. T3/88

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

"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

MAGNAVOX CD PLAYERS CDB-650, $289; CDB-470, $159; CDB-471, $169; CDB-472, with remote, $209; CDB-473 remote/volume, $259. Trade-ins are welcome. ***DIGITAL SOUND DISTRIBUTORS,*** Rt. 2, Box 183, Walton, IN 46994, (219) 626-2756. T3/88

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TRADE CLASSIFIED ADVERTISING RATES: 55 cents per word *including name, address and zip code—prepaid only. 10% discount for six insertions.* Speaker Builder *cannot accept responsibility for the claims of either the buyer or the seller.*

FOR SALE

JBL 18″ subwoofer, 2245H driver, works great, includes computer tuned cabinet, 120W mono amp, crossover, $475. Scott Lewis, 3546 Robson St., Indianapolis, IN 46201, (317) 352-0749.

Stax DA-80M 90W Class A DC Monoblocks, $1,000/pair; PS Audio Model 2 amplifier, $200; Luxman TA-1 tonearm, $75; Magnepan Unitrac tonearm, $75; Ivie IE-10A real-time analyzer, $700; Ivie IE-20B noise generator, $100; Kenwood KT-52B digital tuner, $150; Supex SDT-77 MC transformer, $25; FRT3G MC transformer, $25; Dual 1218 changer, $25; KLH Model Twenty-Four tuner/changer, $25; Sony TC-355 tape deck, $50. John, (619) 284-0181.

Four JBL 8″ woofers model 117-H1, never used, still in original carton, cast frame, PVC surround, white ribbed Aquaplas cone, used by JBL in their L46 system. Very beautiful and rugged woofers. Wish to keep in pairs. $50 each. Jeff Bagby, (317) 455-2406.

Richter Scale Series III, $250; Infinitesimals, $200; EPI LS841 auto speakers, $185; SEAS CA25RE4X/DC, $45/pair; Gold Ribbon Concepts, ribbons, 2 pair sealed; pair Heath A9 mono integrated amps. R. Omegna, 44 Cold Hill Rd., Morristown, NJ 07960, (201) 543-2971 evenings until 10 p.m.

Naim NAP110 power amp, $250; NAC42S preamp, $150; Snaps power supply, $125; all mint, $500 for all. Falcon Acoustics designed speakers with two KEF B110B 5¼″ and one 1½″ KEF T52B dome tweeter in each 1.5 cu. ft. vented teak enclosure, excellent condition, $275/pair; Lang power amps, mono Class A 20W kits, stuffed boards, all parts, never used, $270/pair; Denon 207 MC cartridge, 2 hours, $100; Becker 10″ 910A100 woofers, never used, $60/pair. Dan Hildreth, 1514 E. Stephens, Tempe, AZ 85283, (602) 838-2789.

Dynaudio 21W54, non-mps, $100/pair; Rogers LS3/5A slightly modified, $350/pair; Dyna ST-70 with polypropylene capacitors, $90; Pair Dyna MK IIIs with KT-88s and some mods, $100 each; Pair Heath W-7Ms, $75 each. Ron Howanec, PO Box 3473, Virginia Beach, VA 23454, (804) 491-8638.

Ariston RD-40 AC turntable, very low hours, high definition sound, fine condition, b/m, $99 without arm, with Grace 707 arm, Grado pickup, $25 additional. Send SASE for good review from *Sensible Sound*. Eric Anderson, 308 NW Linden, Ankeny, IA 50021, (515) 964-7710.

Unused pair Strathearn ribbons with transformers. Suitable for Spangler/McKenzie mod. *SB* 3/85, $300. Dean Price, 1616 Millbrook S.E., Grand Rapids, MI 49508, (616) 241-5324 evenings.

Ohm Walsh 4 speakers, 1 year old, fine condition, list $1900, $1,200 or best offer. Jim Hess, 2013 10th St. Ct., Coralville, IA 52241, (319) 337-9817.

WANTED

Acoustat monitors X, 3 or 4 or Koss 1A and information on modifying the Acoustat ES servo amp, schematics, etc. Also electrostatic speakers for parts, panels, transformers, etc. Tom Bourdeau, 10870 East Court Dr., Windsor, Ontario, Canada N8R 1E8, (519) 735-9023.

Pair of fiberglass Focal EGGS for 7" drivers. R. van Wynen, 322 Oakdale Ave., Corte Madera, CA 94925, (415) 924-2950.

Seek classic mono (or stereo) system—MacIntosh tube amps, Klipschorn speaker(s). Steve Hauser, 2235 MacBride Dr., Iowa City, IA 52240, (319) 335-1890 days.

Sanyo Plus T55 AM/FM stereo tuner. Stanley H. Grycz, 2935 Crehore St., Lorain, OH 44052, (216) 288-9480.

Pair replacement diaphragms for Altec drivers model 808-8B. Andrew R. Lewis, 3929 S. Sherman, Englewood, CO 80110, (303) 781-5573.

CLUBS

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

CONNECTICUT AUDIO SOCIETY is an active and growing club with activities covering many facets of audio—including construction, subjective testing, and tours of local manufacturers. New members are always welcome. For a copy of our current newsletter and an invitation to our next meeting, write to Richard Thompson, 129 Newgate Rd., E. Granby, CT 06026, (203) 653-7873.

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.

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.

AUDIOPHILES AND EQUIPMENT BUILDERS in the Long Island area—well established club looking to expand and share out experience. Monthly meetings, record critiques, technical help. Island Audio Club, 589-4260 (Suffolk), 271-4408 (Suffolk and Nassau), 825-2102 (Nassau).

THOSE INTERESTED IN AUDIO and speaker building in the Knoxville-East Tennessee area please contact Bob Wright, 7344 Toxaway Dr., Knoxville, TN 37909-2452, 691-1668 after 6 p.m.

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

ESL DIY'ERS: A new electrostatic loudspeaker do-it-yourselfers group is now forming. Our purpose is to share valuable theory, how-to, and parts source information for building our own state-of-the-art electrostatic loudspeakers. For further information, please write (SASE please) to: Neil Shattles, 829 Glasgow Dr., Lilburn, GA 30247.

SAN DIEGO AUDIO SOCIETY forming for hi-fi tinkerers and do-it-yourselfers. If you enjoy collecting, building, rebuilding and repairing classic audio equipment, especially tube-type, call Mike Zuccaro (619) 271-8294 (evenings & weekends). Old timers and engineers welcome.

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 721464, Berkley, MI 48072-0464.

About Your Personal Classifieds . . .

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

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

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

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

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 211, Boston, MA 02126.

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.

NEW JERSEY AUDIO SOCIETY meets monthly. Emphasis is on construction and modification of electronics and speakers. Dues includes monthly newsletter with high-end news, construction articles, analysis of commercial circuits, etc. Meetings are devoted to listening to records and CDs, comparing and A-Bing equipment. New members welcome. Contact Bill Donnally, (201) 334-9412 or Bob Young, 116 Cleveland Ave., Colonia, NJ 07067, (201) 381-6269.

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

WASHINGTON AREA AUDIO SOCIETY (N. VA, MD and DC) is looking for sincere audiophiles who are eager to devote their time and get involved with the direction of the society and the publication of a monthly newsletter. Please contact: Horace J. Vignale, 13514 Bentley Circle, Lake Ridge, VA 22192-4316.

THE WESTERN NEW YORK Audio Society (WNY Audio Society) is an active and growing audio club located in the Buffalo area. We issue a quarterly newsletter and hold meetings the first Tuesday of every month. Our meetings have attracted many local and distant manufacturers of audio related equipment. We are involved ein all facets of audio—from building to purchasing at discount prices. For a copy of our current newsletter and information regarding our society, please write to M.A. Monaco, WNY Audio Society, PO Box 312, N. Tonawanda, NY 14120.

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

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

THE INLAND AUDIO SOCIETY IN THE SAN BERNADINO-RIVERSIDE AREAS, now in its third year of existance, is inviting audiophiles and music lovers in the San Bernardino, Riverside, Orange and Los Angeles counties to join us at our bi-monthly meetings and through our quarterly publication, in the pursuit for that elusive sonic truth. We provide a forum for auditioning equipment, sampling live music for educational purposes, guest presentations, discussing recordings, and the sharing of ideas, tips, theories, opinions, experience, and new product news relating to audio systems. Additionally we cater to the hobbyist who designs, builds and/or modifies electronic components and tranducing gear. Write for information concerning membership, dues and subscription. IEAS, PO Box 77, Bryn Mawr, CA 92318, (714) 793-9209.

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

SIXTIES MUSIC FANS. The first club for fans of Sixties Era music wants to meet you. Strong audio and quality recording emphasis. How-to information shared. Worldwide membership. Informative, entertaining, and provocative newsletter published bimonthly. Help the world remember when rock had real artistic merit. Free brochure: send SASE or IRC to Classic Rockers Music Club, PO Box 1043C, Stevens Point, WI 54481.

TUBE AUDIO ENTHUSIASTS. Northern California club meets every other month. For next meeting announcement send a self-addressed, stamped no. 10 envelope to Tim Eding, 2113 Charger Dr., San Jose, CA 95131.

THE VANCOUVER AUDIO SOCIETY publishes a bimonthly newsletter with technical information, humor and items of interest to those who share our disease. We have 40 members and meet monthly. Six newsletters per year. Call (604) 251-7044 or write Dan Fraser, VAS, Box 4265, Vancouver, BC, Canada V6B 3Z7. We would like to be on your mailing list.

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

Advertising Index

morel

high fidelity range

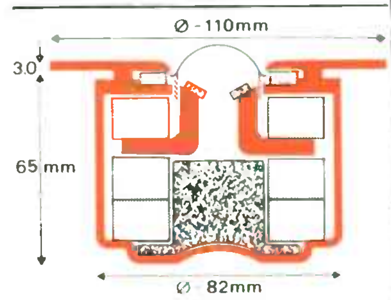
The MDT 33 is an extremely fast Tweeter, using a 28mm (1⅛") diameter voice coil and a chemically treated soft dome, and is ideally suited for two way systems with the possibility of a lower than normal crossover frequency, as well as for three and multiple way systems.

Incorporating the Morel Hexatech voice coil technique, aluminium wire wound on an aluminium former and using flexible wire termination ensures excellent high frequency performance with exceedingly high power availability. The power handling is further enhanced by using Ferrofluid in the magnetic circuit.

The magnetic system itself is an ingenious Morel double magnet design and is completely enclosed. By venting into the enlarged area of the double magnet system, a low resonant frequency of 500Hz is obtained with a remarkably smooth roll off from 1000Hz through this damped resonance area. The subsequent wide range response of 1400-20000 ± 0.6dB is obtained with a harmonic distortion of below 0.8% over the entire range. The distortion figures quoted are with an input power giving an output level of 96dB at 1 metre. The MDT 33 sensitivity is 92.5dB for 1 watt 1 metre, and a power handling capability of from 100 to 500 watt subject to crossover frequency.

With such a dome tweeter design, the acoustic qualities at lower than normal crossover frequencies are excellent with an absence of honking, and even at the more normal crossover frequencies this excellent acoustical behaviour is evident to the ear. With the lower crossover frequency available and high capability, it is ideal for consideration in two way systems using a 10" or 12" woofer.

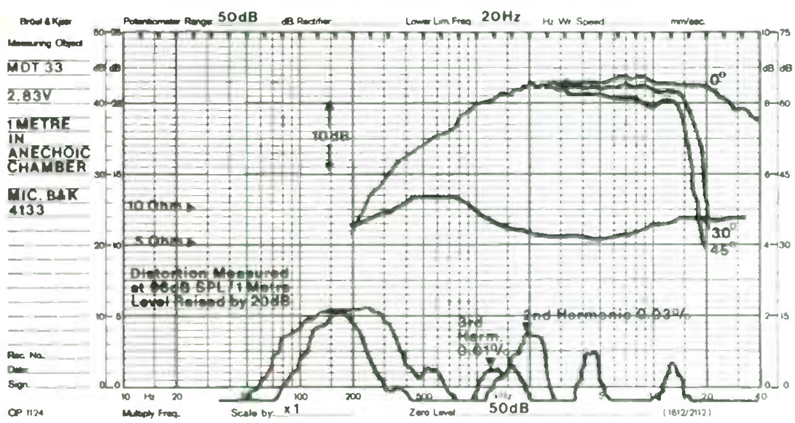
To utilise the dome at the lower than normal crossover frequency available makes it necessary to have a sharp roll off below 1400Hz of minimum 12dB per octave to protect the tweeter from mechanical damage. This makes it ideal for use with active systems.



Specification

Overall Dimensions	Ø - 110mm × 68mm	Vas	0.016 litre
Face Plate Thickness	3mm	Moving Mass including Air Load	0.44 grams
Voice Coil Diameter	28mm (1⅛")	Effective Dome Area	8.5 cm²
	Hexatech Aluminium	Dome Material	Treated Fabric
Voice Coil Former	Aluminium	Frequency Response	1400-20000 ± 0.6dB
Number of Layers	2		(1000-40000 ± 5dB)
DC Resistance	5.2 ohms	Resonant Frequency	500 Hz
Nominal Impedance	8 ohms	Power Handling Din:	
Voice Coil Inductance @ 1 Khz	0.09mh	X Over 1400 Hz	100W
Air Gap Width	0.75mm	X Over 5000 Hz	500W
Air Gap Height	2.5mm	Transient Power 10ms	1500W
Voice Coil Height	2.7mm	Sensitivity	92.5dB (1W/1M)
Flux Density	1.95T	Rise Time	10μs
Force Factor (BXL)	4.76 WB/M	Intermodulation Distortion	
Rmec	2.09ns/m	for 96dB SPL	<0.2%
Qms	0.66	Harmonic Distortion	
Qes	0.38	for 96dB SPL	<0.8%
QT	0.24	Nett Weight	1.2 kg

Specifications given are as after 24 hours of running



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