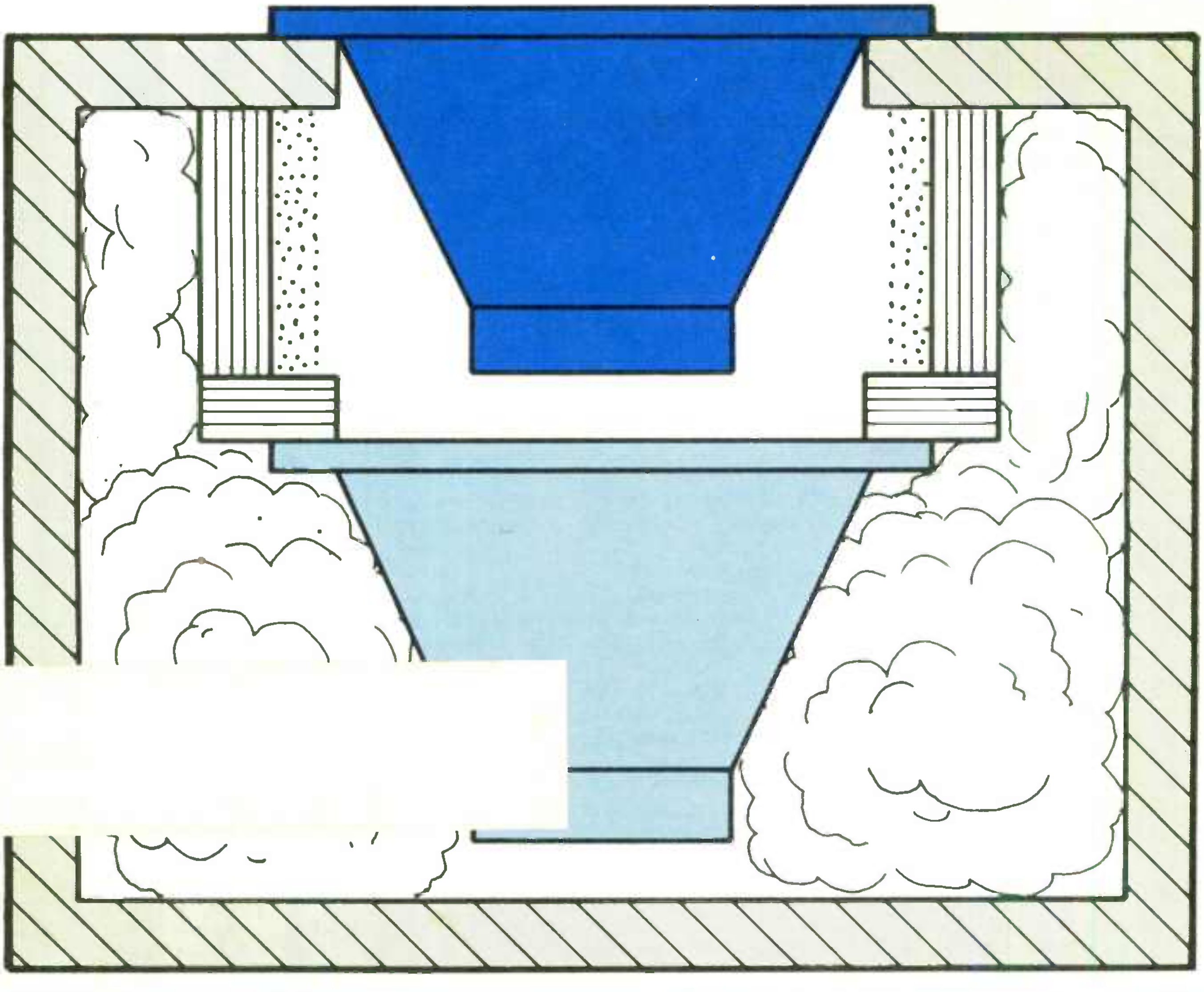
8 9

THREE/85

U.S. $4.00
CANADA $5.00

SPEAKER BUILDER

CONSTANT PRESSURE BOX



Good News

SCIENTIFIC DESIGN SOFTWARE has released Version 2.0 of its Computer-Aided Speaker Design software for the Apple II computer. This package allows complete modeling of loudspeaker systems and accurately predicts system response.

The program calculates and plots the response and displacement-limited functions for sealed and vented systems. You can also assess the effect of an electronic filter, and you have full control over graph scales and resolution. The program allows you to design 6, 12, 18 and 24dB/octave crossovers complete with impedance correction circuits. Nine utility programs calculate various often-used functions including reference efficiency from Thiele/Small parameters, vent tunings and passive radiator mass.

Version 2.0 comes as a two-disk set, which includes the main program disk and a file disk that allows data storage on up to 800 speaker drivers. The file disk is shipped loaded with data on more than 120 drivers. Computer requirements include an Apple II with 64K bytes of memory, Applesoft ROM (read-only memory) and two disk drives. The Grappler parallel printer interface is supported for screen printouts.

For further information, contact Scientific Design Software, PO Box 3421, Northridge, CA 91323.

Fast Reply #HH777

The new **ELECTRO-VOICE** TL3512 subwoofer provides high-output reproduction of very low frequencies. Based on the vented-enclosure modeling techniques of Thiele/Small, the TL3512 combines high efficiency and sensitivity (99dB, 1W at 1 meter) with low distortion and extended low-frequency performance in a 9-cubic-foot enclosure. By covering one port and applying appropriate boost-and-cut equalization, you can "step-down" the low-frequency limit from the normal 38Hz (3dB down) to 28Hz, a useful extension for many applications.

The TL3512 employs a DL18W very low frequency reproducer. The woofer's high-linear cone-excursion ability and 400W long-term average power capacity contribute to the TL3512's high acoustic output ability. The unit is constructed of black, vinyl-clad $\frac{3}{4}$-inch particle board and equipped with a sturdy metal grille screen.

Additional information is available from Electro-Voice, 600 Cecil St., Buchanan, MI 49107.

Fast Reply #HH453

The new Medallion transformer from **ACOUSTAT** delivers many sonic improvements in all Acoustat Magne-Kinetic (MK) electrostatic speakers. These improvements include increased dynamic range without strain; a warmer, smoother midrange; extended low-frequency response with better control and tightness of bass transients; extended high-frequency response with improved transient reproduction; and enhanced imaging and depth retrieval.

The Medallion transformer will be used in all Acoustat speakers made under the new ownership of the David Hafler Company. Acoustat will also offer a modification program for owners of existing Acoustat speakers without Medallion transformers. The modifications will be done at the Acoustat plant in Fort Lauderdale, Florida, and will include full testing, plus check-out and replacement of any marginal components.

Information about the modification program or the Medallion transformer is available from Acoustat, 3101 SW First Terrace, Fort Lauderdale, FL 33315.

Fast Reply #HH1037

REVOX of Switzerland has introduced a moderately priced three-piece mini-subwoofer system, the Piccolo-Bass, which consists of two Piccolo satellite speakers plus a subwoofer.

The subwoofer is engineered to radiate extremely low bass frequencies (48Hz to 120Hz, $\pm$3dB) and designed to blend with any decor. Measuring just $15\frac{3}{8}$ by $14\frac{3}{16}$ by $14\frac{5}{8}$ inches, the subwoofer can be placed virtually anywhere in a room without affecting bass reproduction. Complementing the subwoofer is a pair of Piccolo satellite speakers. Each mini-speaker, which measures just $5\frac{1}{2}$ by $8\frac{11}{16}$ by $5\frac{3}{4}$ inches, incorporates a $4\frac{3}{16}$-inch-diameter midrange driver and $\frac{3}{4}$-inch dome speakers. Frequency response is 80Hz to 22kHz, $\pm$3dB. The satellite speakers and the subwoofer feature an unobtrusive flat black finish, with a perforated metal grille.

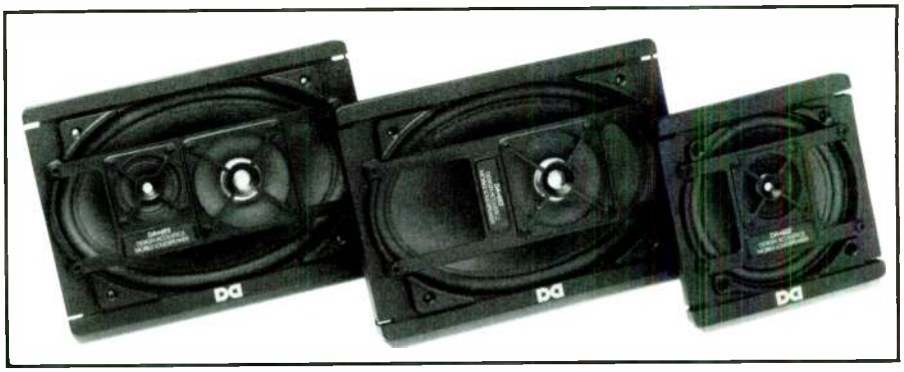
The Piccolo-Bass system is designed for both audio and video applications. In addition to augmenting any stereo system, the new ReVox subwoofer system may be used with stereo television receivers or as the rear speakers for the extra

surround-sound channel on movie videotapes and videodisks. The subwoofer system is also ideal for extending an audio system to other rooms.

For details, contact ReVox America, 1425 Elm Hill Pike, Nashville, TN 37210.

Fast Reply #HH393

Three new auto loudspeaker systems are now available from **DESIGN ACOUSTICS.** The speakers feature a mounting system that allows attachment on the front (or top) as well as behind the rear deck or door panel.



At the top of the line is the DA-693, a three-way system that features a 6-by-9-inch woofer, a 3-inch midrange and a 2-inch tweeter. Impedance is 4Ω and frequency response 40Hz to 20kHz. It can be used with amplifiers rated at up to 80W. The DA-692 is a two-way system with a 6-by-9-inch woofer and 3-inch midrange/tweeter. Frequency response is 45Hz to 20kHz, and it will handle up to 80W of amplification. The DA-602 is a two-way system with a 6-inch round woofer and a 2-inch tweeter. Power-handling capability is rated at 60W with a 4Ω impedance. Frequency response is 55Hz to 20kHz.

The speakers come with grilles, hookup wire and connectors, and mounting hardware and templates. For price and other details, contact Design Acoustics, a division of Audio-Technica, 1221 Commerce Drive, Stow, OH 44224.

Fast Reply #HH345

SPEAKER BUILDER
MAGAZINE

Edward T. Dell, Jr. *Editor/Publisher*

Contributing Editors

Robert M. Bullock, Bruce C. Edgar, G.R. Koonce, Nelson Pass

Barbara Jatkola *Managing Editor*

Karen Hebert *Office Manager*

Carol Symonowicz *Circulation Director*

Ruth S. Wilder *Design & Prod. Dir.*

Glen C. Birbeck *Technical Director*

Techart Associates *Drawings*

Advertising Representative

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

Editorial and Circulation Offices

Post Office Box 494
Peterborough, New Hampshire 03458

Speaker Builder is published quarterly by Edward T. Dell, Jr., PO Box 494, Peterborough, NH 03458.

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

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

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

A Note To Contributors

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

Speaker Builder Magazine (US ISSN 0199-7920) is published quarterly at $12 per year; $20 for two years, by Edward T. Dell, Jr. at 5 Old Jaffrey Rd., Peterborough, NH 03458 USA. Second class postage paid at Peterborough, NH. **POSTMASTER:** If undeliverable send PS form 3579 to **SPEAKER BUILDER** PO Box 494, Peterborough, NH 03458.

About This Issue

If you've ever looked longingly at a new speaker box and wanted to build one for yourself, you'll understand **John Cockroft's** fascination with Ivor Tiefenbrum's Isobarik system. Mr. Cockroft's knowledge of closed-box theory applied to a photo of the two-drive design yielded surprisingly good results and an exciting project (p. 7).

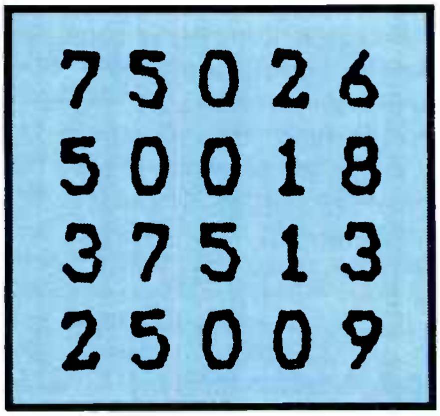
Bob Bullock's fearless forays into the wilds of complex passive crossovers has convinced him that some beasts in that territory just aren't worth taming. He has found some much more tractable electronic animals, which he finds do a better job (p. 14). Some of you may wish to use an update of the editor's simple crossover design (p. 20) to implement Mr. Bullock's configurations.

The legendary Strathearn ribbon driver (*SB* 4/84, p. 30) has generated a lot of excitement among speaker buffs—and a lot of frustration as well. **Don Spangler** and **Mark McKenzie** did a Fast Fourier Transform "Cat Scan" of the Strathearn and offer some radical remedial surgery to mend its shortcomings (p. 22).

Ambience is the Cinderella of two-channel stereo. Attempts to wed her to the prince—Dave Hafler's three-speaker Dynaquad four-channel system, the Bose direct/reflecting speaker and AR's new MGC system—are all admirable efforts to represent the reflected room sounds in a recording. **Dan McGillicuddy** takes a cue from Hafler and works out the complex variations starting on page 31.

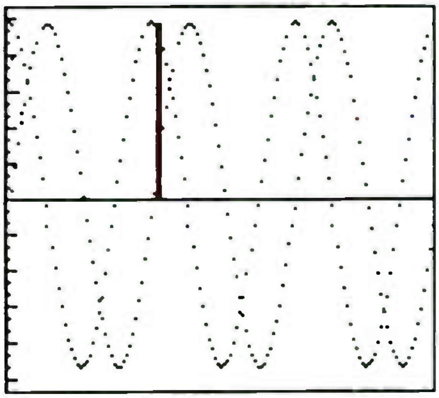
SPEAKER BUILDER

VOLUME 6 NUMBER 3 AUGUST 1985



15

37



24

FEATURES

DEPARTMENTS

Editorial

AUDIO'S IMAGE PROBLEM

Audio, I think, has a severe image problem, a kind of catch-22 situation that is a curious study in human shortsightedness. If any of you are old enough to remember the state of available musical fare after World War II, you know that we had fewer than a dozen symphony orchestras in the US. People who speculated about this relative scarcity usually said we Americans were just a rough, uncultured lot.

FM radio soon appeared, followed quickly by Goldmark's long-play disk, and in a relatively short span of a decade or so, we had a virtual explosion of musical appetites across America. Music boomed. Orchestras blossomed. And reproduced music created a hunger for live music.

Despite that evidence, the record makers, broadcasters and deans of engineering schools have generally regarded audio as a not very challenging technology that can easily be made "good enough" for the "general public." Most engineers view audio as a vocation for those candidates who cannot make it in the more demanding disciplines. Engineering professors regard audio in roughly the same way medical-school faculties regard podiatry.

Today a few signs of change are beginning to appear. Hollywood people who have been content to make 60 percent of their films with single-channel sound are beginning to perceive the attraction and the profitability of the sonic marvels of *Apocalypse Now, Raiders of the Lost Ark* and *Star Wars.* The vast video giant whose audio signal is compressed, narrow and distorted may finally be developing some semblance of a respectable audio signal that requires more than two tin cans and a string to transmit.

Like most scientific disciplines, today's computer-aided tools are not only giving us more information about more parameters affecting sound reproduction, they are also making it evident that we know a lot less about what is relevant to good sound than we had suspected.

The good news is that the tools for discovery are becoming accessible to us amateurs. This issue has two fine examples in Bob Bullock's third crossover article and Don Spangler's and Mark McKenzie's work on the Strathearn ribbon transducer. Their tools are more powerful and less costly than anyone could have anticipated. Dan McGillicuddy's and John Cockroft's articles demonstrate how much knowledge is available to experimenters these days.

The new knowledge and tools are only the beginning. The fun begins in the reality testing that we can now do to follow up on that knowledge. That is not drudgery or discipline. It is the fun of building a better device and executing a new trick or compromise that solves a heretofore baffling difficulty.

Audio is a magical ingredient. Most mortals never suspect their latent appetite for it. When we finally know enough about audio, and a tribe of young turks appears who are sharp enough to see the potential of realistically reproduced sound, then audio will come into its own as a respected profession and pursuit.

I regret to report the apparent failure of SRC Audio of Dallas, Texas. We have been unable to determine what has happened, but the company is evidently caught in the usual legal tangle of bankruptcy proceedings. A number of readers report checks cashed or bank card charges executed but no goods delivered. We are deeply distressed that advertising in our pages resulted in reader losses.

Readers should be aware that federal law requires mail-order vendors to deliver goods within 30 days or keep customers fully informed about delivery dates. Reputable firms will not execute a charge on your credit card until they are ready to ship the goods. If you do not receive an order within 30 days of the charge date, contact the vendor and ask for an explanation. If the explanation is not satisfactory, call your credit card company or the bank handling the card and ask for a refund.

If you order using a personal check and the delivery is delayed 30 days or more, you might try stopping payment on your check, but this is unlikely to be effective because in most cases, the vendor will have already deposited it.

Some of you raise the question of why we accept ads from businesses that go bankrupt. Need I say that we would avoid it if we could? Ads are placed far ahead of publication and mailing dates, and bankruptcies can occur overnight.—E.T.D.

AN ISOBARIK SYSTEM

A DARK HORSE BROUGHT TO LIGHT

BY JOHN COCKROFT

In recent years, audio journals have contained considerable information on almost every type of speaker imaginable—with the exception of the Isobarik design. This is unfortunate, because I am convinced that the Isobarik system has the potential to reproduce the most accurate, uncolored sound of any existing system. If the Isobarik principle were examined as exhaustively as the others, I am sure this potential could be realized.

The Isobarik system is capable of radiating clean, detailed, greatly extended bass on a par with transmission-line speakers, with all their fast transient response, in a box only slightly larger than that required for an acoustic suspension design using the same speaker unit. In the systems I constructed during my experiments, the cone motion was extremely well-controlled down to the lowest registers in the unit's free-air resonance area.

To the best of my knowledge, the Isobarik principle was invented by Ivor Tiefenbrun, owner and chief designer of Linn Products, a Scottish firm that manufactures audiophile products. He has world patents pending on his design, so you may construct Isobarik speakers for your personal use, but not for commercial purposes. *(Isobarik speakers are available in the US from Audiophile Systems, 6842 Hawthorn Park Drive, Indianapolis, Indiana 46220.—Ed.)*

ISOBARIK BACKGROUND. In the Isobarik system *(Fig. 1)*, one speaker creates the acoustic environment for the second speaker, which radiates the sound. The two speakers should be identical and are coupled electrically and acoustically at the opposite ends of a short tunnel mounted behind the front baffle. This places one speaker in front of the other. The back of the rear speaker opens into the box and is loaded just like an acoustic suspension speaker. It rolls off in the same predictable way according to the system Q_{TC} (i.e., f_3 equals f_c for a Q_{TC} of 0.707, f_3 equals $0.7861f_c$ for a Q_{TC} of 1, and so on). As the frequency goes lower and the sound drops to inaudibility, the cone continues to move in and out. Eventually, because of the box loading, the excursions begin to round off somewhat, but by then the frequency is very low.

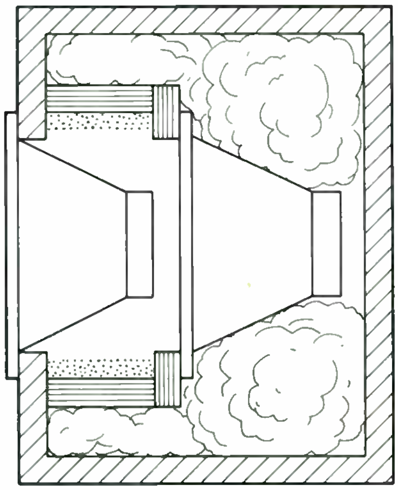


FIGURE 1: Cross-section of a typical Isobarik system enclosure showing the speaker location and the loading tunnel.

When I referred to the sound in the previous paragraph, I was speaking figuratively, as the rear speaker is buried in the box, and its radiation is only a small part of the total radiation (the less the better). The front speaker, whose back opens into the tunnel and the face of the rear speaker, does not see the box loading at all, except maybe indirectly at the lowest frequencies. Throughout most of the frequency range, the front speaker thinks that it is in an infinitely large box. When the front speaker moves back, the rear speaker moves back. When the front speaker moves forward, the rear speaker moves forward. The pressure between the two speakers remains constant. This is what the word "Isobarik" means—constant pressure.

Any difference in linearity between the two speakers affects both of them, and they tend to correct each other's actions, thus reducing distortion. Mr. Tiefenbrun says that his woofers' distortion approaches that of a well-designed midrange speaker. As I see it, the rear speaker acts as a dancing partner for the radiating speaker. When the front speaker goes forward or backward, the rear speaker follows the front's lead. But the rear speaker is so tactful that the front speaker hardly knows it is in a box at all. Because of this, the front speaker retains its free-air resonance (or close to it).

I have not seen the Tiefenbrun patent disclosures, and before I started to experiment, I knew very little about Isobarik systems. I had seen an advertisement for one of the speakers that showed a small cut-away photograph of the speaker's internal construction. I took it from there by applying standard closed-box design as proposed by Small[1] and others. I wanted something useful from the start, and I was somewhat conservative in my construction. Apparently that paid off, as I had no bad experiences during any of my experiments.

It seems that my speakers are acoustically larger than the Linn units (i.e., the Linn speakers have a higher

Q_{TC}). My Q_{TC} values ranged from 0.644 to about 1.13. Based on what information I could find (and which might be entirely misleading), I think the Q_{TC} of the Linn SARA speaker might be between 1.5 and 1.7. I would appreciate any further information in this area. Of the units constructed, only the one with a Q_{TC} of about 0.644 seemed slightly overdamped. This was quite a small speaker, and I was using it on a base about 2 feet from the floor. When I placed it on the floor, right up against the back wall, even it behaved extremely well.

PROJECT ONE. A few years before, I had designed a small acoustic suspension system for the 5¼-inch Peerless TO125F polypropylene woofer and the Peerless KO10DT8 1-inch dome tweeter. I have always been fond of this little woofer, even though the free-air resonance of about 55Hz seemed a bit high. To compensate for this, I designed a box with a Q_{TC} of about 0.7 that I planned to use with a preemphasis filter to lower f_3 to about $0.49f_c$. If I remember right, the box had a net volume of about 462 cubic inches. This placed f_c at about 87Hz. I placed the 12.5dB peak of my preemphasis filter at $0.51f_c$, or an f_3 of about 44Hz. Using the crude measuring instruments I had at my disposal, I came within about 3Hz of where I had hoped to be.

Because I liked the Peerless TO125F and it was small, I decided to begin my experiments with it. (This unit is available from McGee Radio, 1901 McGee St., Kansas City, MO 64108-1891, for $15.95.) Size was a consideration because I had to build the speakers on the breakfast bar in my small apartment. The box (*Fig. 2*), which was about the same volume as my small acoustic suspension speaker, had to be an entirely different shape to allow for the tunnel.

I made the box of ¾-inch particle board (see Materials List), which is held together by Elmer's white glue. You may also use nails to strengthen the construction. The secret is to have clean, square joints. I pre-glued all the surfaces, then final glued them, using plenty of the sticky stuff. I let it run out of the joints, then wiped off the excess. I used fixtures to hold the parts square while the glue was drying. When the glue was completely dry (12 to 24 hours), I ran a bead along all the inside joints and forced the glue into the joints with my finger.

After lining the tunnel with ¼-inch felt, I glued the tunnel assembly to the rear of the front baffle, which was eventually held in place with RTV silicone rubber. If you cannot find ¼-inch felt, use several layers of thinner material available in fabric stores. A Royal blackboard eraser is made of six 1-by-5-inch pieces of ¼-inch felt. I also covered the top of the woofer box with ¼-inch charfoam, a polyester urethane material often used in lining camera cases. I just happened to have this on hand, and it might not be necessary.

The Linn Isobariks use parallel electrical connections, resulting in a very low impedance of about 2.5Ω at the crossover point. This could be a difficult load with some amplifiers. I originally used this hookup, then changed to series connections. I have been unable to detect any sonic differences, and I am sure my amplifier thanks me.

The first tweeter I used was a Peerless KO10DT8 1-inch dome. I then switched to a Radio Shack 40-1375 ($14.95) leaf tweeter. The 40-1375 reportedly has an almost resistive impedance of 8Ω. Although I did not measure the impedance, this is the finest-sounding tweeter I have ever owned.

If you use the Peerless unit, you should add a felt diffraction ring around the dome. The ring I used was a ½-inch slice of dense foam pipe insulation known as Armourflex, which was handy at the time. Rings made of ½-inch felt about 3½ inches in diameter, with a 2-inch hole in the center, are available from many suppliers, including Spectrum Loudspeakers (2136 Perth St., Toledo, OH 43607). Glue the ring in place around the dome, making sure it is centered and the joint is completely sealed. A square ring should work as well as a round one.

The tweeter just sits on top of the woofer box in a grubby, little frame of balsa wood whose only purpose is to hold the tweeter upright. Any method you choose should work as well.

I originally installed a first-order crossover at about 3kHz, using a coil and capacitor with a Zobel circuit of 4μF and 15Ω across the series pair of woofers. The 40-1375 requires a higher cutoff point, however, so I removed the coil and let the woofers run free at the high end, which the catalog says is 4kHz.

The crossover shown in *Fig. 3* may be used with either tweeter. I put a 4.5μF built-up Mylar capacitor in series with the tweeter and an attenuation circuit consisting of a 4Ω, 5 or 10W resistor in series with the tweeter and a 10Ω pot in parallel with it. To simplify things, you could replace the pot with a resistor of about 3Ω. I left the Zobel in place, as it did not seem to hurt anything. To test this, I listened

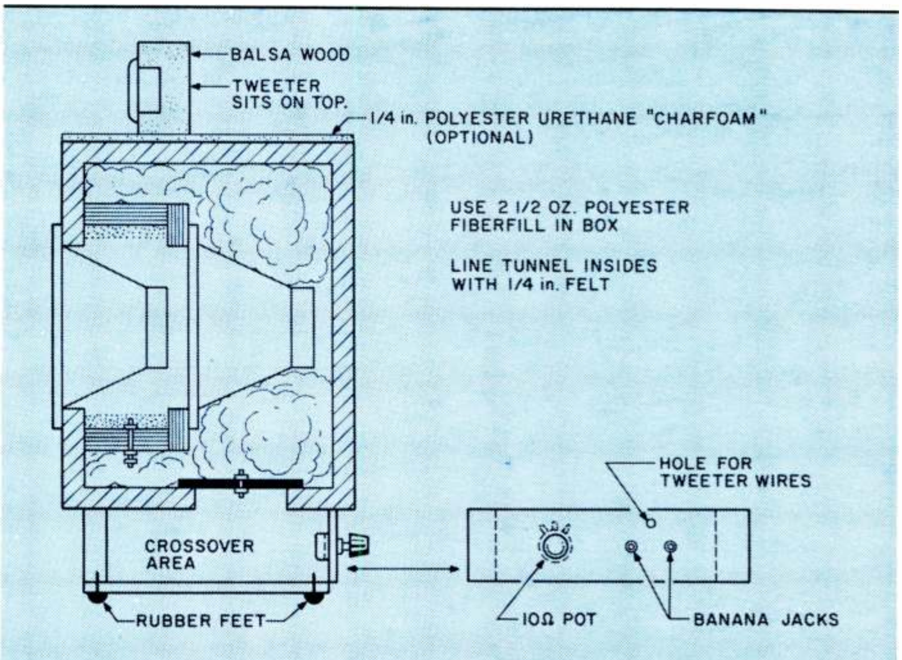


FIGURE 2: Cross-section of Project One. Note the location of the wire terminals inside the box. Make the tunnel part of the front baffle and use RTV to secure it in place.

to FM interstation hiss on another speaker with the same setup, with and without the Zobel in place. I was convinced that the hiss sounded smoother with the Zobel. I must confess, though, that some of my acquaintances tell me that I get a bit dingy at times, so I will let you solve the Zobel problem yourself.

I used about 2½ ounces of polyester fiberfill stuffing in the 450-cubic-inch net enclosure. (Do not stuff the tunnel.) Below the box proper, I made a base consisting of 3-inch-wide boards along the front and sides, set in about ¾ inch on the front and sides and extending to the rear. A ¼-inch Masonite panel fastened to the bottom of the base with rubber feet holds the crossover elements, which I wanted to keep outside the box for easy adjustment.

This speaker performed just about the way I had hoped. The sound appeared to go down smoothly to somewhere below 60Hz, and by 50Hz, there was much less sound. Listening to this speaker the first night was disconcerting in light of how I thought a 5¼-inch speaker *should* sound. In this case, f_3 was about $0.632f_c$ in a system with a Q_{TC} of 0.707, or about 58 percent lower bass. The system retained a flat curve and extremely clean and detailed sound.

After some weeks of listening, I decided to try my bass boost filter on this speaker. (See the sidebar on page 12 for an updated version of the author's boost circuit.) I arbitrarily assumed an f_3 of 55Hz and set the 12.5dB peak of my filter to $0.51f_c$. I found that this point allowed a system with a Q_{TC} of 0.797 to reach a new f_3 of about $0.49f_c$. I set the filter to about 28Hz. Listening with a frequency generator confirmed that I was in the right ball park. After several days of listening, I raised the peak to about 33Hz, and I liked the sound better. In this case, f_3 is $0.6f_c$ for a Q_{TC} of about 0.7.

The sound is open, full, extremely detailed and very musical. The bass is not just noise or a resonance; it is bass *music*, just as it should sound. This sound is comparable only to that of the few transmission-line speakers I have heard. I cannot imagine a better woofer for a large electrostatic speaker, particularly if it is biamped.

PROJECT TWO. I became involved in my second experiment when I remembered an 8-inch acoustic suspension speaker stored at my son's home.

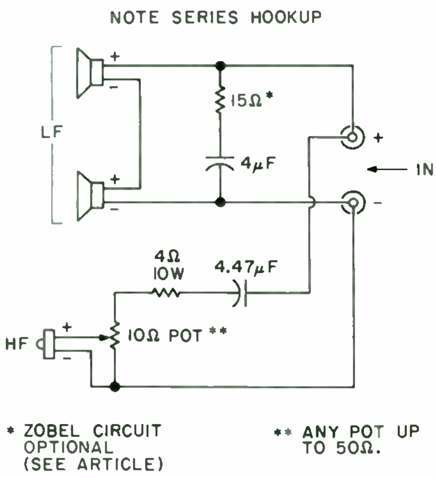


FIGURE 3: Suggested crossover for Project One. You may use this crossover with either the Peerless or Radio Shack tweeter.

I had designed this speaker for the Radio Shack 40-1021 8-inch polypropylene woofer ($19.95). My unit had the following specs: $f_{sa}=30$Hz, $Q_{TS}=0.529$, $V_{AS}=3.332$ cubic feet. I then designed a box with a Q_{TC} of 1.

This was a potentially fine speaker, but it had one serious problem: it could not handle even a reasonable amount of power. The box was just too large for the woofer. I began reducing the box volume with wooden blocks, and by the time I had filled it with 440 cubic inches of blocks, the speaker performed reasonably well. I rechecked f_c and Q_{TC} and found them to be about 64Hz and 1.13, respectively. The gross internal volume was now about 1,600 cubic inches.

Building another box, especially one of this size, in my small kitchen was not a very inviting prospect, so I decided to use the original box and build a removable tunnel on the front, over the original woofer. First I removed the woofer and crossover, then taped off the tweeter leads. I left the tweeter in its place to keep the hole plugged. The woofer leads went directly to the rear terminals, as I planned to biamp the system. I used 7 ounces of polyester fiberfill in the box. I might have been able to eliminate this, but I was not sure how the experiment would turn out, and I knew the stuffing would not do any harm.

I built the tunnel (*Fig. 4*) out of ½-inch plywood and lined it with ¼-inch felt. Acoustic foam such as charfoam might also work well. I glued the tunnel on a square of ¼-inch Masonite, which I had cut to the tunnel's inside dimensions and which served as a mounting flange. I mounted the tunnel just like a speaker, using 12 sheet-metal screws and a bead of Duxseal. I mounted electrical terminals in one of the tunnel walls and connected wires externally in series with the rear terminals. (The box is not a thing of beauty, but it works well.)

If you want to build a box from scratch, use a gross internal volume of 1,575 to 1,600 cubic inches. Don't go larger. If you go much smaller, I would be interested to find out what happens. It makes sense to install the tunnel inside the box, as in Project One. If you use the dimensions shown in *Fig. 4* (leaving off the flange and gluing the tunnel directly in place), the enclosed volume of the speaker and

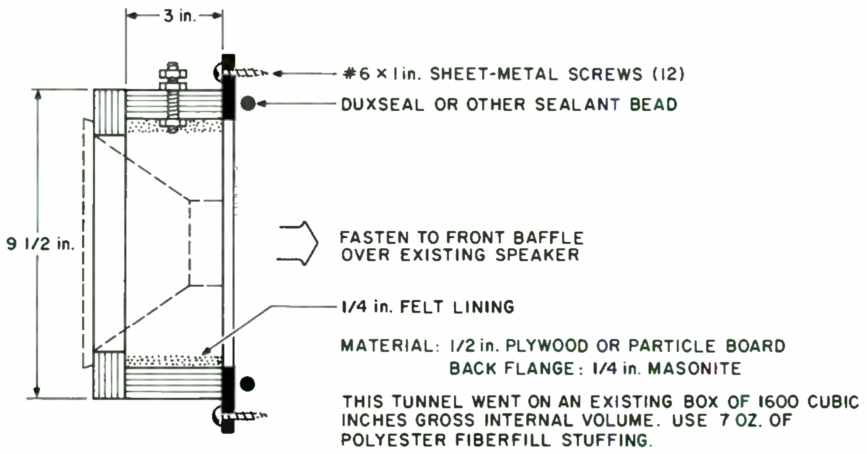


FIGURE 4: Detail of the add-on loading tunnel for Project Two. Note the terminals for the external wires to the rear.

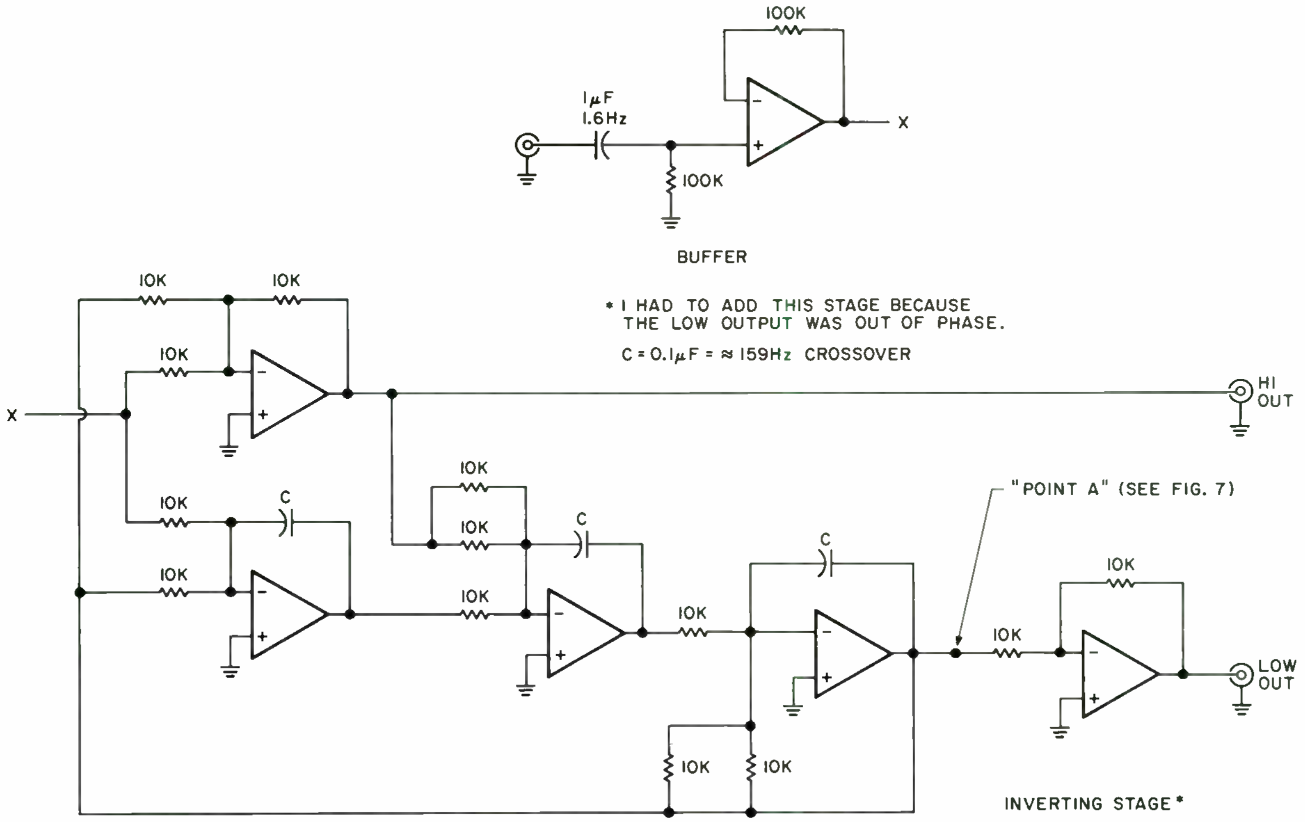


FIGURE 5: Schematic diagram of the active third-order symmetric crossover used to combine Projects One and Two.

tunnel is about 365 cubic inches. Add that much to the gross volume of the box. It will not hurt to use a few glue blocks between the tunnel and the front baffle for rigidity.

I do not think tunnel size is critical. In the cut-away drawing of the Linn SARA speaker, it looks as though the front speaker's magnet actually protrudes beyond the plane of the rear speaker's mounting ring. In any case, the tunnel dimensions I used worked fine.

MATERIALS LIST FOR PROJECT ONE

2-by-4-foot sheet of ¾-inch particle board (number of pieces)
box front and rear—two 10½ by 10½ inch
box sides—two 9 by 8½ inch
box top & bottom—two 10½ by 8½ inch
loading tunnel back—one 7 by 7 inch
loading tunnel sides—two 3 by 5½ inch
loading tunnel top & bottom—two 3 by 7 inch
base sides—two 3 by 7¾ inch
base front—one 3 by 9 inch

12-by-18-inch sheet of ¼-inch Masonite (number of pieces)
base bottom cover plate—one 9 by 8½ inch
box terminal plate—one 5 by 5 inch
base rear panel—one 3 by 9 inch

To combine Projects One and Two, I built an active third-order symmetrical crossover along the lines of the Ashley and Kaminsky design *(Fig. 5)*.[2] Using 0.1µF capacitors at points C will produce a crossover point of about 159Hz. You can scale that figure to produce any crossover point you wish. I can detect no sonic indication of the sound passing from one speaker to the other at crossover.

Perhaps you might use the Isobarik woofer with a conventional midrange. If so, a higher crossover point is required. Use the following capacitor values for the indicated crossover points: 0.032µF for 500Hz, 0.022µF for 722Hz and 0.01µF for 1,590Hz. The Radio Shack woofer works well at any of these points. I would suggest using the best midrange you can find, perhaps one of the new polypropylene dome units.

Figure 6 shows the power supply I used for the crossover. Any similar bipolar unit should work as well. If you want to use one woofer, you may use the summing amplifier[3] in *Fig. 7* with two of the electronic crossovers in *Fig. 5*. I recommend a crossover of 160Hz or so, which means that you cannot use a system with a normal midrange unit. Since this summing amplifier inverts, you need not use the final inverting stage in *Fig. 5*. Attach the summing circuit to the place marked point A. If you want to use this circuit for some other purpose, you must also use the buffer section in *Fig. 5* and the power supply in *Fig. 6*.

Isobarik speakers tend to be chunky (and ugly) in shape. If most other speakers were made that way, they would probably be subject to standing-wave problems. Isobariks do not have this problem because the internal tunnel acts as a baffle impeding the standing waves. The best way to handle the shape, both aesthetically and practically (and maybe even acoustically), is to make separate bass, midrange and treble units and stack them as in KEF's large monitor speakers. If you do this, you might want to place the crossover network for Project One in some other location to lower the stack height and bring the speakers closer together. Perhaps you could place the crossover units in the base used to raise the woofer to the correct height. The crossover will then remain easily accessible.

Rounding off the corners of the boxes or changing the woofer height might result in further sonic improvement. In my original design, the center of the woofer was 16 inches from the floor. Changing this height might make a difference in the sonic quality of the bass. If so, you could use a stand to raise the woofer.

I put my 5½-inch Isobarik unit on top of my 8-inch unit. After extended listening tests, I decided that this is the most pleasing and satisfying speaker system I have ever built. Using a frequency generator, the sound goes down very smoothly to somewhere in the upper twenties, with much less doubling than I would expect from such a lightweight speaker.

A speaker with a Q_{TC} of 1.13 should have an f_3 of about $0.75f_c$. Since the f_c in this box is about 64Hz, that would make f_3 about 48Hz. From listening to the swept speaker, I believe the new f_3 is around 30Hz. The bass then becomes 0.625 of the original f_3 (48Hz), or 60 percent. This compares favorably to the 0.632, or 58 percent, lowering I achieved with the 5¼-inch unit. I have a feeling that because the two woofers work against each other in

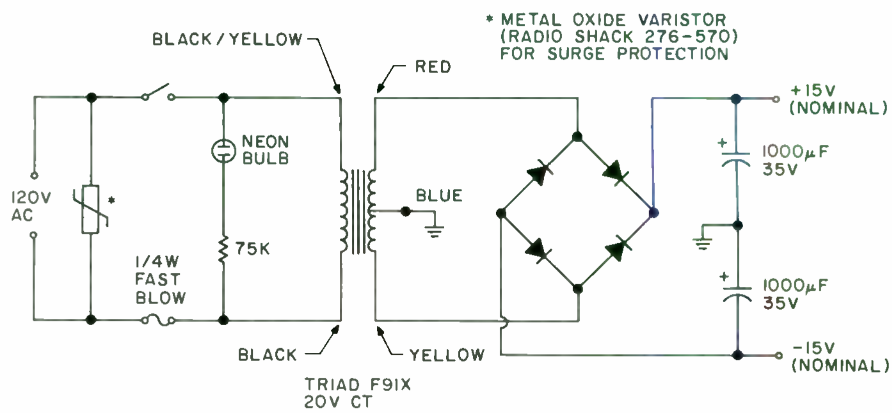


FIGURE 6: Schematic diagram of the power supply for the active crossover in Project Two. Any similar bipolar supply will work just as well.

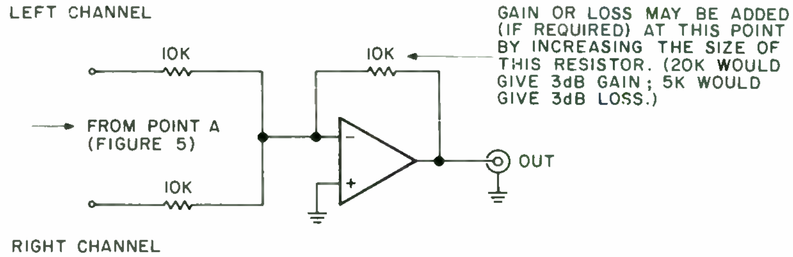


FIGURE 7: This summing amplifier allows you to use one Project Two woofer with two Project One satellites. Note that this circuit also inverts, eliminating the inverting stage in Fig. 5.

times of nonlinearity, the curves of varying Q's will be more alike than is the case with acoustic suspension speakers. If this were true, it would mean that for a given response curve, several box sizes would be possible.

Thanks to the kindness of John Alm of Sound Goods, an audiophile store in West San Jose, California, I was able to listen to my speaker system for a while in a superior environment with high-quality equipment. I listened to Moussorgsky's "Pictures at an Exhibition" and "A Night on Bald Mountain" on compact disk. I was not prepared for the performance that came from my ugly speakers. Even though the sound was in mono (I have only one of these beasts), I have never heard such startling monolithic clarity and such exquisite and delicate tonal separation. I had no sensation of listening to speakers, especially in the bass, and instead felt as though I was hearing live instruments. Even at volume levels higher than I normally dare to use, the cone remained in control, with no observable problems. I switched to the 5¼-inch unit (as a full range) for a bit, and it was the same, except for the very deep bass, which of course was not there.

"How could these be cheap Radio Shack speakers?" I thought. The answer is that a speaker is just a speaker. If it finds the right environment, it will perform well; it it doesn't, it won't. It is staggering to think of a system with a couple of Gold Sound 12-inch polypropylene woofers with an efficiency of 94dB/watt/meter, a rating of 120W each and an f_{sa} of 18Hz. That is one nice thing about the Isobarik principle: twice the speakers can handle twice the watts. The Linn DMS and PMS speakers apparently use the KEF B139 (highly modified, they claim) with useful output at 10Hz. Linn's philosophy is that all sonic wavefronts begin at DC, so the better the speaker's performance is at very low frequencies, the more realistic it will sound.

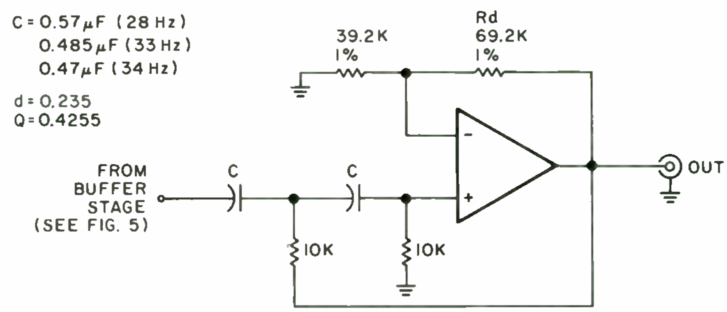
I recently constructed a variant of this design, which uses a push/pull configuration that eliminates the need for a loading tunnel. The woofers are a pair of Radio Shack 4-inch units ($9.95 each). The net V_B is about 160 cubic inches, which might be a little too large. The system weighs about the same as a bowling ball because of a new damping material I have developed. I prefer its sound to that of most commercial speakers. I won't go into all the details, but I hope I have interested some of you in the possibilities of Isobarik loading and that some serious experimentation and dialogue will follow.

A Bass Boost Circuit

The bass boost circuit I used was about ten years old and quite outdated. As a replacement, I suggest the circuit[3] in *Fig. S-1,* which requires that the buffer stage in *Fig. 5* be inserted ahead of it. The resistors can be 5 percent tolerance, but it might be better to use 1 or 2 percent ones for the damping resistors. You may scale the capacitors to change the peak point. A 0.47μF value results in a peak of about 34Hz.

This is a very versatile circuit. To change the damping, you need only change the value of resistor Rd. Everything else remains the same. To make a useful experimental instrument out of it, I suggest using a four-position switch at Rd. At the first position, install Rd as shown. This produces a damping factor of 0.235 and a peak of about 12.5dB (I aimed for 12dB). In the second position, install a 58.8k resistor, producing a damping factor of 0.5 and a 6dB peak. This is just right for equalizing a sixth-order vented system. The third position would have a 22.6k resistor. It yields a damping factor of 1.414, which is the maximally flat Butterworth curve. This would be most useful for a subsonic rumble filter. At position four, you could place a couple of handily spaced banana jacks that would accept any resistor, mounted on banana plugs. A pot might also be handy here.

I also use banana jacks for the capacitors. Use two buffers and two filters for stereo. This filter requires the bipolar power supply in *Fig. 6* (one only, of course, for either mono or stereo).



ICs-USE NATIONAL LF353N DUAL BI FET; RADIO SHACK 276-1715 (SOMETIMES TL082 IS SUBSTITUTED.)

FOR STEREO: TL084C QUAD BI FET; RADIO SHACK 276-1714

FIGURE S-1: Bass boost circuit for use with Project One. The peak is about 12.5dB.

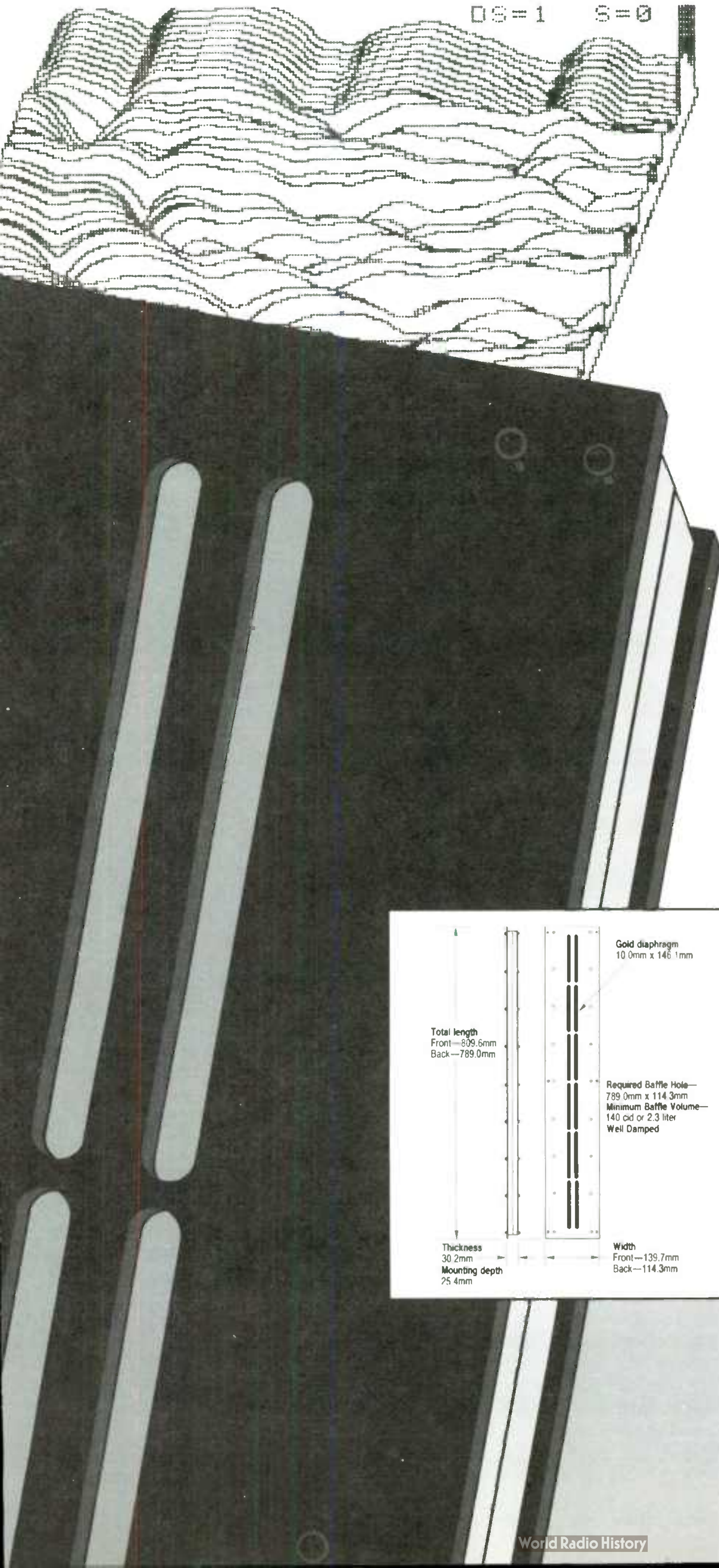
ABOUT THE AUTHOR

John Cockroft is a senior cryogenics technician at the Stanford Linear Accelerator Center, where he has worked for 22 years. His interest in designing and constructing loudspeaker systems dates from around 1958.

REFERENCES

1. Small, Richard H., "Closed-Box Loudspeaker Systems," *JAES,* December 1972 and January/February 1973.
2. Ashley, J. Robert and Allan L. Kaminsky, "Active and Passive Filters as Loudspeaker Crossover Networks," *JAES,* June 1971 (Fig. 11 and Table IV).
3. Lancaster, Don, *Active-Filter Cookbook,* Howard W. Sams & Co., Inc., 1975.

FAST REPLY #HH915



THE GOLDEN RULES

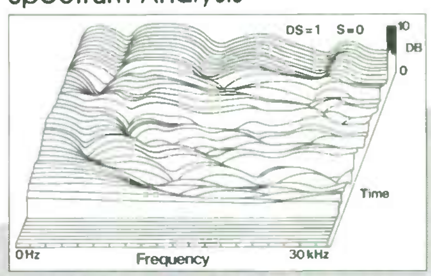
Historically, the ribbon loudspeaker has been an interesting alternative to high-performance loudspeaker design since the early 1930's. Though cost-effective production technology has not been available until now, ribbons were seen by many as an attractive solution to the inherent problems which continue to plague even the best dynamic, electrostatic, and field-type transducers today. Vacuum technology and computerization has enabled Gold Ribbon Concepts to present an affordable esoteric loudspeaker for the 21st century. For example. . . . . .

Most dynamic cone loudspeakers are rated at approximately 3% Total Harmonic Distortion (THD). The GOLD 3.0 ribbon satellite's THD is 1/10th (.3%) this amount. And with a moving mass close to a gram per 1100 grams of magnet, the rise and decay speed of the GOLD 3.0 is so fast that it can best be exhibited as a live sound reinforcement transducer, where the dynamic range exceeds even digital recording capabilities.

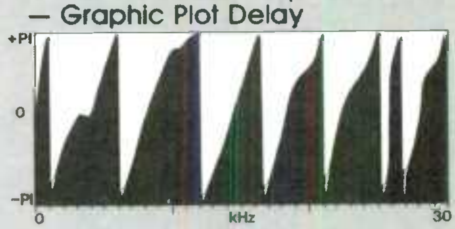
The GOLD 3.0 operates from 200 hz to 30kHz, thereby releasing speaker designers from the complex phase and distortion problems which result from the use of cross-over networks in the delicate vocal region above 200 Hz. And for unparalleled accuracy, Gold Ribbon Concepts uses computerized manufacturing techniques to match each GOLD 3.0 voice coil to within an incredible thickness tolerance of 25 atoms.

The performance of the Gold Ribbon Concepts GOLD 3.0 ribbon satellite is best illustrated in the full expression of a beautiful piece of music.

GOLD 3.0 Time Delay Spectrum Analysis



GOLD 3.0 Phase Response — Graphic Plot Delay



GOLD RIBBON CONCEPTS

211 East 11th Street
Iowa City, Coralville, IA, 52241, USA
(319) 351-9144/1-800-841-GOLD

PART III

PASSIVE CROSSOVER NETWORKS

ACTIVE REALIZATIONS OF TWO-WAY DESIGNS

BY ROBERT M. BULLOCK III
Contributing Editor

All-pass and constant-power networks are ideal for home speaker builders because you can design them from passive components by formula and they have response properties considered desirable for loudspeaker applications. You may also actively realize these networks by formula to produce the same desirable properties, as well as some additional benefits not possible with the passive versions.

This time, I want to show you how to actively implement the two-way networks in Part I of this series (*SB* 1/85, p. 13) using the 30Hz Rumble Filter kit from Old Colony (KF-6, $19.75; PO Box 243, Peterborough, NH 03458). Next time, I will do the same for the three-way networks in Part II.

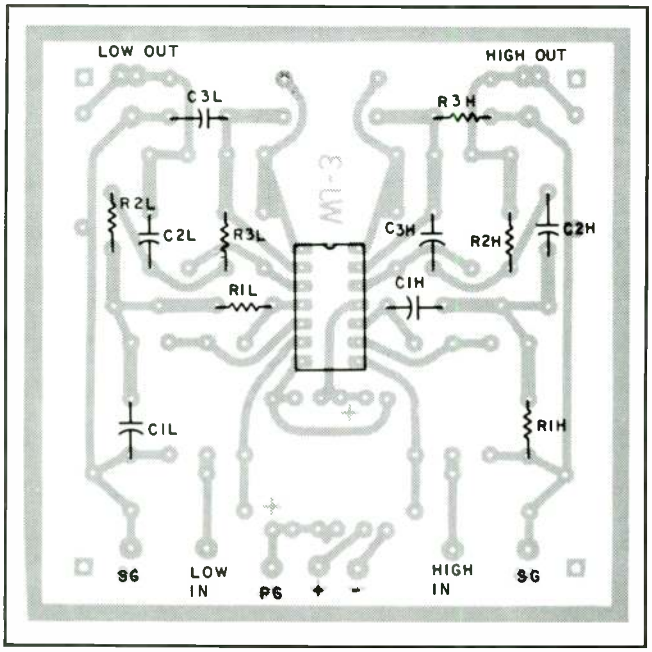


FIGURE 1: Stuffing guide for the rumble filter circuit, showing only the filter-shaping components. The left circuit is configured as a low-pass filter and the right circuit as a high-pass filter.

BENEFITS & DETRIMENTS. I strongly recommend that you read Martin Colloms's excellent account of the many benefits of using active crossover networks.[1] One benefit that he does not discuss is also of tremendous value to the home builder: you can accurately build active networks by formula without making the problematic assumptions that passive designs require.

Remember that passive ladder design is based on the assumptions that loudspeaker loads are resistors and that crossover inductors have zero resistance. At best, the former assumption can be only approximately met in practice, and even high-quality inductors have enough resistance to introduce distortion into the filter response. Thus, the response accuracy of a passive network is inevitably compromised by loudspeaker impedance fluctuations and inductor losses.

Active networks do not have inductors, and they are effectively isolated from loudspeaker impedance fluctuations by the intervening amplifier. Thus, you can avoid the two most difficult problems of passive design. As an additional benefit, the low-power resistors and small capacitors needed for active networks are available at closer tolerances than the large inductors and capacitors used in passive networks, so it is easier to produce a highly accurate active crossover.

My initial decision to go active was motivated by the above considerations in conjunction with subwoofer design. To cross over to a subwoofer at 150Hz, I found that large crossover inductors with more than an ohm loss were necessary. Further, the subwoofer impedance fluctuations were still rather significant at 75Hz, which led me to believe that they should be equalized out. That is when I found that equalizing a woofer's mechanical and acoustical characteristics requires extremely large inductors (approximately 25mH) and capacitors (approximately 2,000μF). I concluded that a passive crossover designed by formula was not practical for such a low crossover frequency.

I then purchased a second amplifier and built an active network using the rumble filter kit. I was impressed by the accuracy of the crossover and by the ease with which it could be built. This led me to buy a third amplifier and convert to three active channels. Now I use passive networks only when designing for someone else or when I use a fourth channel (I cannot yet afford a fourth amplifier).

My conversion to active crossover networks has improved my system's sound more than any other modifica-

TABLE 1			
FIRST-ORDER NETWORK			
Fc	C(UF)	R3L	R3H
100	.0150	106103	106103
150	.0150	70736	70736
200	.0150	53052	53052
300	.0150	35368	35368
400	.0150	26526	26526
500	.0150	21221	21221
1000	.0150	10610	10610
2000	.0015	53052	53052
3000	.0015	35368	35368
4000	.0015	26526	26526
5000	.0015	21221	21221
8000	.0015	13263	13263
10000	.0015	10610	10610

TABLE 2			
SECOND-ORDER APC NETWORK			
Fc	C(UF)	R2,3L	R2,3H
100	.0150	106103	106103
150	.0150	70736	70736
200	.0150	53052	53052
300	.0150	35368	35368
400	.0150	26526	26526
500	.0150	21221	21221
1000	.0150	10610	10610
2000	.0015	53052	53052
3000	.0015	35368	35368
4000	.0015	26526	26526
5000	.0015	21221	21221
8000	.0015	13263	13263
10000	.0015	10610	10610

tion I have made. I have noticed a dramatic increase in detail and texture, especially at higher frequencies. The inevitable pops and ticks from records are not nearly as objectionable because even though they are still present, the duration of each one is so much shorter. This makes the apparent average noise level much lower and is perhaps responsible for a feeling of increased dynamic range. Overall, the sound reproduction through active networks has much more clarity and impact than it does through passive networks.

The main objection to active crossover networks is economic—i.e., you need a separate amplifier for each crossover channel. My solution was to use three amplifiers costing a total of $600 rather than one $600 super amplifier. My feeling is that inexpensive amplifiers do not introduce as much distortion as passive crossover networks. I urge you to look into this alternative and am sure you will be gratified by the sonic results if you follow through with it. Now on to the details of active two-way design.

RUMBLE FILTER AS CROSSOVER. You can easily convert the 30Hz rumble filter kit into a two-way first, second or third-order crossover network for a single stereo channel by adding a few resistors and capacitors. You need two kits for fourth-order net-

TABLE 3					
SECOND-ORDER CPC NETWORK					
Fc	C(UF)	R2L	R3L	R2H	R3H
100	.0150	75026	75026	75026	150053
150	.0150	50018	50018	50018	100035
200	.0150	37513	37513	37513	75026
300	.0150	25009	25009	25009	50018
400	.0150	18757	18757	18757	37513
500	.0150	15005	15005	15005	30011
1000	.0150	7503	7503	7503	15005
2000	.0015	37513	37513	37513	75026
3000	.0015	25009	25009	25009	50018
4000	.0015	18757	18757	18757	37513
5000	.0015	15005	15005	15005	30011
8000	.0015	9378	9378	9378	18757
10000	.0015	7503	7503	7503	15005

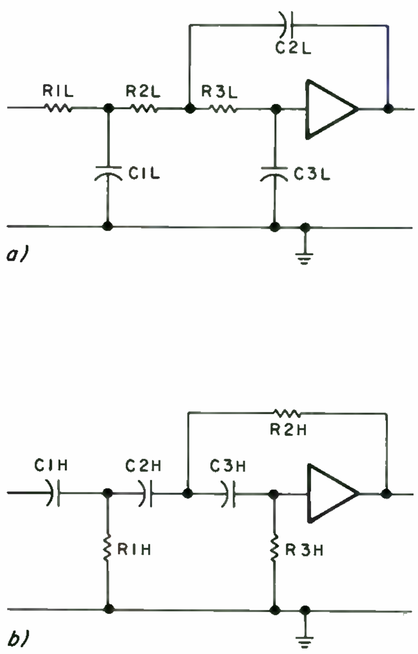


FIGURE 2: The filter-shaping circuitry. Figure 2a is the low-pass circuit and Fig. 2b the high-pass circuit.

TABLE 4					
SECOND-ORDER EQUAL COMPROMISE NETWORK					
Fc	C(UF)	R2L	R3L	R2H	R3H
100	.0150	40935	137508	89222	126179
150	.0150	27290	91672	59481	84119
200	.0150	20468	68754	44611	63089
300	.0150	13645	45836	29741	42060
400	.0150	10234	34377	22305	31545
500	.0150	8187	27502	17844	25236
1000	.0150	4094	13751	8922	12618
2000	.0015	20468	68754	44611	63089
3000	.0015	13645	45836	29741	42060
4000	.0015	10234	34377	22305	31545
5000	.0015	8187	27502	17844	25236
8000	.0015	5117	17189	11153	15772
10000	.0015	4094	13751	8922	12618

TABLE 5

THIRD-ORDER NETWORK

Fc	C(UF)	R1L	R2L	R3L	R1H	R2H	R3H
100	.0150	109282	779793	140171	76188	29915	524094
150	.0150	72855	519862	93447	50792	19943	349396
200	.0150	54641	389897	70085	38094	14958	262047
300	.0150	36427	259931	46724	25396	9972	174698
400	.0150	27321	194948	35043	19047	7479	131024
500	.0150	21856	155959	28034	15238	5983	104819
1000	.0150	10928	77979	14017	7619	2992	52409
2000	.0015	54641	389897	70085	38094	14958	262047
3000	.0015	36427	259931	46724	25396	9972	174698
4000	.0015	27321	194948	35043	19047	7479	131024
5000	.0015	21856	155959	28034	15238	5983	104819
8000	.0015	13660	97474	17521	9524	3739	65512
10000	.0015	10928	77979	14017	7619	2992	52409

works. I suggest you familiarize yourself with the rumble filter by reading Jung's article,[2] which covers its design and capabilities. The kit contains a circuit board with two identical filter circuits and all the necessary components, except the shaping resistors and capacitors.

I put the low-pass filter on the left and the high-pass filter on the right of *Figure 1*, which shows only the board's filter-shaping components. (See *The Audio Amateur* 4/75, p. 17, for the complete, original stuffing guide.) The suffixes L and H are used to distinguish between the low-pass and high-pass filter-shaping components. *Figure 2* shows the schematics for the portion of the circuits containing these components. The schematics are for third-order filters. For a second-order crossover network, jumper R1L and C1H and delete C1L and R1H. If you jumper R2L and C2H and delete C2L and R2H, you have a first-order crossover. These arrangements for first and second order are somewhat different from the ones Jung uses, but they are equivalent and make it physically easier to place the parts.

The filter alignment procedure I used starts with the selection of a convenient stock value C for one of the capacitors. The other capacitors are then easily obtainable multiples of C. The resistor values are calculated from the radian crossover frequency $\omega_c = 2\pi f_c$ and C. The nearest 1 percent resistor values are more than adequate. Even with 10 percent capacitors, the active network should be more accurate than a comparable passive network. If the capacitors are measured and matched, even greater accuracy is possible.

FIRST-ORDER NETWORK. This is a simple case because there is only one type of crossover network and hence only one set of design formulas. If C is a convenient, standard capacitor value, the design formulas for the low-pass and high-pass sections are as follows:

Low Pass	High Pass
C1L = omit	C1H = jumper
C2L = omit	C2H = jumper
C3L = C	C3H = C
R1L = jumper	R1H = omit
R2L = jumper	R2H = omit
R3L = $1/(C\omega_c)$	R3H = $1/(C\omega_c)$

I will use this format for all the networks. Note that you may use different values for C in the low and high-pass sections, but making them equal reduces the number of different component values needed.

SECOND-ORDER NETWORKS. For an all-pass crossover (APC), with a convenient C value chosen, the formulas are as follows:

Low Pass	High Pass
C1L = omit	C1H = jumper
C2L = C	C2H = C
C3L = C	C3H = C
R1L = jumper	R1H = omit
R2L = $1/(C\omega_c)$	R2H = $1/(C\omega_c)$
R3L = R2L	R3H = R2H

For a constant-power crossover (CPC) or a compromise network, let A be defined as in *Fig. 8* of Part I (p. 15). Calculate M as follows:

$$M = \frac{(A + \sqrt{A^2 - 2})}{2}$$

Then, with a convenient choice for C, the formulas are:

Low Pass	High Pass
C1L = omit	C1H = jumper
C2L = 2C	C2H = C
C3L = C	C3H = C
R1L = jumper	R1H = omit
R2L = $1/(2MC\omega_c)$	R2H = $A/(2C\omega_c)$
R3L = $M/(C\omega_c)$	R3H = $2/(AC\omega_c)$

TABLE 6

FOURTH-ORDER APC NETWORK (BOTH BOARDS)

Fc	C(UF)	R2L,R3L	R2H	R3H
100	.0150	75026	75026	150053
150	.0150	50018	50018	100035
200	.0150	37513	37513	75026
300	.0150	25009	25009	50018
400	.0150	18757	18757	37513
500	.0150	15005	15005	30011
1000	.0150	7503	7503	15005
2000	.0015	37513	37513	75026
3000	.0015	25009	25009	50018
4000	.0015	18757	18757	37513
5000	.0015	15005	15005	30011
8000	.0015	9378	9378	18757
10000	.0015	7503	7503	15005

- Designed by Sidereal Akustic specifically for high quality audio circuitry
- Metalized polypropylene
- Extremely low DA, DF and ESR
- Multi-guage stranded, oxygen free high-purity copper leads
- Special oxidation inhibiting PVC lead insulation
- Hand-soldered lead termination
- Extended contact area lead attachment
- Non-permeable lead material for low hysteresis distortion
- Superior winding and lead attachment for minimum self inductance
- Exceptional intertransient silence
- D.C. operation: blocking, coupling, bypass, R.F. circuitry, R.C. circuits, computers and high sensitivity instrumentation
- A.C. operation: power factoring, line filtering, high level audio (loudspeaker filter networks)
- High mechanical stability for freedom from "singing" and microphonic effects
- Low leakage, high insulation resistance
- No bypass required — If bypassed with capacitors from other manufacturers, there will be no benefit. In fact, we have occasionally observed the SiderealKap to assume the inferior qualities of capacitors from other manufacturers, when these capacitors are used to bypass the SiderealKap. If you must bypass, use a small value SiderealKap.
- 100% quality testing
- Designed to meet thermal shock, vibration and moisture resistance specifications
- Self-healing, high reliability design
- Competitive O.E.M. and distributor pricing
- Most values now in stock for quick delivery

™ Sidereal Akustic Audio Systems Inc.
1969 Outrigger Way, Oceanside, CA 92054 [619] 722-7707

FAST REPLY #HH778

THIRD-ORDER NETWORKS. Only one type of third-order network exists. For a convenient choice of C, the formulas are:

Low Pass	High Pass
$C1L = C$	$C1H = C$
$C2L = C$	$C2H = C$
$C3L = 0.1C$	$C3H = C$
$R1L = 1.02996/(C\omega_c)$	$R1H = 0.718057/(C\omega_c)$
$R2L = 7.34938/(C\omega_c)$	$R2H = 0.281943/(C\omega_c)$
$R3L = 1.32108/(C\omega_c)$	$R3H = 4.93947/(C\omega_c)$

FOURTH-ORDER NETWORKS. Now things get a little more involved because you need two kits and there is more than one type of crossover network. The procedure is to build two second-order crossover networks, then cascade them to form the fourth-order network. Thus, you need four kits for two stereo channels.

The fourth-order APC is the easiest case because it uses two identical boards. For a convenient C value, use the following formulas:

Low Pass	High Pass
$C1L = \text{omit}$	$C1H = \text{jumper}$
$C2L = 2C$	$C2H = C$
$C3L = C$	$C3H = C$
$R1L = \text{jumper}$	$R1H = \text{omit}$
$R2L = 1/(\sqrt{2}C\omega_c)$	$R2H = 1/(\sqrt{2}C\omega_c)$
$R3L = R2L$	$R3H = \sqrt{2}/(C\omega_c)$

Build two boards with these formulas, then connect the low-pass output of one board to the low-pass input of the other. Do the same with the high-pass sections. You have now cascaded second-order filters to obtain fourth-order filters.

You use the same cascading procedure for the CPC and compromise networks as you did for the APC, but now the two boards are not identical. For a compromise crossover, start by letting K be defined by the following formula:

$$K = 1/10^{(P/20)}$$

where P is the peaking in decibels in the voltage magnitude response $(0<P<3)$. For a CPC, K equals $\sqrt{2}/2$. Use these values of K to calculate A1 and A2 as follows:

$$A1 = \sqrt{1+K} + \sqrt{1-K}$$

$$A2 = \sqrt{1+K} - \sqrt{1-K}$$

Then use these values to calculate M1 and M2:

TABLE 7A

FOURTH-ORDER CPC NETWORK (FIRST BOARD)

Fc	C(UF)	R2L	R3L	R2H	R3H
100	.01000	8882	285198	147040	172268
150	.01000	5921	190132	98027	114845
200	.01000	4441	142599	73520	86134
300	.01000	2961	95066	49013	57423
400	.01000	2220	71300	36760	43067
500	.01000	1776	57040	29408	34454
1000	.01000	888	28520	14704	17227
2000	.00100	4441	142599	73520	86134
3000	.00100	2961	95066	49013	57423
4000	.00100	2220	71300	36760	43067
5000	.00100	1776	57040	29408	34454
8000	.00100	1110	35650	18380	21534
10000	.00100	888	28520	14704	17227

$$M1 = \frac{(A1 + \sqrt{(A1)^2-0.4})}{2}$$

$$M2 = \frac{(A2 + \sqrt{(A2)^2-0.4})}{2}$$

Now choose a convenient C value and determine the component values for the first board from the following equations:

Low Pass	High Pass
$C1L = \text{omit}$	$C1H = \text{jumper}$
$C2L = 10C$	$C2H = C$
$C3L = C$	$C3H = C$
$R1L = \text{jumper}$	$R1H = \text{omit}$
$R2L = 1/[10(M1)C\omega_c]$	$R2H = A1/(2C\omega_c)$
$R3L = M1/(C\omega_c)$	$R3H = 2/[(A1)C\omega_c]$

For the second board, use these formulas:

Low Pass	High Pass
$C1L = \text{omit}$	$C1H = \text{jumper}$
$C2L = 10C$	$C2H = C$
$C3L = C$	$C3H = C$
$R1L = \text{jumper}$	$R1H = \text{omit}$
$R2L = 1/[10(M2)C\omega_c]$	$R2H = A2/(2C\omega_c)$
$R3L = M2/(C\omega_c)$	$R3H = 2/[(A2)C\omega_c]$

Finally, cascade the two boards, preferably with the A1 board first, and you are done.

BONUS NETWORK. Because the rumble filter kit allows you to work with third-order filters as basic sections, you can realize one more network without any new formulas. Use the third-order formulas to build two identical boards and then cascade them. The result is a sixth-order APC for one stereo channel. If you decide to use this network, remember to reverse the polarity of the tweeter just as you do for second-order networks.

CALCULATIONS & DESIGN TIPS. I think you will agree that the calculations for an active network are a little simpler than those for a comparable passive design. If you are willing to accept a preselected crossover frequency and C value, you can use *Tables 1–8* to read off resistor values. The second and fourth-order compromise *Tables 4* and *8* are for the equal compromise—i.e., the combined magnitude response has a 1.5dB peak, and the combined

TABLE 7B

FOURTH-ORDER CPC NETWORK (SECOND BOARD)

Fc	C(UF)	R2L	R3L	R2H	R3H
100	.01000	26606	95206	60906	415892
150	.01000	17737	63471	40604	277261
200	.01000	13303	47603	30453	207946
300	.01000	8869	31735	20302	138631
400	.01000	6651	23802	15226	103973
500	.01000	5321	19041	12181	83178
1000	.01000	2661	9521	6091	41589
2000	.00100	13303	47603	30453	207946
3000	.00100	8869	31735	20302	138631
4000	.00100	6651	23802	15226	103973
5000	.00100	5321	19041	12181	83178
8000	.00100	3326	11901	7613	51986
10000	.00100	2661	9521	6091	41589

TABLE 8A

FOURTH-ORDER EQUAL COMPROMISE NETWORK (FIRST BOARD)

Fc	C(UF)	R2L	R3L	R2H	R3H
100	.01000	9380	270044	139712	181303
150	.01000	6253	180029	93141	120869
200	.01000	4690	135022	69856	90652
300	.01000	3127	90015	46571	60434
400	.01000	2345	67511	34928	45326
500	.01000	1876	54009	27942	36261
1000	.01000	938	27004	13971	18130
2000	.00100	4690	135022	69856	90652
3000	.00100	3127	90015	46571	60434
4000	.00100	2345	67511	34928	45326
5000	.00100	1876	54009	27942	36261
8000	.00100	1173	33756	17464	22663
10000	.00100	938	27004	13971	18130

power response has a 1.5dB dip. I have postponed a computer-assisted design program until the next time so that I can do both two-way and three-way networks in the same program.

The value of C you choose determines the range of the resistor values. I usually try to choose it so that the smallest resistor will be 1kΩ or more. You should consult Jung[2] for the best type of capacitors and resistors to use. Once you have chosen C, there are only three other possible capacitor values—0.1C, 2C and 10C. Clearly, if C is a stock value, so are 0.1C and 10C. You can obtain 2C by paralleling two capacitors of value C.

The rumble filter kit does not include the necessary ±15V power supply. I have used the Old Colony Gately Power Supply (KF-3, $27), and Ed Dell recommends the Suzler preamp power supply from Old Colony (KL-4, four versions available). I am sure many of you will have your own ideas regarding power supplies.

As for op amps, the rumble filter kit comes with a 4136 device, which I replaced with a TL074.

Another consideration is sensitivity matching. The first stage of each rumble filter kit is set up to supply gain with the resistors RI and RF in Jung's original stuffing guide. The amount of gain is given by the following formula:

$$G = 20\log(1 + RF/RI)$$

This makes it very easy to adjust not only for different woofer and tweeter sensitivities, but also for different amplifier gains. I set up mine with RF equal to 2.21kΩ and a 10k pot for RF. I then calibrated the pot dial for gains up to 14dB. That way, I can use the crossover with a variety of loudspeakers and amplifiers.

OTHER CROSSOVER KITS. For those of you who are willing to use a third-order active network at crossover frequency 60, 120, 250, 500, 1k, 2k, 5k or 10kHz, Old Colony offers the Jung Electronic 2-Way Crossover (SBK-CIA, $24.75), which includes not only the rumble filter kit, but also the necessary filter-shaping components. The low-pass filter in this kit is realized by the "equal-resistor" method, so the kit component values are different from those given by my third-order formulas. The "convenient capacitor value" method I used has the potential for greater accuracy without the need for constructing custom capacitor values, but I am sure you will find a vast improvement in performance regardless of the method used to realize the crossover.

I would also like to mention the Old Colony Electronic Crossover, Kit A (KC-4A, $8). If you can use crossover frequency 60, 120, 240, 480, 960, 1920, 5k or 10kHz, a single kit gives you a second-order CPC, including the filter-shaping components. If you cascade two of these kits at the same crossover frequency, you have a fourth-order APC. You may also use this kit to realize the other second and fourth-order networks by applying my formulas and supplying your own filter-shaping components. The filter-shaping components are R1–R4 and C1–C4. All you have to do is make the following identifications:

Low Pass	High Pass
C1 = C2L	C3 = C2H
C2 = C3L	C4 = C3H
R3 = R2L	R1 = R2H
R4 = R3L	R2 = R3H

For a fourth-order network, you must build two kits and cascade them. Finally, you may use the kit for first-order designs with the following formulas:

Low Pass	High Pass
C1 = omit	C3 = jumper
C2 = C3L	C4 = C3H
R3 = jumper	R1 = omit
R4 = R3L	R2 = R3H

If you want more information about these two kits, consult Ed Dell's rework of his *SB* article[3] on page 20. Among other things, he discusses op amp choices, the supplementary circuitry in each kit and using the kits for three-way active networks. I plan to cover this last topic in detail in the next installment.

TABLE 8B

FOURTH-ORDER EQUAL COMPROMISE NETWORK (SECOND BOARD)

Fc	C(UF)	R2L	R3L	R2H	R3H
100	.01000	18977	133481	76229	332293
150	.01000	12651	88987	50819	221529
200	.01000	9488	66740	38114	166147
300	.01000	6326	44494	25410	110764
400	.01000	4744	33370	19057	83073
500	.01000	3795	26696	15246	66459
1000	.01000	1898	13348	7623	33229
2000	.00100	9488	66740	38114	166147
3000	.00100	6326	44494	25410	110764
4000	.00100	4744	33370	19057	83073
5000	.00100	3795	26696	15246	66459
8000	.00100	2372	16685	9529	41537
10000	.00100	1898	13348	7623	33229

REFERENCES

1. Colloms, Martin, *High Performance Loudspeakers, Second Edition,* Halsted Press (New York), 1980.

2. Jung, W.G., "A L.F. Garbage Filter," *The Audio Amateur* 4/75, p. 14.

3. Dell, E.T., "Electronic Crossovers Revisited," *SB* 3/82, p. 22.

ELECTRONIC CROSSOVERS REVISITED

BY EDWARD T. DELL

Many *Speaker Builder* readers have written to us in the wake of Walt Jung's series in *TAA* on IC distortion (Issues 2 and 3, *TAA* 1978) asking why we don't update past projects that contain ICs, especially those using the 741.

The 741 has become an embarrassment and needs replacing by better descendants, especially in the electronic crossover we published in *TAA* 2/72 (see *Fig. 1*). We have updated the Old Colony circuit card for this crossover to accommodate dual in-line eight-pin packages. See *Fig. 2* for the new card layout. Jung advises using either the National LF351 or the Texas Instruments TL071 in this circuit as a direct replacement plug-in for the 741.

When we come to the three-way crossover, however, we discover a snag. Our little design is a two-pole configuration giving a 12dB/octave slope. A growing body of evidence shows the 12dB slope has problems avoided by 18dB and 6dB slopes. Our crossover has that drawback as well as some notch cancellation in the way we derive the center channel by cancellation of 180-degree out-of-phase frequencies.

Note that in *Fig. 1*, IC3 inverts the main signal and feeds it to the uninverted low-pass and high-pass portions of the audio band. The output of IC4 is thus a center channel consisting of the main signal with the high and low signals cancelled out.

PARTS LIST
(FIG. 1)

All resistors ±1% metal film.
C1, C2 ±5% low leakage polystyrene or polypropylene
C3, C4 ±1% polystyrene
IC1, 2, 3, 4 μALF35IN or MC34081P
Solder all posts to copper foil for best long-term performance.

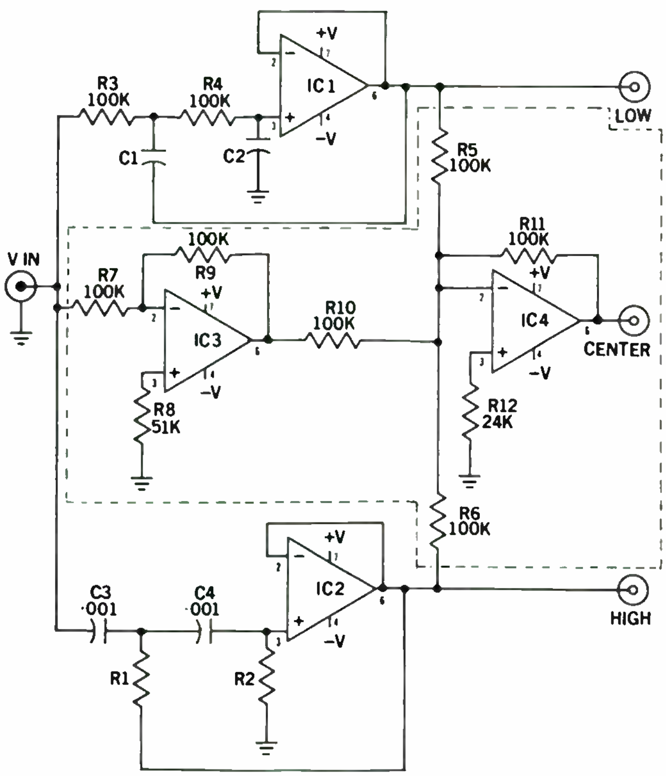


FIGURE 1: The original crossover circuit published in *TAA 2/72* can be improved by substituting the National LF351 or the TI TL071 for the old 741s.

TABLE A

BUTTERWORTH RESPONSE

	High Pass*		**Low Pass****	
Frequency (Hz)	**R1**	**R2**	**C1**	**C2**
60	1.88M (1.82M)	3.75M (3.65M)	0.038μF (0.047μF)	0.019μF (0.022μF)
120	938k (909k)	1.88M (1.82M)	0.019μF (0.022μF)	0.0094μF (0.01μF)
240	469k (475k)	938k (909k)	9400pF (0.01μF)	4690pF (0.0047μF)
480	234k (221k)	469k (475k)	4690pF (0.0047μF)	2340pF (0.0022μF)
960	117k (121k)	234k (221k)	2340pF (0.0022μF)	1170pF (0.001μF)
1920	58.6k (56.2k)	117k (121k)	1170pF (0.001μF)	586pF (500pF)
5k	22.5k (22.1k)	45k (43.2k)	450pF (470pF)	225pF (220pF)
10k	11.3k (11k)	22.5k (22.1k)	225pF (220pF)	112pF (120pF)

*for C3,4 = 0.001μF **for R3,4 = 100k
( ) = nearest practical E24 value

Equations

two-pole high-pass, C1 = C2 = 0.001μF; R2 = 2R1; R1 = $\frac{1.13 \times 10^{8}}{f}$

two-pole low-pass, R1 = R2 = 100k; C1 = 2C2; C2 = $\frac{1.13 \times 10^{-6}}{f}$

You can use the crossover in other ways, however. In *Fig. 3*, we cascade two DG-13R boards to produce a 12dB/octave, three-way crossover. Values for the two boards should be the two crossover points: for example, card 1 is set for the low frequency, card 2 is for the high frequency.

In *Fig. 4*, we adapt the DG-13R board to a two-way, 18dB/octave configuration. You need only choose one crossover point. The first card omits one resistor/capacitor pair in each active filter, reducing it to a 6dB/octave configuration, with lower distortion and phase problems.

Figure 5 shows the crossover's ultimate form. Four DG-13R boards and eight ICs produce a three-way, 18dB/octave slope to low, mid and high pass bands. You'll have to choose two frequencies, one for cards 1 and 2 and the other for cards 3 and 4. See *Fig. 3* for the 6dB/octave wiring instructions for cards 1 and 4.

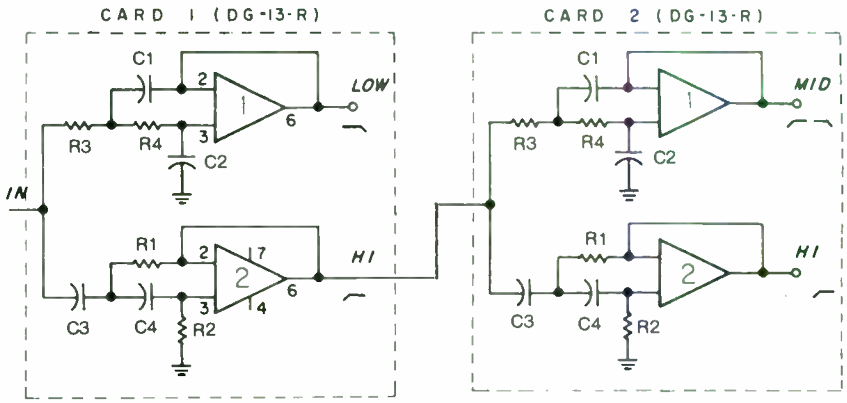


FIGURE 3: 12dB/octave, three-way, one-channel Butterworth.

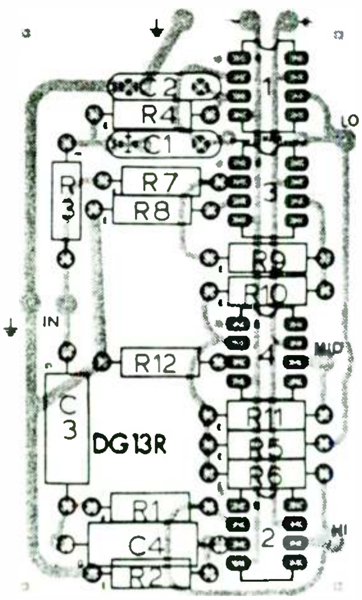


FIGURE 2: Layout guide for DG-13R board.

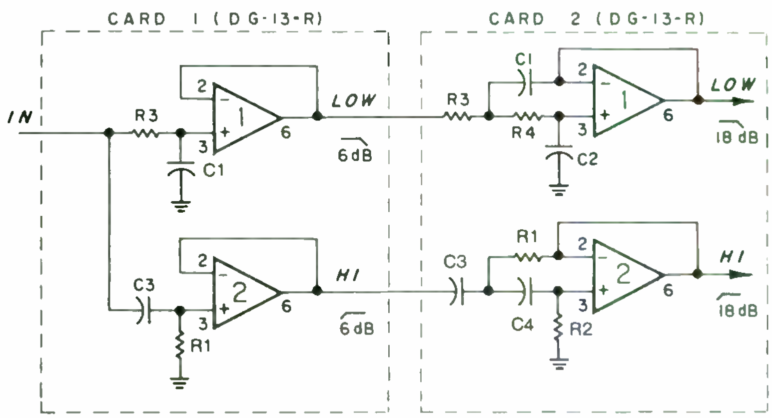


FIGURE 4: 18dB/octave, cascaded, two-way crossover, one channel. Use jumpers in place of R4 and C4. Omit R2, C1, R4 and C4. Frequency-determining elements are the same for both cards.

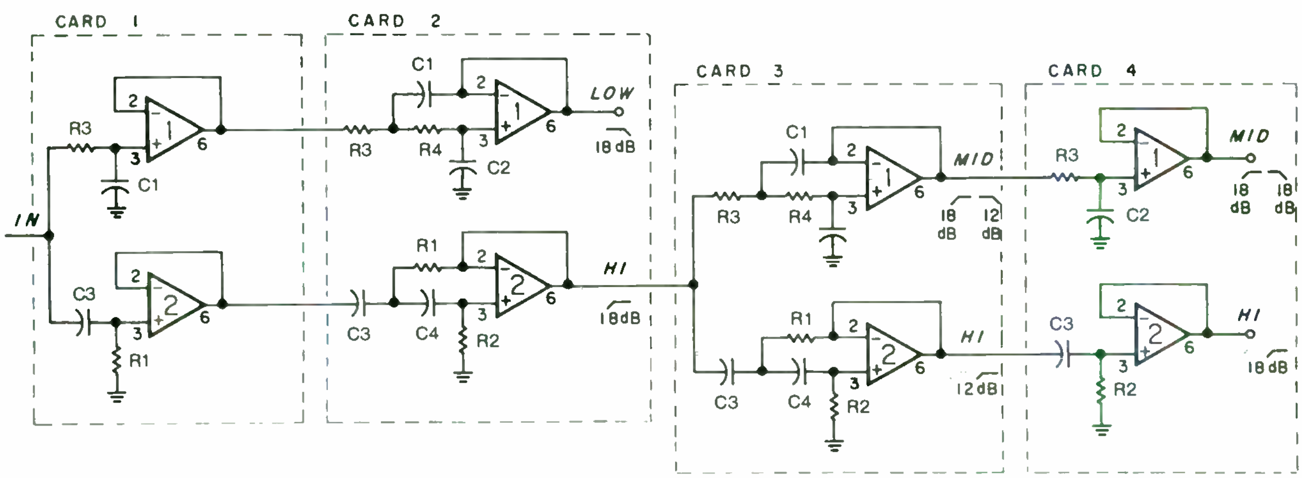


FIGURE 5: The three-way, 18dB/octave network utilizes four single-channel boards for one audio channel. For boards 1 and 4, omit R2, C1, R4 and C4. Install jumpers in place of R4 and C4. Choose one frequency for cards 1 and 2, the second frequency for cards 3 and 4.

MODIFIED STRATHEARN RIBBON SPEAKER

BY DON SPANGLER and MARK D. McKENZIE

Having heard improvements of anti-resonant filtering applied to the Jordan module (*TAA* 4/84, p. 54), we decided to undertake a similar project for the Strathearn SLC2 midrange ribbon driver.

Modifying the Strathearn driver involved about a year's worth of testing. During that time, we tested several damping materials in the diaphragm, the speaker's free air performance, various enclosure types, electrical direct drive versus drive through the stock toroid transformer, and numerous driver arrangements, including the addition of a super-tweeter. We have sorted through the results and believe we have come up with an excellent-sounding, top-end speaker that measures ±2dB from 350Hz to 20kHz.

Our tests included exhaustive pulse testing and Fast Fourier Transformation (FFT) analysis, along with comprehensive listening tests. We have found that pulse testing, when properly interpreted, correlates strongly with listening tests, much more so than any other electrical tests we have tried. Because of this, we would like to begin with an analysis of the pulse-testing process.

WHY USE PULSES? Three main types of test signal are commonly used to analyze speakers—the venerable sine wave, noise (white and pink) and pulses. These test signals provide varying amounts of data and require differing amounts of time to perform. Keep in mind that any test signal that does not begin and end during the time that the observer is logging data does not reveal information on the transient operation of the device under test (DUT).

A sine wave provides the least amount of transient information. *Figure 1* shows an oscilloscope presentation of a sine wave, while *Fig. 2* shows frequency versus magnitude. If an interval of limited duration is observed, you cannot determine the beginning or ending time of the sine wave. Within this observed interval, the changing magnitude values form a pattern that is constant through time. Although the data on the magnitude of the DUT output at the frequency of the sine-wave signal has very high resolution, you cannot predict how the DUT will perform when it is asked to start or stop producing the sine wave.

Pink and white noises are made up of random transients. They reveal more transient information than a sine wave, but because of their random nature, no transient sequence ever occurs in isolation. A transient episode always begins before the preceding episode ends and ends after the next one begins. *Figure 3* is an oscilloscope presentation of white noise, and *Fig. 4* shows frequency versus magnitude.

A pulse signal has one rising and one falling transient. The duration of the pulse is always less than one-half cycle of the highest frequency of interest. *Figure 5* shows an oscilloscope presentation, while *Fig. 6* shows frequency versus magnitude. Thus, with one easy test signal, you get data on the spectrum of the DUT, in which the transient data's contribution is balanced against that of the DUT's steady-state performance.

Sine-wave data or pulse data is often presented in a graphic form called a Bode plot. The magnitude of a number of frequencies is noted and marked on a graph. You can do this manually with pen and paper or automatically with a sweep generator and chart recorder. The two differ as follows: with pulse data, all frequencies of interest are present simultaneously in the magnitude/time plot, whereas with sine waves, the frequencies are discrete in time. To separate the frequencies making up a pulse, you must use a mathematical process called Fast Fourier Transformation (FFT). These transformations are based on the idea that complex waveforms are made up of a number of simpler waveforms. In this particular application, the complex waveform is a pulse, and the simple waveforms are sine waves.

In most circumstances, the FFT is

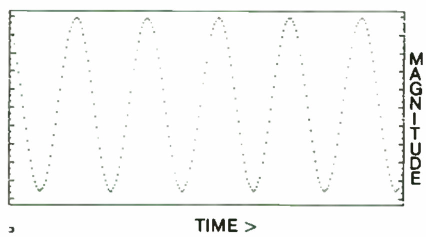


FIGURE 1: Oscilloscope presentation of a sine wave.

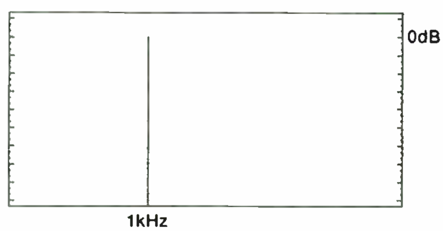


FIGURE 2: Frequency versus magnitude of a sine wave.

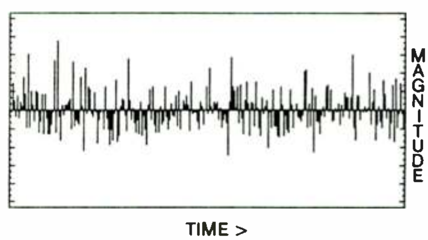


FIGURE 3: Oscilloscope presentation of white noise.

transparent, and the mechanics of the transformation are not important to the end user. It is, however, beneficial to have some knowledge of how the shape of the pulse information determines the frequency response that the FFT produces. To help you understand the process better, I will present three examples of pulse responses, the math of the transformations and the resultant frequency response of the basic building blocks of all complex transient behavior. See *Table 1* for the names and sources of the equipment used for data acquisition and analysis.

SAMPLE PULSE RESPONSES. Suppose you wish to determine the magnitude of a perfect pulse at 1kHz. To do so, you would use a 1kHz sine wave and a 1kHz cosine wave (which is a sine wave shifted 90 degrees so that the points of maximum magnitude and the zero crossings are reversed). Align the three signals so that every point in time at which the perfect pulse is sampled matches an equivalent point on the sine and cosine waves *(Fig. 7)*. Then multiply every point in time of the perfect pulse sample window by the values of the sine and cosine at the matching degree positions. In the perfect pulse, only one time unit has an associated magnitude value; all other time units are zero (a good reason for transient accurate speakers, as the FFT becomes simple). Add all the sine products together and square the sum. Do the same with the cosine values. Then add the two squared values together, and the magnitude of the perfect pulse at 1kHz will be the square root of the sum.

For example, if the pulse magnitude is 1 and corresponds to degree 10 of the sine, the sine product is 1 times 0.174, or 0.174, and the cosine value is 1 times 0.985, or 0.985. All other time units are zero, so the total for the sines is 0.174 and for the cosines 0.985. If you square the two numbers, sine equals 0.03, and cosine equals 0.97. Add the two products together (0.03 + 0.97 = 1) and take the square root of the sum for a final magnitude value of 1. This is the equivalent value of a 1kHz sine-wave building block of the pulse. Repeat the process for all other frequencies of interest. In a perfect pulse, all other frequencies have the same magnitude. This final magnitude value remains the same with any sine or cosine degree positions used.

Should the input pulse not be perfect *(Fig. 8)*, with a value of 1 at one sampling point and a value of 0.5 at the next, the FFT would change in the following ways. The degree 10 values would stay the same—i.e., the sine product would be 0.174 and the cosine product 0.985. But the degree 20 values would change from zero to a sine product of 0.171 and a cosine product of 0.47. Add the sine values for a total of 0.345 and the cosine values for a total of 1.455. If you square the two numbers, sine equals 0.119, and cosine equals 2.117. Add the two products together (0.119 + 2.117 = 2.236) and take the square root of the sum for a final magnitude value of 1.5.

Should the imperfect pulse's frequency of interest be ten times lower, the degrees with which the sample would coincide would become degrees 10 and 11, for which the summed squared sine equals 0.072, and the summed squared cosine equals 2.179. Add these two together, then take the square root for a 100Hz magnitude value of 1.5.

Should the frequency of interest be ten times higher than 1kHz, the coinciding sine and cosine degrees would be 10 and 200. For these sample points, the summed squared sine equals zero, and the summed squared cosine equals 0.26. Add the products together and take the square root for a magnitude value of 0.5. If you convert these values to decibel equivalents, with 1kHz as the reference frequency, the frequency response of the imperfect pulse will be as follows: 100Hz, 0dB; 1kHz, 0dB; 10kHz, –9.5dB.

Figure 9 shows the frequency response of an imperfect pulse. You can see the response deviations and pulse alterations for a driver that is limited in reproduction of the higher frequencies. Should the driver be limited in low-frequency response, the pulse response might look something like *Fig. 10*. This is the same as the first imperfect pulse, except now the second sampled magnitude value is negative.

For the 1kHz analysis, degrees 10

TABLE 1

EQUIPMENT USED

1. Apple II, any series
2. IQS 401, any series
   (IQS, Inc, 2610 N. Greenbrier, Santa Anna, CA 92706)
3. any calibrated microphone
4. any suitable amplifier

MATERIALS LIST

Material	Supplier
Strathearn midrange ribbon speaker, SLC2	Audio Concepts 1631 Caledonia Street LaCrosse, WI 54601
Capacitors for amplifier, crossover and corrective prefilters: WIMA MKPIO series (polypropylene)	GSI Audio 578 Neperhan Avenue Yonkers, NY 10701 WonderCaps (old style) also suitable
Corrective filter coils	(see note)*
Resistors for corrective filter, 10Ω 2Ω	Radio Shack #271-132 Two Radio Shack #271-131 (two 1Ω units in series)
Foam inserts for driver	Frost King Air Conditioner Part #F1524, local hardware store
Hot glue, Mortite, Liquid Nail	Local hardware store
Particle board	Local lumber dealer
Bituminous felt	Local building supplier
Sonex	Illbruck 3800 Washington Ave. N. Minneapolis, MN 55412 or local audio dealer

* For the coils, purchase two Radio Shack #278-1302 devices. Hookup wire is 18 gauge solid. Each contains three rolls of wire on suitable plastic forms for the project. Unwind all of the wire. On two of the forms, tightly wind 55 turns of wire. These will serve as the 109μH coils. On the other four coils, wind 53 turns of wire. Two should be wired in series and will serve as the 207μH coils. Do not mount the coils with a magnetic material. Hot glue is suggested.

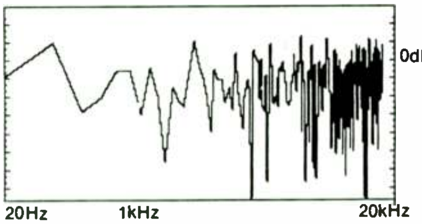


FIGURE 4: Frequency versus magnitude of white noise.

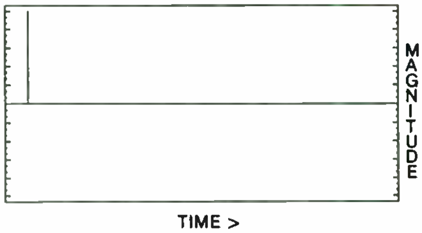


FIGURE 5: Oscilloscope presentation of a pulse signal.

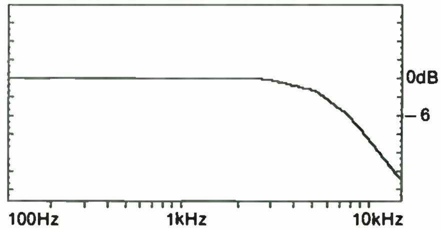


FIGURE 9: Frequency of the imperfect pulse in Fig. 8.

and 20, the summed squared sine value is zero, and the summed squared cosine value is 0.265. If you add the products together and take the square root, you come up with a final magnitude value of 0.5.

For the 100Hz analysis, degrees 10 and 11, the summed squared sine value is 0.006, and the summed squared cosine value is 0.244. If you add the products together for a value of 0.25 and take the square root, the final magnitude value is 0.5.

For a 10kHz analysis, degrees 10 and 200, the summed squared sine value is 0.119, and the summed squared cosine value is 2.117. Add the products together for a value of 2.236 and take the square root to produce a final magnitude value of 1.5. Using 1kHz as the reference frequency, the logarithmic converted magnitude values are 100Hz, 0dB; 1kHz, 0dB; 10kHz, +9.5dB. *Figure 11* is a frequency response graph of the second imperfect pulse.

Before moving on to the modification, we should mention a special type of sine-wave testing using tone bursts. A tone burst allows you to observe test beginning and ending, but obscures the degree to which the transient behavior alters the summed output. Because of the defined rate of magnitude change, a sine wave has only one frequency. During the start of the sine-wave burst, the transient performance of the DUT is excited at the sine-wave frequency and at every frequency below the fundamental. Nothing above the fundamental is excited because the duration of the signal has exceeded one-half wavelength of the fundamental.

Transient information is provided during a tone burst only during the first and last quarters of the sine-wave pulse. The sine wave that occurs between the first and last quarters is not transient but steady state. The number of cycles for which the steady-state portion of the fundamental lasts will

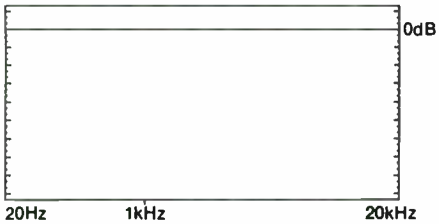


FIGURE 6: Frequency versus magnitude of a pulse signal.

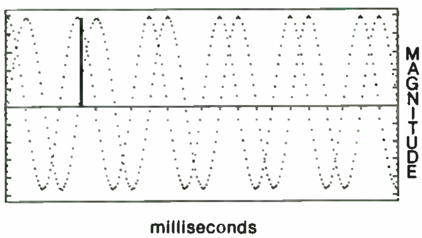


FIGURE 7: To determine the magnitude of a perfect pulse, use a sine wave and a cosine wave. Align the three signals so that the perfect pulse matches an equivalent point on the sine and cosine waves.

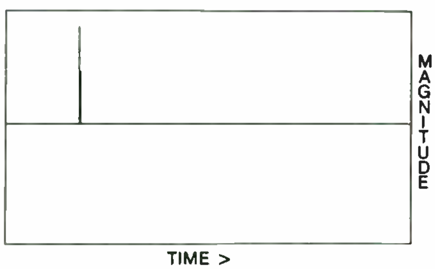


FIGURE 8: An imperfect input pulse.

reduce the contribution of the transient portion in a summed analysis. If one complete cycle of a sine wave is divided into four quarters, then for a tone burst that lasts for seven cycles, the transient portion would come from the first and last quarters, for a summed portion of 2. The steady state from the remainder of the tone burst results in a summed portion of 26. An FFT of this tone burst would show a central spike representing the fundamental and a slight spread in frequencies about 15dB lower. A 3dB decrease occurs for every power of 2 increase of the fundamental over the number of transient portions.

PLANNING THE MOD. When planning a modification, you should have some idea of what you would like the speaker to do. In the case of an ideal speaker, you would want it to reproduce a pulse perfectly. It should jump precisely on a transient, not compress the peak magnitude and not overshoot or ring on the decay.

It is a good idea to know how the speaker under consideration—in this case, the Strathearn ribbon speaker—fails to meet these ideal criteria. You can determine this by testing the Strathearn in free air and observing the results (see *Figs. 12* and *13*). *Figure 12* shows the magnitude versus time (or pulse response) of the speaker. The Strathearn exhibits very good response in the onset of the pulse, as the energy is tight and very closely bundled in time. The decay of the pulse is not as good as you might wish. The pulse takes too long to drop off, and there is some evidence of a mechanical resonance causing ringing in the decay. Also, the main body of the energy being delivered is slightly spread over time.

In *Fig. 13,* the FFT-derived frequency response of the pulse, the Strathearn exhibits two elevated plateaus separated by a suck-out in the upper midrange/lower treble. The low-frequency plateau starts at about 500Hz and extends to about 2kHz. It is very smooth and even. The higher-frequency plateau centers on the 10 to and subwoofers, so I really have no quarrel with the way it was presented.

To get back to Mr. Weems' tapered pipes, the taper he uses is the reverse of the type generally used in TLs. It is actually more nearly a horn than a TL, whose function is to guide and lose the speakers' back radiation. I am reminded of the late Stewart Hegeman's split, conical slot loaded horns, made for Eico, Lowther, and for his own company. Of course they were different, but there may be a kinship. I have wondered how they might have sounded stuffed.

Just out of curiosity I would like Mr. Weems to place his line in a horizontal position to see whether he notices an improvement in the smoothness and solidity of the sound.

I don't understand his statement concerning the use of thinner material for undamped pipes (or maybe I just don't want to). I also believe his equation for optimum speaker placement is invalidated by stuffing only one side of his pipe. The stuffed side will appear longer. This is not a criticism, only a suggestion. Perhaps with a little more work he just might get out of the doghouse and produce a speaker system that is dog-gone good. And who knows, if he can find a long-haired cat he could even have dynamic damping! But leave the rabbits to Bruce Edgar.

John Cockroft
Mountain View, CA 94041

Mr. Weems replies:

Dynamic damping? Such a shameless suggestion must give "paws" lest the SPCA become involved. But I must admire such an original concept. It would surely catapult speaker design to a higher category of science. As to long-haired breeds, try Persian, it should work as well as rug padding.

In regard to your suggestion of putting tapered pipes in a horizontal position, 27" high, that would, of course, lower the driver and elevate the port. Unfortunately, having spent my time and energy in speaker experiments instead of more lucrative goals, my old house lacks enough wall space to do that without moving large pieces of furniture. And that would alter the acoustic environment. Room position always seems to be a major influence.

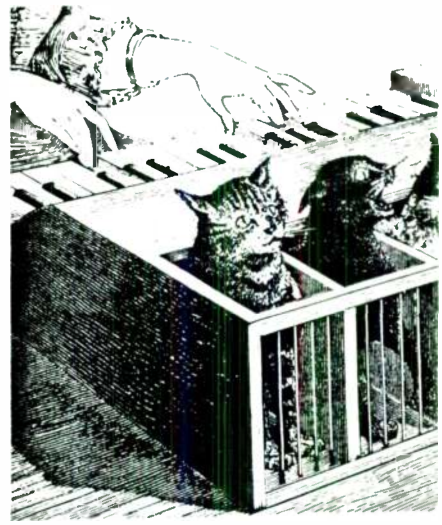
I agree with you on the two Radio Shack drivers, 40-1022 and 40-1376. They are better than most of the larger and more expensive models.

I think the tapered pipe owes something to Voigt's early designs. I'm not familiar with the internal anatomy of Hegeman's horns, but I remember the speakers.

Both sides of the pipe should be stuffed. The stuffing in the throat was not shown in the drawings, but it is specified in the article.

As to undamped pipes, I don't understand anything at all about them. Or maybe I just don't believe them. They were popular in England a generation ago and authors were adamant about using thin materials.

I still have my pipes for the 6½" Peerless woofer but, sad to say, I did not follow my own advice about screening the ports to keep critters out. The small pipe, for the 4½" woofer, became a mouse house. Even worse, a morgue. The mouse mother perished, perhaps by a heart attack induced by the high SPL. The death of the entire family followed but I didn't find the source of the odor until the pipe was saturated with it. The enclosure had to be burned. A Viking's funeral for the little fellows.



WARRANTY WOE

In Gary Galo's "Transmission Line Loudspeakers, Part II: Application" (*SB*, 2/82) he states, "Solder leads (*never* use slip-on speaker connectors!)"

In Audio Concepts instructions for its Quartz kit (which uses the excellent Dynaudio drivers) the builder is warned to use the slip-on connectors provided with the kit and *not* solder the speaker connections. Further, they state the warranty is void if the leads are soldered.

What gives? These are the only two instances I can recall addressing the question and they definitely are not in agreement.

I always soldered the connections until being warned against the practice in the Audio Concepts instructions. Help!

Dave Schneider
Marion, IA 52302

Audio Concepts is talking about warranty. Galo is talking about sound. Mr. Galo's advise is consonant with Walt Jung's advice on this matter. Audio Concepts, rightly, is being conservative in advising against a practice that, also rightly, voids a warranty. Of course, careless or inept soldering could damage a driver's leads where they attach to the cone. Galo's advice assumes prudence and care, as well as a desire for good sound that is more important than warranty cards.—Ed.

ment. More on this in a moment.

Your third modification raises questions I dealt with when I evaluated the unit. I agree with you. I don't like putting unnecessary circuitry into the signal path either—particularly for the high frequencies. My initial inclination was to connect it as you described, so I phoned Audio Control and questioned them about it. They said the equalizer section of the unit was designed to receive tape output levels only, which are lower in level than a preamp's main output, and also independent of the volume control setting. Running the preamp's output into the equalizer section via the crossover raises the equalizer section's distortion level because of the slight overloading that may occur at high volume settings.

Your letter prompted me to investigate this further. My measurements indicate you should have no problem connecting the unit as you describe. The equalizer's clipping level is between 7.6 and 7.8V, measured at 20Hz, 100Hz and 1kHZ (the highest frequency likely to be passed to the equalizer from the crossover; most crossover frequencies will be well below this). Harmonic distortion just below clipping is .016% at 100Hz and .045% at 20Hz. Apparently the equalizer section is not prone to the overload problems Audio Control believes. Its performance seems identical to the crossover section. On this evidence, I recommend connecting the unit as you describe.

The supply voltages (which are ±13V in my sample) limit the clipping level of the equalizer and crossover. The maximum ratings for TL075 op amps are ±18V. When you replace the unregulated supply in the Richter Scale with a regulated supply of ±15 to ±16.5V, you raise the clipping and performance levels.

The space on the right hand side of the circuit board in the Richter Scale chassis is large enough to add a regulated supply. When you remove the power transformer, you free a space measuring approximately 3" x 7" x 2¾". You cannot use Audio Control's transformer since the secondary voltages aren't high enough. Mount the new power transformer in an external box, then feed the secondary AC voltages to the rectifiers and regulators mounted in the free chassis space. The Gately supply circuit board (Old Colony KF-3) will fit here.

If you want more brute force regulation as in the Jung PAT-5/WJ1A preamp supply (TAA, 1/78, 3/79), the power transformer, rectifier bridges and input filter caps can be mounted in an outboard chassis. A parallel set of input filter caps, the regulation circuitry and the output filter caps can then be mounted in the free space in the Richter Scale chassis. With a judicious layout everything should fit. I did this in my own electronic crossover.

The Audio Control power supply schematic has two errors. The 22Ω resistors between the 470µF filter capacitors are not shown. The schematic shows the capacitors connected in parallel. The same error exists in both the positive and negative supply sections. The resistors labeled 22Ω, connecting the 470µF output filter caps to the zener diodes, should be labeled 2.2kΩ. Again, the same error is repeated for the positive and negative supplies.

Your procedure for setting the woofer level should produce exactly the same results as my Procedure B. I agree with your assessment of the worth of the Richter Scale; its value as a piece of test equipment alone justifies the expense.

PIPES, PETS & PORTS

About your TTT Test—I've been thinking of adding an equalizing circuit to reduce the roll-off.

I was quite interested in David Weems' cathouse—I mean tapered pipes, (but the man does seem to have a way with animals). I was interested because what he created must be a relative to a little system I designed and built last November. Undoubtedly I was anticipating Thanksgiving when the idea occurred to me to make a stuffed version of the Bose Acoustic Wave enclosure. I had inadvertently thrown away the article describing it, but as I recall it was a six foot pipe, open at both ends, with the speaker placed a third of the way from one end.

I decided to stuff the pipe to make it shorter and chose three feet. I placed the 4" inch speaker 12" in from one end. The cross section was about 25 square inches. I settled on a density of about .6 lbs/cu. foot.

Although I have spent very little time with this system, I hear natural sound similar to other Transmission Lines (TLs) I have built.

I made the pipe in a straight line and placed it in a horizontal position on a 27" tall stool with the line about ¼ to ½" from the back wall. The speaker is facing upward, as suggested by Roy Allison (a man who knows what he's talking about).

Mr. Weems and I both chose the same drivers for our projects, which appeared in the same issue of *SB*, 2/87. This should invite some comparisons. I have used these little units in many projects. Of all the Radio Shack drivers, the 40-1022 seems the most consistent, is also inexpensive and sounds good in a variety of applications. On a cost-for-sound basis they are high on my list.

The 40-1376 tweeter listed as ¾", is really a ⅜" tweeter. Its dispersion is exceptional, and in most rooms needs a diffraction ring to limit the dispersion and thus cut down on destructive room reflections. This unit is the closest thing I know of to a point source. It seems to be a silver colored version of the Audax Tw74.

About my project, the "Demonstrator," *SB*, 2/87 it isn't "a tiny double woofer" but a full-range system. This concept was created to help those trying to create woofers

ment. More on this in a moment.

Your third modification raises questions I dealt with when I evaluated the unit. I agree with you. I don't like putting unnecessary circuitry into the signal path either—particularly for the high frequencies. My initial inclination was to connect it as you described, so I phoned Audio Control and questioned them about it. They said the equalizer section of the unit was designed to receive tape output levels only, which are lower in level than a preamp's main output, and also independent of the volume control setting. Running the preamp's output into the equalizer section via the crossover raises the equalizer section's distortion level because of the slight overloading that may occur at high volume settings.

Your letter prompted me to investigate this further. My measurements indicate you should have no problem connecting the unit as you describe. The equalizer's clipping level is between 7.6 and 7.8V, measured at 20Hz, 100Hz and 1kHZ (the highest frequency likely to be passed to the equalizer from the crossover; most crossover frequencies will be well below this). Harmonic distortion just below clipping is .016% at 100Hz and .045% at 20Hz. Apparently the equalizer section is not prone to the overload problems Audio Control believes. Its performance seems identical to the crossover section. On this evidence, I recommend connecting the unit as you describe.

The supply voltages (which are ±13V in my sample) limit the clipping level of the equalizer and crossover. The maximum ratings for TL075 op amps are ±18V. When you replace the unregulated supply in the Richter Scale with a regulated supply of ±15 to ±16.5V, you raise the clipping and performance levels.

The space on the right hand side of the circuit board in the Richter Scale chassis is large enough to add a regulated supply. When you remove the power transformer, you free a space measuring approximately 3″ x 7″ x 2¾″. You cannot use Audio Control's transformer since the secondary voltages aren't high enough. Mount the new power transformer in an external box, then feed the secondary AC voltages to the rectifiers and regulators mounted in the free chassis space. The Gately supply circuit board (Old Colony KF-3) will fit here.

If you want more brute force regulation as in the Jung PAT-5/WJ1A preamp supply (TAA, 1/78, 3/79), the power transformer, rectifier bridges and input filter caps can be mounted in an outboard chassis. A parallel set of input filter caps, the regulation circuitry and the output filter caps can then be mounted in the free space in the Richter Scale chassis. With a judicious layout everything should fit. I did this in my own electronic crossover.

The Audio Control power supply schematic has two errors. The 22Ω resistors between the 470µF filter capacitors are not shown. The schematic shows the capacitors connected in parallel. The same error exists in both the positive and negative supply sections. The resistors labeled 22Ω, connecting the 470µF output filter caps to the zener diodes, should be labeled 2.2kΩ. Again, the same error is repeated for the positive and negative supplies.

Your procedure for setting the woofer level should produce exactly the same results as my Procedure B. I agree with your assessment of the worth of the Richter Scale; its value as a piece of test equipment alone justifies the expense.

PIPES, PETS & PORTS

About your TTT Test—I've been thinking of adding an equalizing circuit to reduce the roll-off.

I was quite interested in David Weems' cathouse—I mean tapered pipes, (but the man does seem to have a way with animals). I was interested because what he created must be a relative to a little system I designed and built last November. Undoubtedly I was anticipating Thanksgiving when the idea occurred to me to make a stuffed version of the Bose Acoustic Wave enclosure. I had inadvertently thrown away the article describing it, but as I recall it was a six foot pipe, open at both ends, with the speaker placed a third of the way from one end.

I decided to stuff the pipe to make it shorter and chose three feet. I placed the 4″ inch speaker 12″ in from one end. The cross section was about 25 square inches. I settled on a density of about .6 lbs/cu. foot.

Although I have spent very little time with this system, I hear natural sound similar to other Transmission Lines (TLs) I have built.

I made the pipe in a straight line and placed it in a horizontal position on a 27″ tall stool with the line about ¼ to ½″ from the back wall. The speaker is facing upward, as suggested by Roy Allison (a man who knows what he's talking about).

Mr. Weems and I both chose the same drivers for our projects, which appeared in the same issue of *SB*, 2/87. This should invite some comparisons. I have used these little units in many projects. Of all the Radio Shack drivers, the 40-1022 seems the most consistent, is also inexpensive and sounds good in a variety of applications. On a cost-for-sound basis they are high on my list.

The 40-1376 tweeter listed as ¾″, is really a ⅜″ tweeter. Its dispersion is exceptional, and in most rooms needs a diffraction ring to limit the dispersion and thus cut down on destructive room reflections. This unit is the closest thing I know of to a point source. It seems to be a silver colored version of the Audax Tw74.

About my project, the "Demonstrator," *SB*, 2/87 it isn't "a tiny double woofer" but a full-range system. This concept was created to help those trying to create woofers

and subwoofers, so I really have no quarrel with the way it was presented.

To get back to Mr. Weems' tapered pipes, the taper he uses is the reverse of the type generally used in TLs. It is actually more nearly a horn than a TL, whose function is to guide and lose the speakers' back radiation. I am reminded of the late Stewart Hegeman's split, conical slot loaded horns, made for Eico, Lowther, and for his own company. Of course they were different, but there may be a kinship. I have wondered how they might have sounded stuffed.

Just out of curiosity I would like Mr. Weems to place his line in a horizontal position to see whether he notices an improvement in the smoothness and solidity of the sound.

I don't understand his statement concerning the use of thinner material for undamped pipes (or maybe I just don't want to). I also believe his equation for optimum speaker placement is invalidated by stuffing only one side of his pipe. The stuffed side will appear longer. This is not a criticism, only a suggestion. Perhaps with a little more work he just might get out of the doghouse and produce a speaker system that is dog-gone good. And who knows, if he can find a long-haired cat he could even have dynamic damping! But leave the rabbits to Bruce Edgar.

John Cockroft
Mountain View, CA 94041

Mr. Weems replies:

Dynamic damping? Such a shameless suggestion must give "paws" lest the SPCA become involved. But I must admire such an original concept. It would surely catapult speaker design to a higher category of science. As to long-haired breeds, try Persian, it should work as well as rug padding.

In regard to your suggestion of putting tapered pipes in a horizontal position, 27" high, that would, of course, lower the driver and elevate the port. Unfortunately, having spent my time and energy in speaker experiments instead of more lucrative goals, my old house lacks enough wall space to do that without moving large pieces of furniture. And that would alter the acoustic environment. Room position always seems to be a major influence.

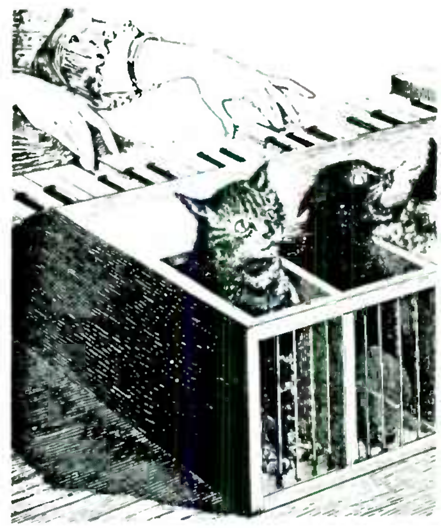
I agree with you on the two Radio Shack drivers, 40-1022 and 40-1376. They are better than most of the larger and more expensive models.

I think the tapered pipe owes something to Voigt's early designs. I'm not familiar with the internal anatomy of Hegeman's horns, but I remember the speakers.

Both sides of the pipe should be stuffed. The stuffing in the throat was not shown in the drawings, but it is specified in the article.

As to undamped pipes, I don't understand anything at all about them. Or maybe I just don't believe them. They were popular in England a generation ago and authors were adamant about using thin materials.

I still have my pipes for the 6½" Peerless woofer but, sad to say, I did not follow my own advice about screening the ports to keep critters out. The small pipe, for the 4½" woofer, became a mouse house. Even worse, a morgue. The mouse mother perished, perhaps by a heart attack induced by the high SPL. The death of the entire family followed but I didn't find the source of the odor until the pipe was saturated with it. The enclosure had to be burned. A Viking's funeral for the little fellows.



WARRANTY WOE

In Gary Galo's "Transmission Line Loudspeakers, Part II: Application" (*SB*, 2/82) he states, "Solder leads (*never* use slip-on speaker connectors!)"

In Audio Concepts instructions for its Quartz kit (which uses the excellent Dynaudio drivers) the builder is warned to use the slip-on connectors provided with the kit and *not* solder the speaker connections. Further, they state the warranty is void if the leads are soldered.

What gives? These are the only two instances I can recall addressing the question and they definitely are not in agreement.

I always soldered the connections until being warned against the practice in the Audio Concepts instructions. Help!

Dave Schneider
Marion, IA 52302

Audio Concepts is talking about warranty. Galo is talking about sound. Mr. Galo's advise is consonant with Walt Jung's advice on this matter. Audio Concepts, rightly, is being conservative in advising against a practice that, also rightly, voids a warranty. Of course, careless or inept soldering could damage a driver's leads where they attach to the cone. Galo's advice assumes prudence and care, as well as a desire for good sound that is more important than warranty cards.—Ed.

FIGURE 10: If a driver is limited in low-frequency response, the imperfect pulse response might look something like this.

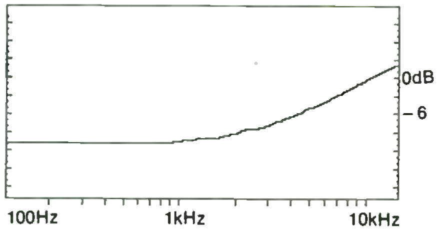


FIGURE 11: Frequency response of the imperfect pulse in Fig. 10.

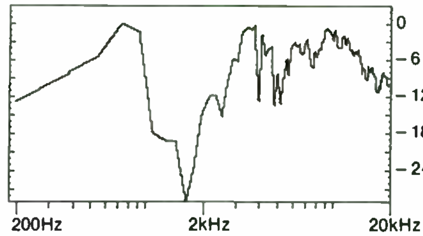


FIGURE 15: The FFT-derived frequency response of the Strathearn with the wallboard and fiberglass dampers removed.

12kHz range and has a greater magnitude and a higher Q than the lower-frequency plateau.

The next task is finding out what is causing these undesirable deviations. First, eliminate as many variables affecting the performance as possible. With this driver, that entails removing all the material (wallboard and fiberglass dampers) obscuring the diaphragm and retesting the unit. See *Figs. 14* and *15* for the results of the new tests.

Because this driver is a dipole, the slightly late contribution of the rear-wave output enhances the lower ranges. This produces the 500Hz to 2kHz plateau. By separating the front and rear outputs, you can eliminate this plateau. The 10 to 12kHz peak is caused by the metal ribbon's resonating on the diaphragm material. The drop in the response above this frequency is a result of the driver's inertial mass exceeding the driver's motor force. The very sharp dip at 1kHz is caused by the resonance of the speaker diaphragm itself. The metal foil in the center of this stretched diaphragm causes the two halves of the diaphragm to resonate at slightly different frequencies. Even with the wallboard inserts that normally cover most of the diaphragm, the leftover output from these resonances causes the suck-out between the two peaks in the frequency response previously measured.

Now what can be done about these deviations from the ideal? Because the plateau from 500Hz to 2kHz is caused by leakage from the rear radiation of the driver, placing the driver in a baffle reduces this problem by about 3dB (see details below). Thin sheets of foam beneath the cut-out pieces of wallboard can better absorb the resonance energy in the diaphragm because the foam's mechanical im-

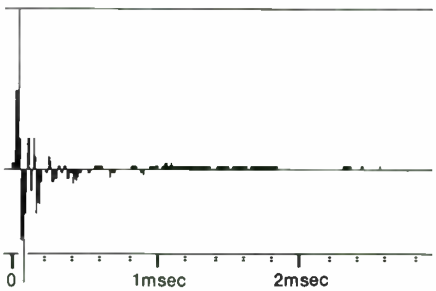


FIGURE 12: Magnitude versus time (or pulse response) of the Strathearn in free air.

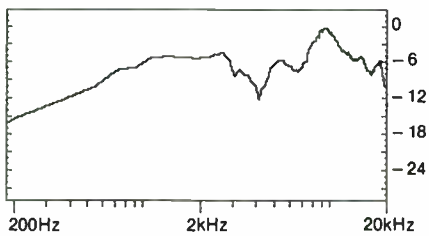


FIGURE 13: The FFT-derived frequency response of the Strathearn in free air.

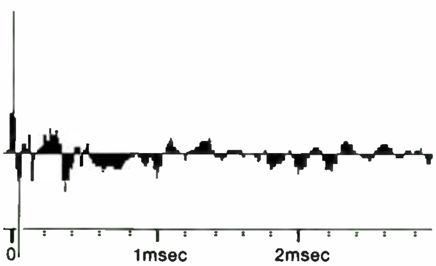


FIGURE 14: Magnitude versus time (or pulse response) of the Strathearn with the wallboard and fiberglass dampers removed.

pedance is closer to that of the diaphragm than is the wallboard's. This also produces an approximately 3dB decrease in the magnitude of the suck-out. To reduce the peak caused by the metal foil resonating, you can insert two resonant networks in the line between the speaker and the transformer. This removes enough of the input to the speaker at the frequencies of interest to balance the increased output from the driver at 10 to 12kHz.

These three modifications result in a ±2dB magnitude deviation of the output from 350Hz to 20kHz. They also improve the transient behavior of the Strathearn. The output decays to zero sooner, and the first negative-going overshoot is reduced *(Figs. 16* and *17). Figure 18* shows the drive voltage after the prefilter *(Fig. 19),* representing the degree of correction applied to this driver. *Figure 20* shows the impedance of the driver and the prefilter, which varies from a high of 14Ω to a low of 7.1Ω.

The prefilter at the input of the transformer is made of two resonant networks cascaded together. One is a series network and the other a parallel resonant network. The resistors should be at least 5W, and the capacitors should be high-quality, low-ESR, solid-state types. If a composite capacitor is needed to achieve the specified values, always parallel the capacitors, never wire them in series. The parts' tolerance should be 2 percent or less. The transformers supplied with the drivers, when hooked up according to the polarity markings on the transformers, are inverting.

MODIFYING THE UNIT. To begin the physical modification of the driver, remove all the damping panels. To remove the wallboard pieces and rear damping panels, take out the eight bolts along the outside edge of the

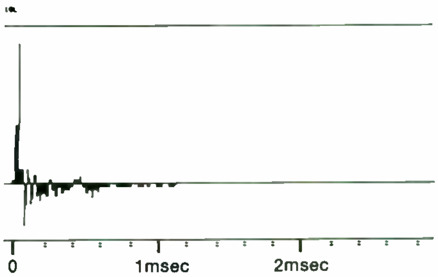


FIGURE 16: Magnitude versus time (or pulse response) of the fully modified Strathearn (see text).

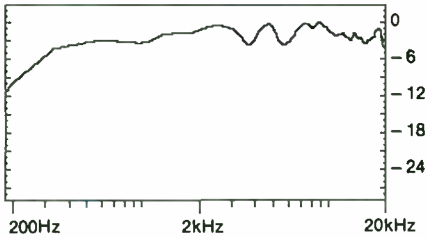


FIGURE 17: The FFT-derived frequency response of the fully modified Strathearn (see text).

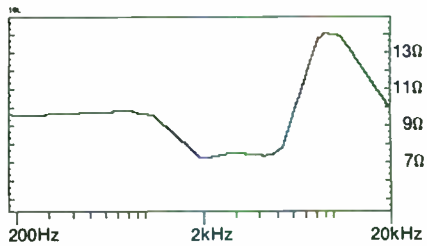


FIGURE 20: Impedance of the driver and the prefilter.

driver. Lift off the rear damping panels, then pry up the wallboard with a small screwdriver. Always pry up the pieces at the outside edges of the driver. The wallboard is not glued in, but glue from the diaphragm occasionally leaks behind the wallboard. Lift the wallboard as straight up and away from the frame as possible. Do not try to "rock out" the pieces, as the magnets are quite fragile, and it is easy to knock them loose. *Photo 1* shows the front of the drive and the front wallboards, while *Photo 2* shows the back of the drive and the back wallboards and damping panels.

The foam inserts are made of ⅛-inch-thick, open-cell polyurethane foam, often used as a light packing material or as a filter for window air conditioners. Precut the foam to fit the area of the exposed diaphragm between the inside of the driver frame and the first magnet bar. The cut need not be that precise, but at least 90 percent of this area should be covered. After inserting the foam, return the wallboard cutouts to their proper positions.

Do not bolt the wallboard in place. Instead, lightly press the pieces together, front to back, until the top outside edges of the wallboard are even with the top of the magnet assemblies. Apply as even a tension as possible. The wallboard should be flat or parallel in relation to the front of the driver. Make sure that the diaphragm is centered evenly between the front and back magnets. Once the pieces are aligned, run a bead of hot melt glue along the edges of the magnet structures and the wallboard and along the magnet support structures at the top and bottom of the wallboard. Hold the wallboard until the hot melt glue hardens, usually about two minutes.

ENCLOSURE CONSIDERATIONS. As we mentioned before, testing in-

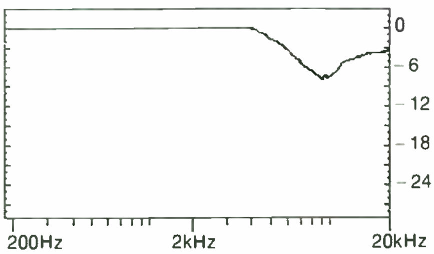


FIGURE 18: The drive voltage after the prefilter in Fig. 19 was inserted in the speaker.

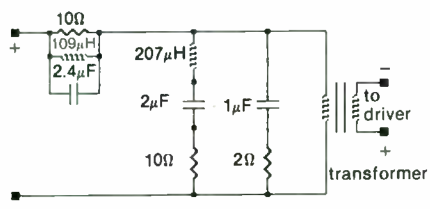


FIGURE 19: This prefilter, inserted at the input of the transformer, is made of two resonance networks cascaded together.

dicated that an enclosure or baffle is necessary for flat response. We chose an open-back enclosure to avoid the inherent boxiness of a closed-box design. The dimensions are not critical. Even a flat baffle will be suitable, as long as a sound path of about 2 feet is maintained between the front and back of the driver.

Figure 21 shows the dimensions of our suggested enclosure. The back is open, and a 2-foot path exists between the front and back of the driver. To avoid the problems of a dipole (mentioned above), you can attenuate the rearwave by inserting three layers of Sonex, separated slightly, behind the Strathearn driver. There is room in the bottom of the cabinet for the corrective prefilter network *(Fig. 19)* and the toroidal transformer.

The Strathearn driver is mounted on two blocks of wood on the inside, top and bottom. These blocks are inset enough so that the driver is flush with the front of the enclosure. Mortite is used to seal the edge of the driver to the enclosure. If a grille cloth is required, you can cut it to the front dimension and hold it in place by pressing against the Mortite.

We used ½-inch particle board for the enclosure and lined it with ½-inch bituminous felt. The felt is the same material used in expansion joints between concrete sections and is available from a building supplier. We then used Liquid Nail to cement the felt to the particle board.

LISTENING TESTS. The final test is listening to your system. Experience has shown that speakers are sometimes impressive at first listen, but tend to become bothersome with time. A speaker system's ability to satisfy the extended-listening test is a good indication of its quality. The modified Strathearns passed this test with flying colors.

Massed violins are very difficult to reproduce with fidelity. The sound should be silky, with only a slight edge, especially in the instruments' low and middle frequency range. This requires that the speaker system have flat frequency and controlled phase response. Many systems reproduce massed violins as metallic and shrill, but the modified Strathearns reproduce massed strings that are very close to the live concert experience.

The Strathearns also reproduce transients faithfully. Cymbals and triangles sound real, and there is no apparent dulling of transients or ringing trig-

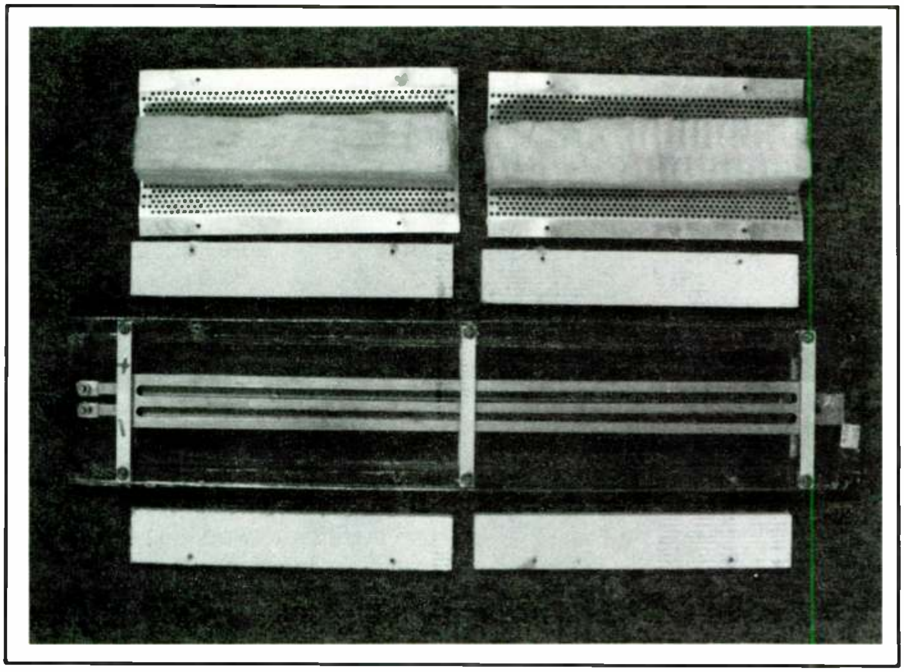


PHOTO 1: Front of the drive and front wallboard pieces.

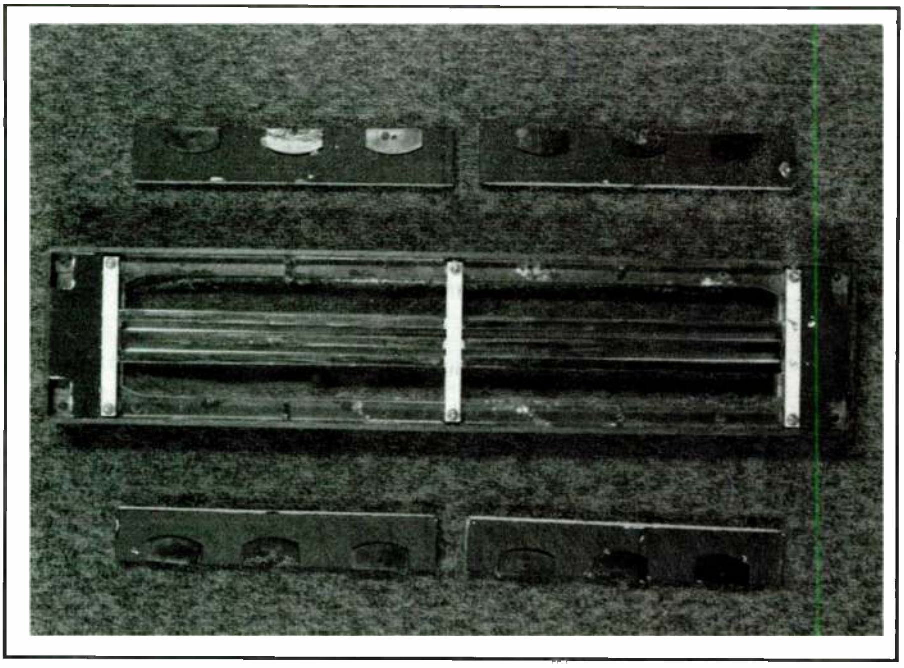


PHOTO 2: Back of the drive and rear wallboards and damping panels.

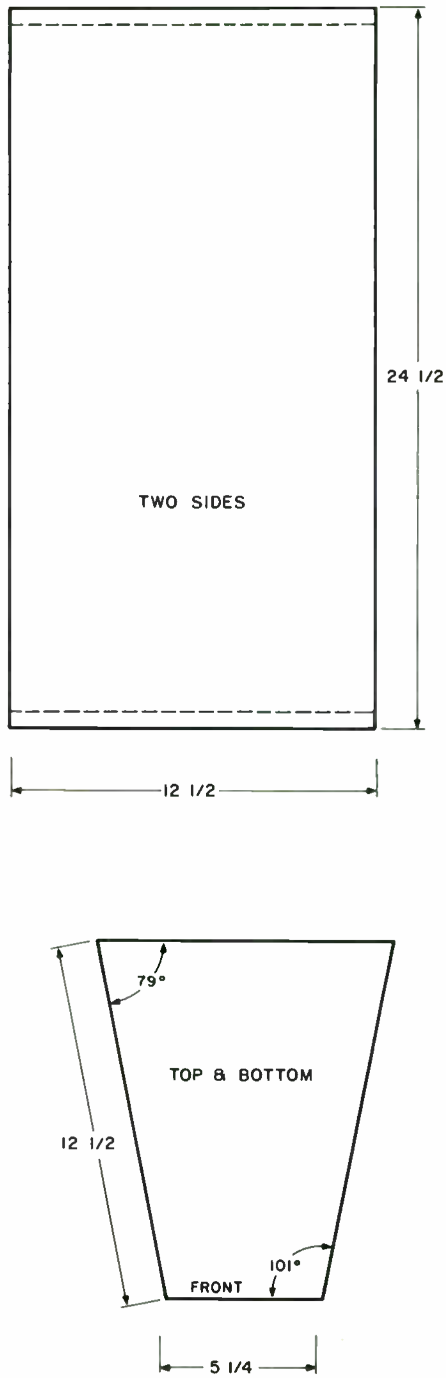


FIGURE 21: The authors recommend these dimensions for their proposed enclosure (see text).

gered by them. Cymbal hits have detail, as the initial hit is followed by a natural decay of the sound. There is no splash, as is often the case with faulty systems. Rim shots are quick to peak and decay, and they have great impact.

The speakers image with pinpoint accuracy and reproduce considerable depth. They do not exhibit the large, blurred image associated with many panel speakers. These drivers should satisfy even the most ardent lover of loud rock music, as they exhibit abundant detail even in very loud passages.

We performed comparison listening tests at two points in the study. First we compared the sound of two stacked Strathearns to that of one driver per side. The stacked drivers had dull transients and lacked high-frequency information. This was confirmed by our FFT tests (see the sidebar on p. 30). The sound with the stacked drivers did have more height, however, but this would be an advantage only if you plan to stand up while you listen.

Our second set of comparison tests were between the Strathearns driven directly from the amplifier (no transformer) and through a corrective pre-filter network with a transformer *(Fig. 19)*. The most important difference we

KITS
from OLD COLONY

Post Office Box 243 • Peterborough, NH 03458-0243

ORDERING INFORMATION Prices, except as noted, are prepaid in the USA and insured. We prefer to ship via UPS, which requires a street address. If you cannot receive UPS delivery, please include an extra $2 for insured service via Parcel Post. We cannot accept responsibility for safety or delivery of uninsured Parcel Post shipments. Charge card orders under $10 please add $1 service charge.

PRICES SUBJECT TO CHANGE WITHOUT NOTICE.

CHARGE CARD PHONE ORDER SERVICE M-F 9am—4pm EDT (603) 924-6371

MasterCard/Visa Cards Welcome. UPS next day and 2nd day air are available to some areas.

FILTERS & SPEAKER SAVER

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

KH-2A: SPEAKER SAVER. [3:77] Turn on/off protection & fast opto-coupler circuitry to prevent damage to your system. 4PDT relay & socket for 2 channels. Each $35.00

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

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

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

CROSSOVERS

KC-4A: ELECTRONIC CROSSOVER, KIT A. [2:72] Single channel, 2-way. All parts including C-4 board and LF351 IC's. Choose frequency of 60, 120, 240, 480, 1k, 2k, 5k or 10k. Each $8.00

KC-4B: ELECTRONIC CROSSOVER, KIT B. [2:72] Single channel, 3-way. All parts including C-4 board & LF351 IC's. Choose frequency of 60, 120, 240, 480, 1k, 2k, 5k or 10k. Each $11.00

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

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

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

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

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

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

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

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

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

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

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

AIDS & TEST EQUIPMENT

KH-7: GLOECKLER PRECISION 101dB ATTENUATOR. [4:77] All switches, 1% metal film and 5% carbon film resistors to build prototype. Chassis, input/output jacks are not included. Each $50.00

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

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

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

KF-4: SINE-SQUARE AUDIO GENERATOR. [4:75] Morrey's MOD kit for Heath IG-18 (IG5218). 2 boards and parts to modify the unit to distortion levels of parts per million range. Each $35.00

● **KG-2: WHITE NOISE/PINK FILTER.** [3:76] All parts, circuit board, IC sockets, 1% resistors, ±5% capacitors. No batteries, power supply or filter switch. CLOSEOUT Each $11.50

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

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

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

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

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

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

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

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

SYSTEM ACCESSORIES

KH-8: MORREY SUPER BUFFER. [4:77] All parts, 1% metal film resistors, NE531 IC's, and PC board for 2 channel output buffer. Each $14.00

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

● **KJ-4: AUDIO ACTIVATED POWER SWITCH.** [3:78] Turn your power amps on and off with the sound feed from your preamp. Includes all parts except box and input/output jacks. CLOSEOUT Each $35.00

● **KK-14A: MacARTHUR LED POWER METER.** [4:79] 2-channel, 2-sided board and all parts except switches, knobs, and mounting clips for LEDs. LEDs are included. No chassis or panel. CLOSEOUT Each $60.00

● **KK-14B: MacARTHUR LED POWER METER.** [4:79] As above but complete with all parts except chassis or panel. CLOSEOUT Each $70.00

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

SBK-E2: NEWCOMB NEW PEAK POWER INDICATOR. [SB 2:84] All parts & board, new multicolor bar graph display; red, green & yellow LED's for 1 channel. No power supply needed. Two for $15.00 Each $9.00

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

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

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

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

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

Perilous Pitfalls

To help you avoid some of the pitfalls in modifying the Strathearn SLC2 speakers, we have tested a number of changes proposed by other speaker builders. *Figure S-1* is the frequency response of a stock Strathearn SLC2 in free air. *Figures S-2* through *S-5* represent the response of the Strathearn midrange ribbon after some modifications that we think are incomplete. These modifications consist of removing the perforated metal and fiberglass damper *(Fig. S-2)* and installing the ribbon in a baffle with or without the rear fiberglass damper *(Figs. S-3* and *S-4)*. *Figure S-5* shows the results of stacking a pair of Strathearn SLC2s.

All these techniques produce measurable—although not always desirable—changes in driver response. Removing the rear fiberglass damper *(Fig. S-2)* produces an increase in the lower midrange response due to increased rear radiation. This balanced high-to-low frequency response is bought at the expense of unloading the diaphragm and allowing its resonances to become more pronounced, with a sharp peak at 2.5kHz and a greater magnitude dip at 3.3kHz.

Installing the Strathearn in a baffle with the rear fiberglass damper intact produces a decrease in output below 2kHz *(Fig. S-3)*, making the driver sound and measure much too bright. Placing the Strathearn in a baffle without the rear fiberglass damper *(Fig. S-4)* produces the flattest response up to 3kHz, but above 3kHz, the undamped diaphragm resonances appear. This configuration produces the greatest low-to-high imbalance, which unmasks the treble resonance.

Stacking two Strathearn SLC2 ribbons produces the frequency response shown in *Fig. S-5*. For this test, we arranged the drivers so that the bottom of the lower driver was 1½ feet off the floor, the same as that of a commercially available system. We placed the

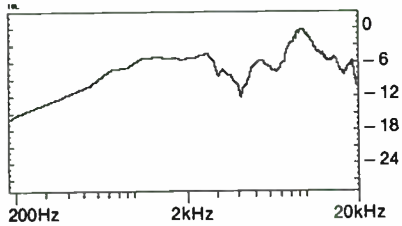


FIGURE S-1: Frequency response of a stock Strathearn SLC2 in free air.

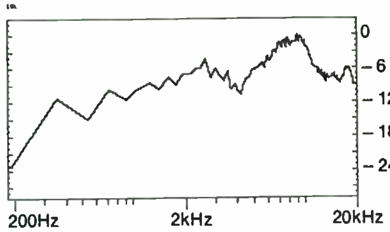


FIGURE S-2: Frequency response of the Strathearn in free air with the rear fiberglass damper removed.

microphone 15 feet from the drivers and 4 feet off the floor, the distance from the floor to the ear of a seated male of average height. The response exhibits three related negative phenomena—a sharp dip centered at about 3.5kHz, a general roll-off above 7kHz and another sharp dip at 17kHz.

This occurs because the Strathearns are not a true line source. Above some key frequency, the individual drivers shift gradually from a line source to a point source. This point source will always be located at the center of the diaphragm. When separate drivers are stacked, the highest frequencies are produced by two point sources separated by a finite distance. The frequency of point-source operation and the distance between the point sources

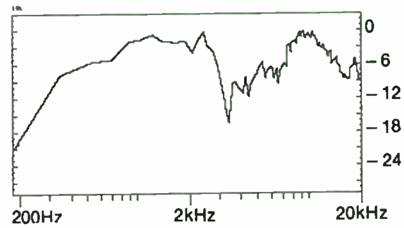


FIGURE S-3: Frequency response of the Strathearn in a baffle with the rear fiberglass damper intact.

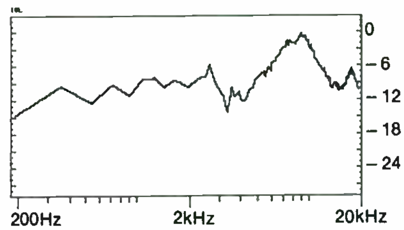


FIGURE S-4: Frequency response of the Strathearn in a baffle with the rear fiberglass damper removed.

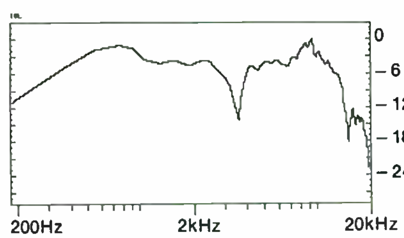


FIGURE S-5: Frequency response of two stock Strathearns stacked vertically.

will combine to cause an interference pattern at some critical ratio. In the case of the Strathearn SLC2 ribbon, that frequency is as low as 3kHz.

noted was with massed violins. The sound with the corrective network was far smoother and more like a concert experience.

Because the output drops swiftly below 350Hz, a simple crossover will suffice and will add almost no coloration to the sound. We suggest adding a 350Hz, 6dB/octave filter at the input to the top-end amplifier in a biamped system. You can do this by inserting a high-quality capacitor between the preamp and amp. The value (in microfarads) is determined by dividing 454.6 by the input resistance of the amplifier. Thus, a 20kΩ input resistance would require a 0.023μF capacitor. Most amplifiers already have a capacitor in the input circuit, and you can use the same formula to calculate a substitute capacitor. Be careful to use the resistance that the capacitor will be loaded. In some units (especially FET front ends), this is different from the amplifier input resistance, as a load resistor precedes the input capacitor.

When all is said and done, we are very pleased with the outcome of our modifications. In our opinion, the modified Strathearns are a superior top-end speaker.

Author's Note: Recent samples of the Strathearn driver indicate that there is a quality-control problem in production. In general, these problem drivers measure less than the specified 0.5Ω. Earlier samples that we tested indicated rather tight quality control with consistent frequency response from unit to unit. This mod is useful only for the original units. Because the problem drivers are not consistent, this mod will not produce the desired results with them.

AN ACOUSTICAL AMBIENCE EXTRACTION SYSTEM

BY DAN T. McGILLICUDDY
Diagrams by Yefim Moshinsky

Stereo recordings generally contain ambient information that could be better extracted and reproduced in the home than it is with today's two-speaker stereo systems. This article describes a system that will extract this information quite well and give you a much better "you are there" feeling.

A BIT OF BACKGROUND. In a concert hall or other recording location, direct sound reaches the listener first. The sound source radiates its signal in all directions, though, not just directly at the listener. The sound bounces off the walls, ceiling and other surfaces before reaching the listener. Because of the size of most recording and performing locations, much of the sound arriving at the listener's ears has bounced off many surfaces. Each surface absorbs some of the sound energy, so the reflected sound is not as loud as the direct sound. In addition, the sound arrives over the course of a second or more.

Reflected sound is called a reverberant field and is responsible for the characteristic sound, or ambience, of a concert hall. The reverberant field of Carnegie Hall, for example, is the main characteristic that makes it sound different from, say, Boston's Symphony Hall.[1]

This reverberant field is largely incoherent—that is, at random phase—so a stereo two-microphone recording system is an ideal setup to capture it. Unfortunately, the invention of the multimicrophone recording system led to less reverberant field pickup. In this method, each instrument or section has a separate microphone aimed at it. Each microphone is often isolated by baffles, preventing it from picking up any other instrument's or section's direct sound. Each microphone is recorded on a separate channel and eventually sent to a mixer to be combined into a two-channel recording.

The mixing operation is done with "pan-pot" controls, which feed each of the respective monophonic signals to both of the "stereo" channels for recording, with only a difference in level between the two channels. These multimike recordings are not true stereo, and they sound deadly dull because the ambient information is not present. In fact, because all phase differences are lost, even localization of individual instruments is more difficult and confusing than in true two-microphone stereo.

Multimike recordings are often processed by artificial reverberation machines such as tape loops, steel plates, springs or the more modern electronic devices I will discuss later. As a much better alternative, some recording companies add extra microphones, set in the audience, to pick up the reverberant field. If the company uses two microphones, and if each is fed exclusively to only one channel of the stereo recording, the lost ambient information can be restored to the mix-down.

As long as the ambient information is recorded separately, why not play it back separately in a four-channel, or quadraphonic, system? Four-channel reel-to-reel tapes can be quite successful in recreating the ambient field for the listener. Unfortunately, when four-channel technologies turned to phonograph records, they came up with many incompatible systems. CD-4, a pseudo-discrete system championed in the US by RCA, works by modulating a 30kHz carrier with the two extra channels. Of course, this system is incompatible with the current system of FM broadcasting, as well as with the other quadraphonic systems.

The SQ system, championed here by CBS, is a matrix system—that is, it sends the four channels through phase-shifting networks and adds them together to come up with two channels. In its simplest form, the decoder of the SQ signals leaves little separation between some of the reconstructed four channels, but improved versions are better in this respect.

Although consumer resistance to this incompatibility has been strong enough to kill off most of the systems, there is some chance of resurrecting the SQ system (for which limited quantities of records are still being produced by Angel and CBS) because it has been approved by the FCC (Federal Communications Commission) for use in FM broadcasting.[2] In addition, Audionics' Space and Image Composer and the Scheiber decoder are available to decode SQ. Both are extremely good at reconstructing the original fields when used with the system described in this article.

Recently, a group of devices that take two-channel recordings and create artificial reverberation by electronic means has become popular for home use.[3] These devices send the signal through a delay line, then feed a portion of the delayed signal back into the input to be delayed again. One or more of these echo loops are placed between and among the channels in various ways to simulate the multiple reflections of the sound wave in the concert hall. In most such devices, you can adjust the initial delay, as well as the degree of recirculation, to try to imitate different-sized spaces.

Some of these devices use analog delay lines. These are usually charge-coupled devices from manufacturers such as Sound Concepts, Bose and Southwest Technical Products. Other devices from Audio Pulse, Advent, Koss and ADS convert the analog signal to a digital form, then send that signal through shift registers or a random access memory. The device then reconverts the signal to its analog form, which is mixed back into the input. These devices are similar to, though simpler than, the artificial-reverberation devices used by many recording companies to add the missing ambience to multimike recordings.[4]

The problem with using these devices in the home is that they lay a reverberant field over the ambient field already present on most recordings. Sometimes, particularly when a recording is relatively dry, the result can be quite pleasing, but at other times the fields clash badly, leading to a muddy sound. In any case, these devices do little to bring out the distinctive sound of the recording location. Rather, they create an artificial location that makes all halls sound the same.

One group of devices, which I will call image processors, come from manufacturers such as Carver, Omnisonic and Sound Concepts. These devices intend to recreate the original sound field of the recording location with the use of just the two main speakers. Most of these are derived from the work of Benjamin B. Bauer, who tried to remove the "inside the head" feeling one experiences when listening to stereo recordings through headphones.[5] He did this by inserting a crossfeed circuit between the channels. He also found that speakers could be made to sound more like headphones if you apply the inverse circuit, which effectively removes the crossfeed between channels that occurs acoustically with loudspeakers.

Manfred R. Schroeder expanded this idea by attempting to remove the effects of the listening room. He called his system "Super-Stereo."[6] Unfortunately, you must compute an exact mathematical model of the listening room, and the system's effect is very limited. Moving even a few feet away from the optimum listening position can completely shatter the "you are there" illusion.

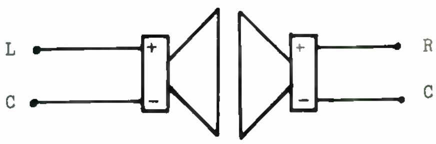


FIGURE 1: In the standard stereo speaker connection, two speakers are connected to two separate amplifiers. The independent impedance is Z.

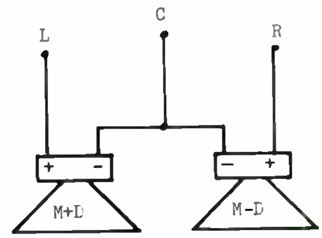


FIGURE 2: With the common wires connected, the standard stereo system has a mono impedance, difference impedance and effective impedance of Z.

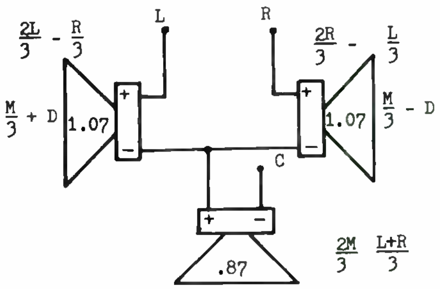


FIGURE 3: Three-speaker configuration in which the independent impedance is 1.5Z, the mono is 3Z, the difference is Z, and the effective is 1.73Z.

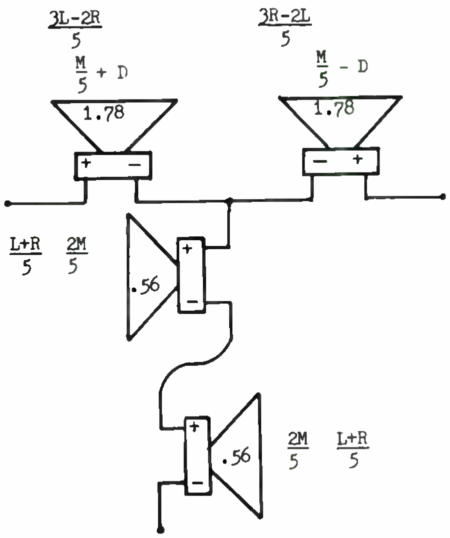


FIGURE 4: Four-speaker configuration in which the independent impedance is 1.67Z, the mono is 5Z, the difference is Z, and the effective is 2.24Z.

AMBIENCE AND ACOUSTICS. Finally, we come to systems designed to extract the ambient information from recordings, using speakers and the listening room itself. One such system, commercially successful as a stereo-pair speaker system, uses nine series/parallel-connected drivers in each speaker. One driver faces front for the direct sound; the remaining eight face backward to reflect off the listening room wall. These reflecting speakers are supposed to form a reverberant field around the listener.

This commercial system uses an active equalizer to extend the bass response of the full-range drivers, leading to low efficiency and, therefore, higher distortion and the use of larger amplifiers. Moreover, the reflected image is delayed only six milliseconds (the time the sound takes to travel 3 feet to the wall and back), leading to a smearing or blurring of the sound. Although this short delay is within the time period in which the precedence, or Haas, effect operates, that effect is used for direction finding and is not effective for sound sources in the same direction. In this case, the echo makes speech more difficult to understand and leads to muddy musical textures.[7]

A better system would use a short time delay between the main speakers in front and a separate set of rear speakers for the ambient sound.[8] Extraction of the ambient information on the recordings can be optimized by using something other than a simple series/parallel connection pattern between the individual speakers. *Figures 3–8* demonstrate how this is done with the use of three through eight speakers.

First, however, refer to *Fig. 1*, which shows the standard stereo speaker connection. Two speakers are connected to two separate amplifiers, each of which can be considered to have one wire with a signal and one wire as a common ground. This arrangement is redrawn in *Fig. 2* with the common wires connected. *Figure 1* shows that all of the left signal appears exclusively in the left speaker and all of the right signal in the right speaker. *Figure 2* shows the left and right signals redefined. I used a monophonic signal, M, common to both channels, and a difference signal, D, which is the difference between the two channels.

$$M = \frac{L+R}{2}$$

and

$$D = \frac{L-R}{2}$$

Therefore,

$$L = M+D$$

and

$$R = M-D$$

These sum and difference signals are used in FM broadcasting so that stereo FM is compatible with mono FM. On phonograph records, the M signal appears as lateral modulation of the groove and the D signal as vertical modulation. The ambient information is largely contained in the D signal.

Figure 3 shows three speakers connected in the manner David Hafler used in his design of an ambient extractor for Dynaco. (The speaker connected to the common ground is usually replaced by a resistor because it has only an M signal. You can also do this with the speakers in any of the given configurations if the speakers have no D signal. Doing so results in little loss of ambience extraction.) As you can see in *Fig. 3*, the ambience contained in the D signal is even further separated from the M signal. Using the configurations in *Figs. 3–8* produces different results. (*Figures 5, 7* and *8* even include two alternate topologies for five, seven and eight speakers, respectively.) The number inside each speaker is the relative level of sound produced by that speaker in that configuration.

You can use these arrangements with any amplifier whose outputs can be connected. Because most headphones have this connection method, you can also use the majority of available amplifiers this way.

I chose the configurations in *Figs. 3–8* to maximize the extraction of ambient information, while keeping the impedances seen by the amplifier channels at or above the levels seen in the standard stereo connection. The impedance of each speaker in *Fig. 1* is defined as Z, so each amplifier sees an independent impedance of Z.

Figure 2 is more difficult, however, since it has two signals to consider. The easiest way to figure the mono impedance seen by each amplifier is to mentally short point L to point R and compute the impedance between that point and point C. Doubling this value produces the mono impedance (MI). To find the impedance seen by the difference signal, mentally open-circuit (disconnect) point C and compute the impedance from point L to point R. Half of this value is the difference impedance (DI). The overall effective impedance, then, is given by the following formula:

$$\text{Effective Impedance} = \sqrt{(MI) \times (DI)}$$

All of these impedances are Z in the

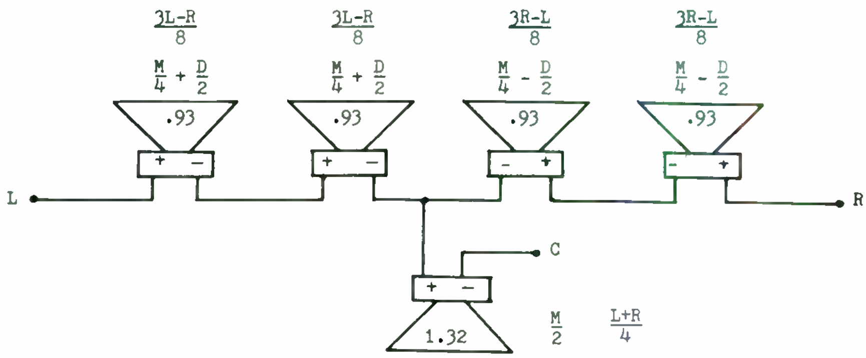


FIGURE 5a: Standard five-speaker topology, with the independent impedance equal to 2.67Z, the mono equal to 4Z, the difference equal to 2Z and the effective equal to 2.83Z.

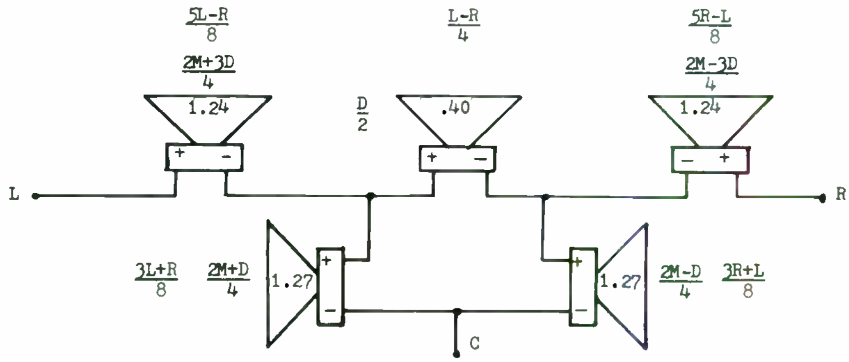


FIGURE 5b: Alternate five-speaker topology. Here, the independent impedance is 1.6Z, the mono is 2Z, the difference is 1.33Z and the effective is 1.63Z.

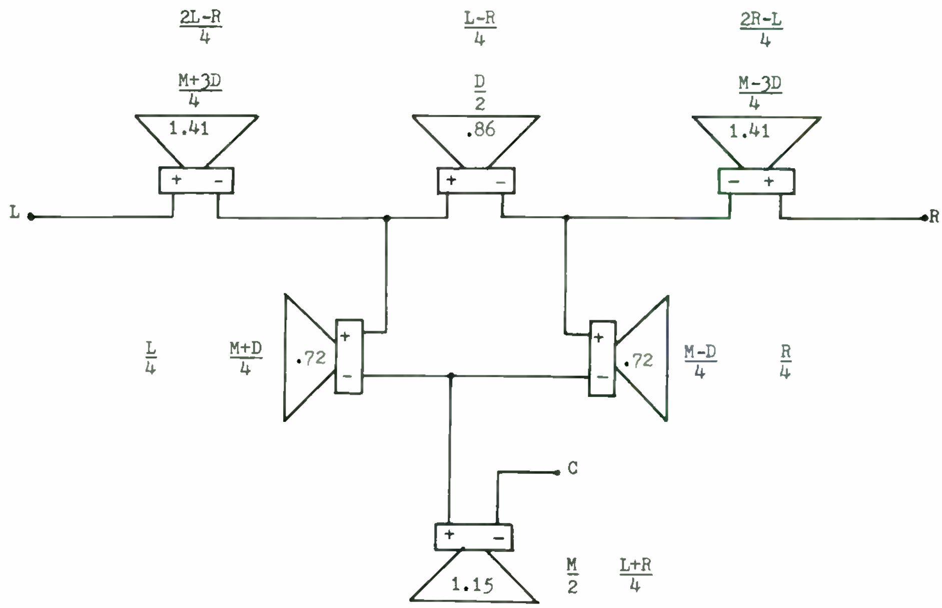


FIGURE 6: This six-speaker arrangement has an independent impedance of 2Z, a mono of 4Z, a difference of 1.33Z and an effective of 2.31Z.

Old Colony Sound Lab's

BOOK SERVICE

PENGUINS

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

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

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

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

McGRAW-HILL

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

MH-2 HOW TO MAKE PRINTED CIRCUIT BOARDS *by Joel Goldberg.* All the topics on making your own circuit boards are covered both in theory and in practical advice: design and layout, artwork preparation, photo layout, silk screening and etching. Each **$6.95**

HOWARD W. SAMS

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

S-19 SOUND SYSTEM ENGINEERING *by Don and Carolyn Davis.* A thorough introduction to sound systems for halls, studios, outdoor locations and much else. 245pp., softbound, 8½x11. Each **$21.95**

DOVER BOOKS

D-1 REPRODUCTION OF SOUND *by Edgar Villchur.* An elegantly simple 92-page paperbound primer on the subject by the man who invented the bookshelf speaker. Loan or give it to friends who want to build a system as good as yours. Each **$2.50**

D-3 MUSIC, PHYSICS AND ENGINEERING *by Harry F. Olson.* A thorough introduction to the physical characteristics of sound and the relationship of sound to musical instruments by the former head of staff at RCA's lab for acoustical and electromechanical research at Princeton, NJ. A classic by one of the giants in the audio field. Good, easy to read chapters on acoustics, mikes and recording, recording and playback systems, as well as an electronic music chapter. 2nd. Ed. (1967) 460pp., softbound. Each **$6.95**

TAB BOOKS

T-4 DESIGNING, BUILDING AND TESTING YOUR OWN SPEAKER SYSTEM (2ND ED.) *by David Weems.* A cookbook approach to speaker design with 10 tests for performance. Individual drivers are specified for actual building projects. Weems likes closed box and reflex enclosures, but is skeptical of transmission lines. Includes a BASIC program for Thiele/Small parameters. 190pp., softbound. Each **$9.95**

WILEY BOOKS

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

Old Colony's

BOOK SERVICE

To order from Old Colony Sound Lab, please write each book's number below with quantity of each and price. Total the amounts and remit by check, money order, MasterCard, or Visa. *For charge card orders under $10 add $1 service charge.* Please add 50¢ for postage on the first book; 25¢ per additional book. Canadians please add 10% for postage. All remittances must be in U.S. funds. *Please use clear block capitals.* NOTE: *Prices are subject to change without notice.*

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

Name ____________________

Street & No. ____________________

City ____________________

State ______________ ZIP ______________

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

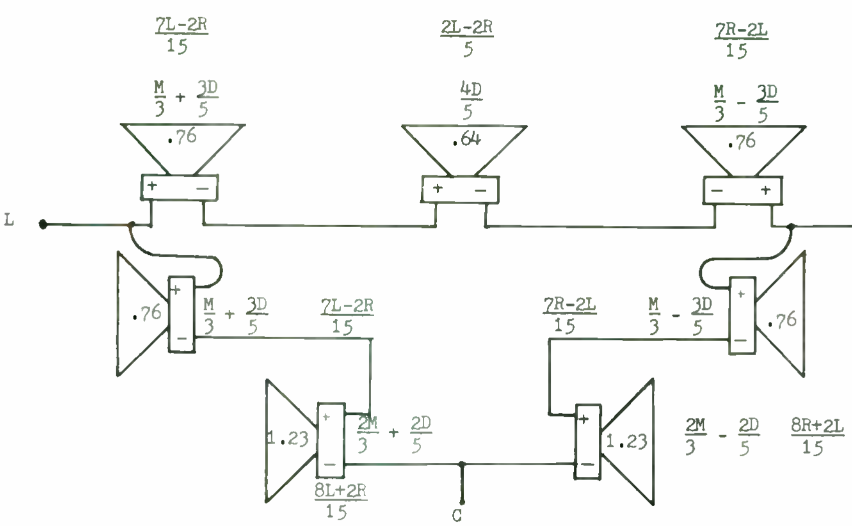


FIGURE 7a: The standard seven-speaker system has an independent impedance of 1.07Z, a mono of 1.5Z, a difference of 0.83Z and an effective of 1.12Z.

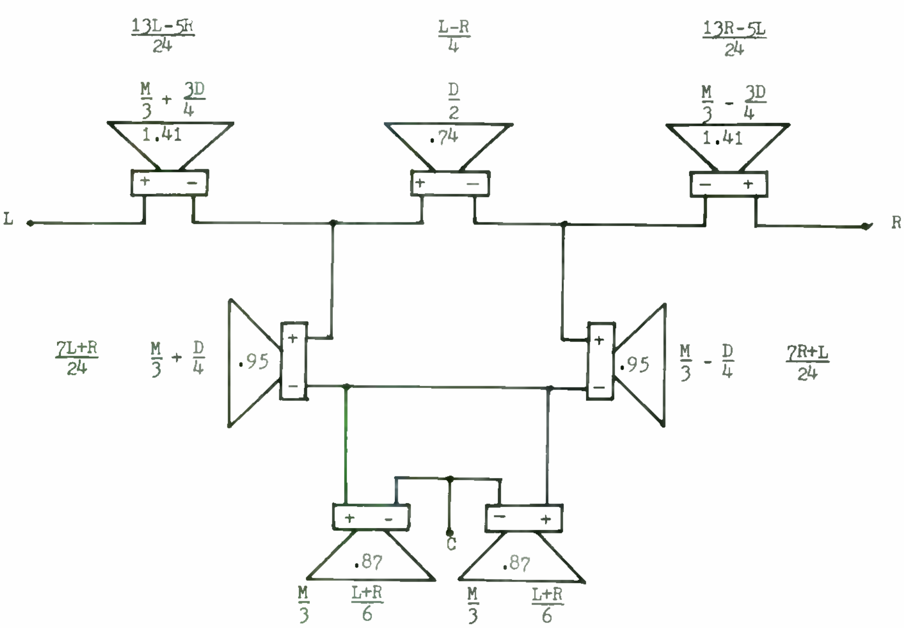


FIGURE 7b: With this alternate seven-speaker plan, the independent impedance is 1.84Z, the mono is 3Z, the difference is 1.33Z, and the effective is 2Z.

Fig. 2 configuration, of course, but the values for *Figs. 3–8* are somewhat different. The independent impedance is most easily found in *Figs. 3–8* by shorting point R to point C and computing the impedance from that point to point L.

AMBIENCE EXTRACTION. To provide a reverberant field that is spread out, David L. Clark and Bernhard F. Muller[9] suggest using speaker cabinets that sit on the floor and aiming the drivers 30 degrees up toward the walls and ceiling of the listening room. *Figure* 9 is the cutting plan for cabinets adapted from their design. *Figure 10* shows the materials you will need for construction. Note that for the front panel, you can use two ¼-inch pieces of plywood rather than one ½-inch piece. This allows you to avoid routing out the driver-mounting areas. Both the front brace and gluing block for the corner between the back and bottom should be length L. Use ordinary white glue, as well as screws, on all edges. To ensure a tight seal, you should also apply silicone or some other sealant to the inside corners and edges.

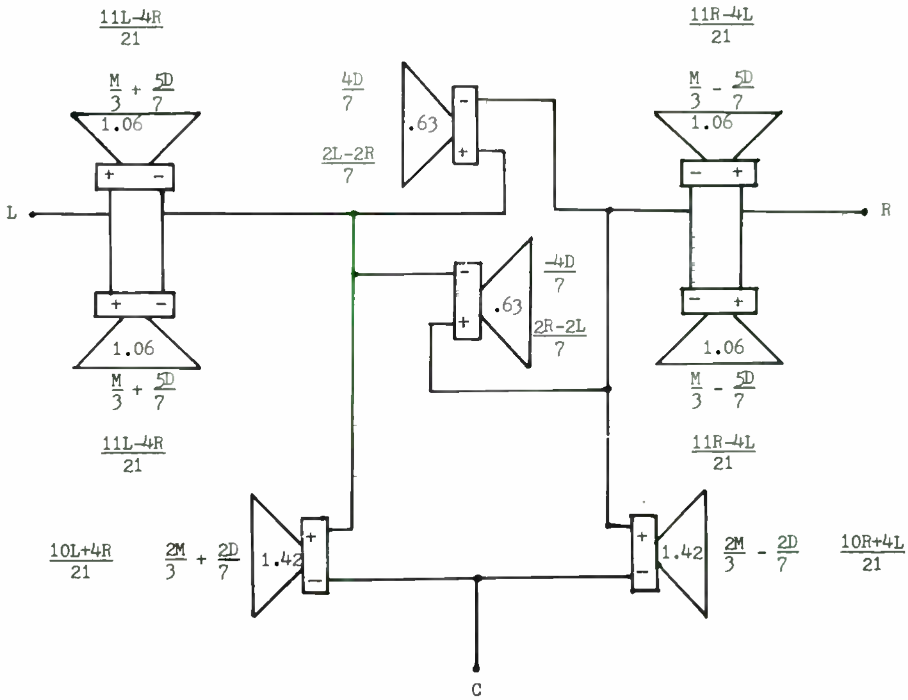


FIGURE 8a: Standard eight-speaker system, where the independent impedance is 0.95Z, the mono is 1.5Z, the difference is 0.7Z and the effective is 1.02Z.

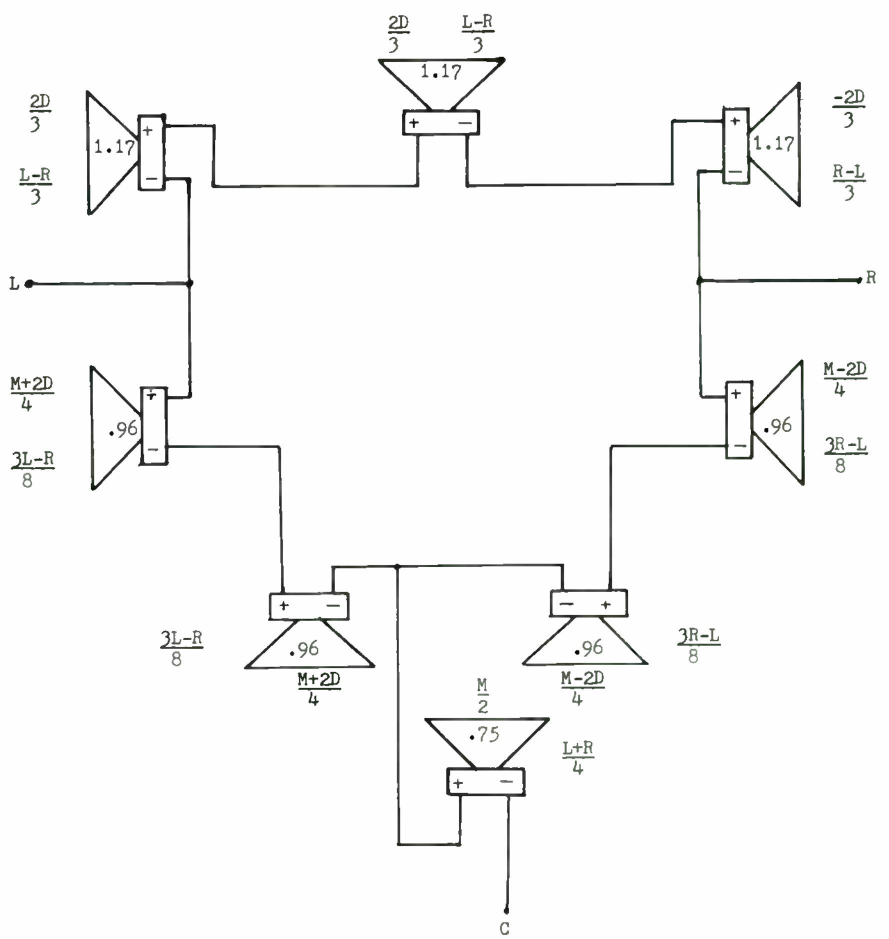


FIGURE 8b: One variation of Fig. 8a produces an independent impedance of 1.41Z, a mono of 4Z, a difference of 0.86Z and an effective of 1.85Z.

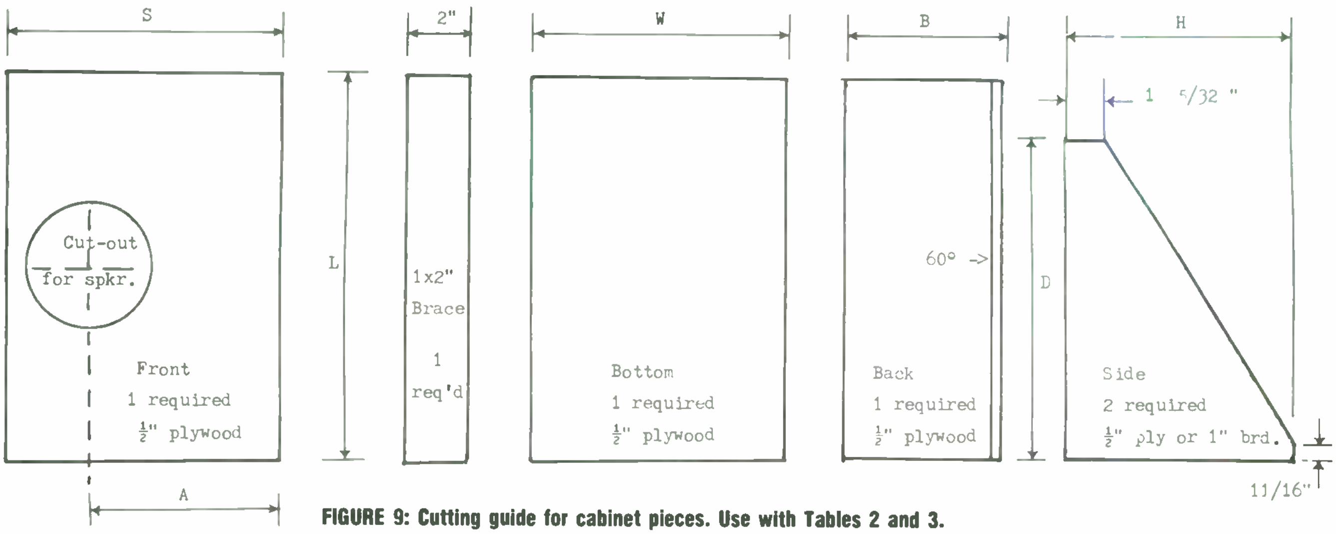


FIGURE 9: Cutting guide for cabinet pieces. Use with Tables 2 and 3.

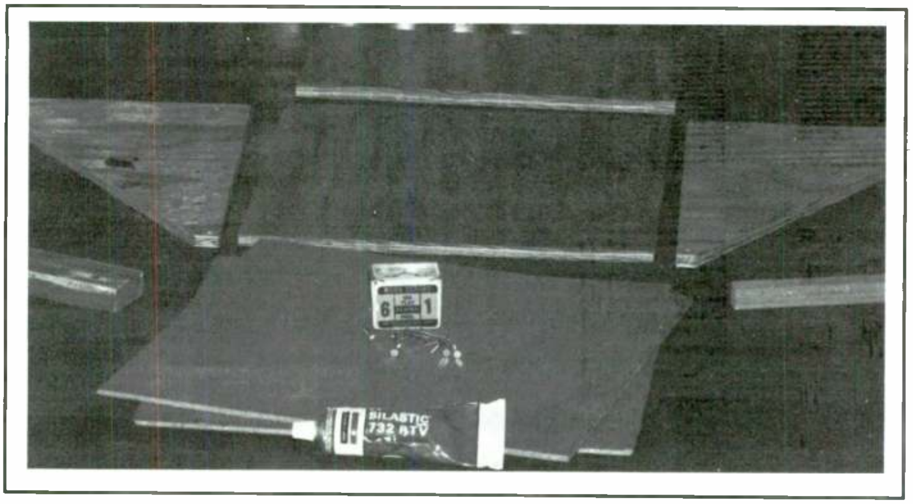


FIGURE 10: Materials needed for construction. Notice that instead of one ½-inch piece of plywood for the front panel, the author has cut two ¼-inch pieces.

Figure 11 shows the cabinet partially assembled, with the 1-by-2-inch brace and gluing block installed, but without the sides. You can paint this subassembly flat black (or white, to match the speaker cone color if you use different drivers) and apply grille cloth after the drivers are installed. Glue and/or screw the two sides into place, with the screws countersunk, and paint them. (See *Fig. 12.*) Perhaps you will want to stain the sides instead if they are made from 1-inch board stock.

Table 1 lists a number of driver alternatives. Two of these are full-range drivers for the simplest systems. I have provided tweeter recommendations for the others. Tweeter A is the CTS 2T2-NFR, tweeter B is the Speakerlab DT50, tweeter C is the KEF T27

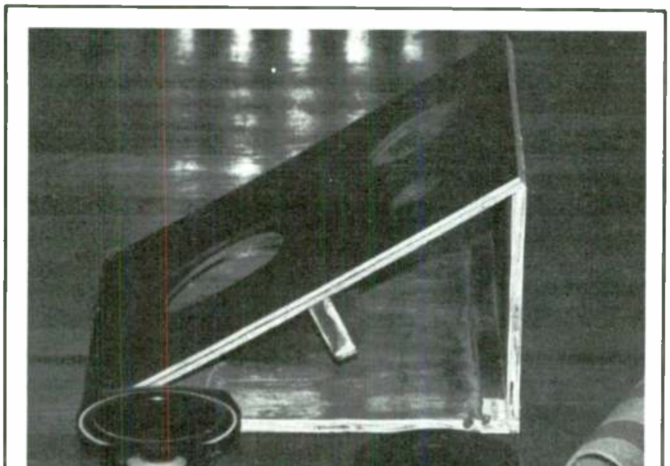


FIGURE 11: In the finished subassembly, note the placement of the brace and gluing block. This box is the vented alignment for the Polydax HD13B25J2C12 and the KEF T27 SP1032. It also appears in Figs. 10, 12, and 13a.

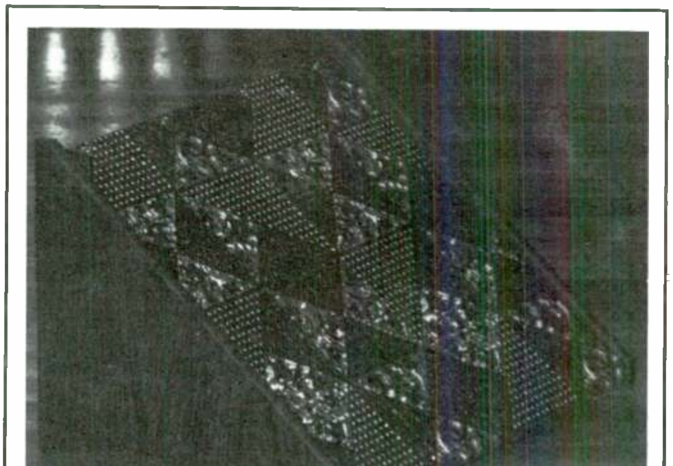


FIGURE 12: The completed speaker. The author finds it easier to glue and caulk a wire through the bottom than to install a barrier strip or other connector for the speaker wire. You can then solder longer wires to these shorter ones. This prevents problems such as disconnected wires and poor contacts.

SP1032, tweeter D is the Polydax HD13D34E, and tweeter E is the Polydax HD13D34H. *Table 2* lists the cutting dimensions for *Fig. 9*, using each driver from *Table 1* and an acoustic suspension design with a 0.7 system Q.[10] *Table 3* is a list of vented-box alignments for most of the drivers from *Table 1*, giving the cutting dimensions for *Fig. 9* and the sizes of the required vents.[11] In some cases, the vents are too long to fit inside the cabinets. For these, cut the vents at an angle between 60 and 45 degrees at one or more points along their lengths, then glue them back together to form bends of between 120 and 90 degrees.

Please note that although the vented alignments provide deeper bass response and are larger, they should be used only in the simpler topologies to prevent a loss of amplifier damping. Also note that the smaller acoustic suspension designs do not have sufficient front panel space to mount the brace.

Figure 13 shows the simple series constant-voltage crossover, including the component values,[12, 13] to be used with tweeter A. C1 is made up of a 15μF non-polarized (NP) electrolytic capacitor in parallel with a 1μF Mylar or other film capacitor. For best results, connect optional resistors to attenuate the tweeter, as shown in *Fig. 14*. When used with woofer CTS 6W10R, R1 should be 2.2Ω, 2W, while R2 should be 22Ω, 1W. When used with woofer CTS 6W10C, R1 should be 3.3Ω, 3W and R2 15Ω, 1W. The crossover frequency is 2.5kHz.

Construct L1 according to *Fig. 16* out of a ½-inch length of ½-inch-diameter wooden doweling, glued between two wooden end pieces, and a 33-foot, 6¾-inch length of AWG #18 wire. Wind that exact length onto the form, measuring out extra wire for the leads. This produces approximately 89 turns, with a weight of about 3.4 ounces. The value of L1 is 0.2548mH. If you follow these dimensions exactly, the inductance will have the correct value.[14]

The crossovers in *Figs. 14* and *15* contain components (C2, C3, R3, L2 and the 8Ω resistor) that cancel out the impedance anomalies of the drivers to which they are connected[15] so that the crossover networks see the nominal 8Ω at all frequencies. The more expensive drivers do not have greater anomalies, but at that level, it is worthwhile to eliminate them. In all cases, you should make the inductors exactly as I have described because their resistance forms all or part of R3.

TABLE 1

DRIVER ALTERNATIVES

Speaker	*	f_S	Q_{TS}	V_{AS}	Eff.	Sens.	Crossover Fig.
CTS 4½SR10B	FR	82	0.43	3.3	0.278	85.2	—
CTS 6W10R	TW-A	40	.29	35	0.685	89.2	14
CTS 6W10C	TW-A	55	0.38	28	0.926	87.2	14
Dalesford D30/110	TW-D	32	0.35	14.86	0.128	84	16
Dalesford D153	TW-D	32	0.36	34.28	0.304	87	16
Great America 6D20J1	TW-B	42	0.31	19.8	0.409	—	15
KEF B110 SP1003	TW-C	38	0.31	23.6	0.434	83.1	16
KEF B110 SP1057	TW-C	38	0.33	23.6	0.348	—	16
Polydax HD10P25FSM	FR	61	0.43	6.57	0.309	87	—
Polydax HD13B25H2C12	TW-D	35	0.29	23.29	0.353	84.2	16
Polydax HD13B25H4C12	TW-D	30	0.18	23.29	0.323	84.4	16
Polydax HD13B25J2C12	TW-C	35	0.33	23.29	0.512	82	16
Polydax HD13B25J4C9	TW-D	32	0.21	23.29	0.324	84.2	16
Polydax HD17B25H	TW-E	38	0.345	26.16	0.653	91	16
Polydax HD17B25H2C12	TW-D	29.5	0.34	39.6	0.261	84	16
Polydax HD17B25H4C12	TW-C	27.5	0.19	39.6	0.402	83.4	16
Polydax HD17B25J	TW-E	38	0.449	26.16	0.282	89	16
Polydax HD17B25J2C12	TW-C	29.5	0.41	39.6	0.216	83.6	16
Polydax HD17B37	TW-E	35	0.239	27.76	0.242	89	16
Polydax HD17B37R2C12	TW-D	30	0.21	38.76	0.464	85.6	16
Polydax HIF13H2C12	TW-D	42	0.29	14.82	0.325	87	16
Polydax HIF113J	TW-E	42	0.319	19.54	0.390	—	16
Polydax HIF13J2C12	TW-D	42	0.35	14.82	0.261	86	16
Peerless K050WG	TW-B	50	0.317	10.8	0.349	88.5	15
Speakerlab M608R	TW-B	42.8	0.383	15.5	0.258	89.3	16

* FR = Full Range; TW = tweeter recommended in text
f_S in hertz; V_{AS} in liters; Efficiency (Eff.) in percent; Sensitivity (Sens.) in dB @ 1W @ 1 meter

Fig. 14 shows the crossover (at 4kHz) for the Speakerlab DT50 tweeter. See *Table 4* for appropriate woofer values. C1 is made up of a 6.8μF NP electrolytic cap in parallel with a 3μF Mylar or film cap. L1 is wound on a half-inch-diameter wooden dowel, as shown in *Fig. 16*, with a wire length of 27 feet, 9¾ inches. Use AWG #16 magnet wire, with about 70 turns and a weight of

TABLE 2

CUTTING DIMENSIONS (Fig. 9)

Speaker	V_b	f_3	L	W	B	H	D	S	A
				Cutting Dimensions					
Polydax HD13B25H4C12	125	118	7⅛	8⅞	5¹⁄₁₆	6⁷⁄₁₆	9⅜	10	5*
CTS 4½SR10B	145	135	8¼	8⅞	5¹⁄₁₆	6⁷⁄₁₆	9⅜	10	5
Polydax HD13B25J4C9	168	108	9⅝	8⅞	5¹⁄₁₆	6⁷⁄₁₆	9⅜	10	5*
Peerless K050WG	195	112	7½	10⅝	6¹⁄₁₆	7⁷⁄₁₆	11⅛	12	6½*
Polydax HIF13H2C12	214	102	8⅛	10⅝	6¹⁄₁₆	7⁷⁄₁₆	11⅛	12	6¼*
Polydax HD17B25H4C12	221	102	8⅜	10⅝	6¹⁄₁₆	7⁷⁄₁₆	11⅛	12	6*
Polydax HD17B37	252	104	9⅝	10⅝	6¹⁄₁₆	7⁷⁄₁₆	11⅛	12	6*
Polydax HD17B37R2C12	263	101	10	10⅝	6¹⁄₁₆	7⁷⁄₁₆	11⅛	12	6*
Polydax HD10P25FSM	267	100	10¼	10⅝	6¹⁄₁₆	7⁷⁄₁₆	11⅛	12	6¾*
Polydax HD13B25H2C12	323	85	10⅜	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	7
Great America 6D20J1	323	96	10⅜	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	6¾
Polydax HIF13J2C12	329	85	10½	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	7
Dalesford D30/110	331	65	10⅝	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	7¼
Polydax HIF13J	340	93	10⅞	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	7
KEF B110 SP1003	379	87	12⅛	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	7¼
Speakerlab M608R	432	79	13⅞	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	6¾
Polydax HD13B25J2C12	434	75	13⅞	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	7
KEF B110 SP1057	439	81	14⅛	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	7¼
CTS 6W10R	472	98	15⅛	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	6¾
Polydax HD17B25H	541	78	17⅜	11½	6⁹⁄₁₆	7¹⁵⁄₁₆	12	13	6¼
CTS 6W10C	740	102	15¼	14⅛	8¹⁄₁₆	9⁷⁄₁₆	14⅝	16	9¾
Polydax HD17B25H2C12	774	61	15⅞	14⅛	8¹⁄₁₆	9⁷⁄₁₆	14⅝	16	9½
Dalesford D153	779	63	16	14⅛	8¹⁄₁₆	9⁷⁄₁₆	14⅝	16	9¾
Polydax HD17B25J	1131	60	23¼	14⅛	8¹⁄₁₆	9⁷⁄₁₆	14⅝	16	9½
Polydax HD17B25J2C12	1279	51	26¼	14⅛	8¹⁄₁₆	9⁷⁄₁₆	14⅝	16	9½

V_b in inches^3; f_3 in hertz; all others in inches
* Woofer must be mounted off-center to allow room for tweeter.

TABLE 3

VENTED-BOX ALIGNMENTS*

Speaker	V_b	f_3	Vent Dia.	Vent Lgth.	L	W	Cutting Dimensions B	H	O	S	A
CTS 4½SR10B	293	72	2	9¼	9⅜	11½	6 9/16	7 15/16	12	13	7
Polydax HD13B25J4C9	304	75	2	20¼	9¾	11½	6 9/16	7 15/16	12	13	6¾
Peerless K050WG	351	71	2	13⅛	11¼	11½	6 9/16	7 15/16	12	13	7
Polydax HIF113H2C12	372	68	2	15	11⅞	11½	6 9/16	7 15/16	12	13	6¾
Polydax HD17B25H4C12	425	78	2	16⅛	13⅝	11½	6 9/16	7 15/16	12	13	6¼
Polydax HD17B37	438	71	2	12⅞	14	11½	6 9/16	7 15/16	12	13	6½
Polydax HD17B37R2C12	445	69	2	13⅜	14¼	11½	6 9/16	7 15/16	12	13	6¼
Polydax HD10P25FSM	521	53	2	8¼	16¾	11½	6 9/16	7 15/16	12	13	7¾
Polydax HD13B25H2C12	532	56	2	13⅝	17⅛	11½	6 9/16	7 15/16	12	13	7
Great America 6D20J1	546	62	2	10⅛	17½	11½	6 9/16	7 15/16	12	13	6¾
Polydax HIF13J	565	59	2	9⅞	18⅛	11½	6 9/16	7 15/16	12	13	7
Polydax HIF13J2C12	574	52	2	12	18⅜	11½	6 9/16	7 15/16	12	13	7
Dalesford D30/110	602	40	2	21⅝	19¼	11½	6 9/16	7 15/16	12	13	7¼
KEF B110 SP1003	617	56	2	10⅜	19¾	11½	6 9/16	7 15/16	12	13	7¼
KEF B110 SP1057	724	51	2	9⅝	14⅞	14⅛	8 1/16	9 7/16	14⅝	16	10¼
Polydax HD13B25J2C12	726	47	2	11¾	14⅞	14⅛	8 1/16	9 7/16	14⅝	16	10
CTS 6W10R	781	64	2½	10¼	16	14⅛	8 1/16	9 7/16	14⅝	16	9¾
Speakerlab M608R	806	43	2	9	16½	14⅛	8 1/16	9 7/16	14⅝	16	9¾
Polydax HD17B25H	929	48	2	8	19⅛	14⅛	8 1/16	9 7/16	14⅝	16	9½
CTS 6W10C	1315	60	2½	3¾	18¾	16¾	9 9/16	10 15/16	17¼	19	12¾
Polydax HD17B25H2C12	1349	38	2½	14¾	19¼	16¾	9 9/16	10 15/16	17¼	19	12½
Dalesford D153	1379	38	2½	13⅛	19⅝	16¾	9 9/16	10 15/16	17¼	19	12¾
Polydax HD17B25J	2436	32	2½	6½	23¼	20⅛	11 9/16	12 15/16	20⅝	23	16½
Polydax HD17B25J2C12	2539	29	2½	12	24¼	20⅛	11 9/16	12 15/16	20⅝	23	16½

V_b in inches^3; f_3 in hertz; all others in inches
* See Table 1 and Fig. 9.

TABLE 4

DRIVER OPTIONS FOR FIG. 14*

Woofer	R1	R2	C2
Great America 6D20J1	3.3Ω, 3W	27Ω, 1W	10μF
Peerless K050WG	3.3Ω, 3W	27Ω, 1W	15μF
Speakerlab M608R	2.2Ω, 2W	—	10μF

* Crossover for the Speakerlab DT50 tweeter.

TABLE 5

WOOFER OPTIONS FOR FIG.15

KEF T27 SP1032 Tweeter

Woofer	C2
KEF B110 SP1003	6.8μF
KEF B110 SP1057	6.8μF
Polydax HD13B25J2C12	6.8μF
Polydax HD17B25H4C12	15μF
Polydax HD17B25J2C12	8.2μF

Polydax HD13D34E Tweeter

Woofer	R1	C2
Dalesford D30/110	2.2Ω, 3W	25μF
Dalesford D153	short	25μF
Polydax HD13B25H2C12	2.0Ω, 3W	3.3μF
Polydax HD13B25H4C12	1.8Ω, 2W	15μF
Polydax HD13B25J4C9	2.0Ω, 3W	15μF
Polydax HD17B25H2C12	2.2Ω, 3W	3.3μF
Polydax HD17B37C12	1.0Ω, 2W	8μF
Polydax HIF13H2C12	short	3.3μF
Polydax HIF13J2C12	0.56Ω, 1W	6.8μF

Polydax HD13D34H Tweeter

Woofer	R1	R2	C2
Polydax HD17B25H	1.0Ω, 2W	—	4.7μF
Polydax HD17B25J	2.2Ω, 3W	39Ω, ½W	6μF
Polydax HD17B37	2.2Ω, 3W	39Ω, ½W	6.8μF
Polydax HIF13J	1.5Ω, 3W	—	6.8μF

about 3.4 ounces. The value of L1 is 0.15925mH.

Several options are available for the crossover in *Fig. 15*. If you use the KEF T27 SP1032 tweeter, crossing over at 3.5kHz, C1 is a 4.7μF NP electrolytic in parallel with a 6.8μF film cap. C3 is made up of parallel NP capacitors with values of 10μF and 3.3μF. L1 is wound on a half-inch-diameter wooden dowel *(Fig. 16)*, with a wire length of 30 feet, 1½ inches. Use AWG #16 magnet wire, with about 75 turns and a wire weight of 3.4 ounces. L1 equals 0.182mH. L2 is wound on a 5/16-inch dowel, with a wire length of 43 feet, 1 inch. Use AWG #34 wire, with about 363 turns and a weight of 0.1 ounce. The value of L2 is 1.33mH. Replace R1 with a short, and do not use R2. R3 is part of L2. Add series resistance if the resistance of the wire in L2 is less than 8.75Ω.

If you use the Polydax HD13D34E tweeter, crossing over at 3.5kHz, the values change. C1 is a 4.7μF NP cap in parallel with a 6.8μF film cap, while C2 is made up of two parallel 4.7μF NP caps. L1 is wound in the same way as with the KEF tweeter, and it has the same value. L2 is wound on a 7/16-inch wooden dowel, with a wire length of 79 feet, 11 inches. Use AWG #30 magnet wire with about 441 turns and a weight of 0.28 ounce. L2, including R3, equals 2.74mH. Add series resistance if the resistance of the wire in L2 measures less than 9.07Ω.

A third option is to use the Polydax HD13D34H tweeter, crossing over at 3.5kHz. C1 is made up of a 4.7μF NP cap in parallel with a 6.8μF film cap, while C3 is a 22μF NP cap. Wind L1 as above, producing the same value. Use a 3/8-inch dowel to wind L2 with a length of AWG #32 magnet wire measuring 48 feet, 9 inches. This produces about 340 turns, a weight of about 0.17 ounce and a value of 1.40mH. R3 is part of L2, but you will probably have to add series resistance to bring the value up to 7.95Ω. If you cannot make an accurate measurement, add a 1Ω, 1W resistor in series with L2.

See *Table 5* for the various woofer options and component values that complement these tweeters.

Connect the finished speakers to two amplifier channels, as shown in *Figs. 3–8*. In any configuration, you may replace any speaker that reproduces only the M signal with an 8 to 10Ω, 10W resistor with little loss of ambience extraction. Place the speakers on the floor, close to and aiming up at

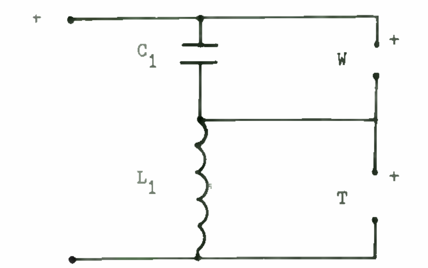


FIGURE 13: Simple series constant-voltage crossover to be used with tweeter A (the CTS 2T2-NFR).

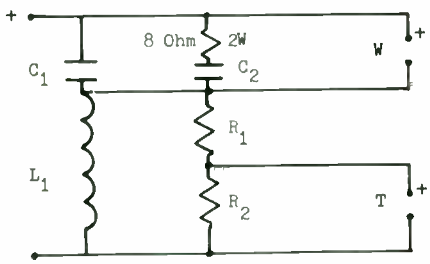


FIGURE 14: Crossover for tweeter B (the Speakerlab DT50). Use with Table 4.

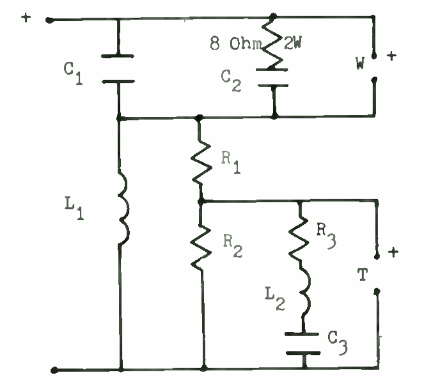


FIGURE 15: Crossovers for tweeters C (the KEF T27 SP1032), D (the Polydax HD13D34E) and E (the Polydax HD13D34H). Use with Table 5.

the walls.[9] Speakers with positive D signals should face the wall to the left of the listening area; those with negative D signals should face the wall to the right. Those speakers with only M signals may face the back wall.

Feed the signals to the amplifier channels from either a delay device or an SQ decoder. The ideal delay device would have an adjustable delay of between 10 and 60 milliseconds,[16] with the same delay applied to both channels. This device should *not* provide any crossfeed between the channels or any "reverberation" of its own. You can set some of the artificial reverberation devices to meet these criteria. Lexicon, Eventide Clockworks, Klark-Teknik, Phoenix Systems, Sound Concepts and Advent, among others, make delay devices that fit the bill. In addition, *TAA* offered a digital delay line in 1/83 (p. 7) and 2/83 (p. 24). With the proper delay setting, you will be transported to the concert hall. You will not hear the individual ambient speakers, but only their effect in opening up your sound environment.

Incidentally, you can also use the speakers as extension speakers if you place them on their backs, facing out from the walls at the listener. If you attach a handle, they make good portable speakers for use in automobiles, and you can carry them with you to avoid theft.

ABOUT THE AUTHOR

Dan McGillicuddy is a consultant in architectural acoustics, psychoacoustics and audio equipment design. He has designed voltage and power amplifiers, power supplies, active filters and other signal processors, and sound-reinforcement systems for auditoriums. He has also studied noise reduction, the control of reverberation, and the perception of many different distortion mechanisms on pure tones, speech and music. He is currently working on a design for an audio image processor.

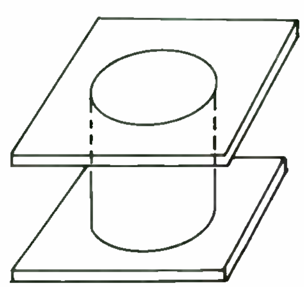


FIGURE 16a: Cutting guide for the inductor form.

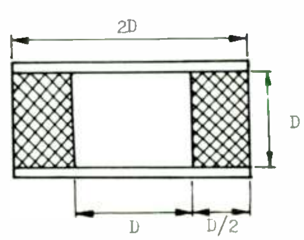


FIGURE 16b: The complete inductor on which you wind the AWG wire.

REFERENCES

1. Beranek, L.L., *Music, Acoustics & Architecture*, John Wiley & Sons, 1962.
2. Bauer, B.B., "A Unified 4-4-4, 4-3-4, 4-2-4 SQ-Compatible System of Recording and FM Broadcasting (USQ)," *JAES* (Volume 27, Number 11), November 1979, p. 866.
3. Schroeder, M.R., "Natural Sounding Artificial Reverberation," *JAES* (Volume 10, Number 3), July 1962, p. 219.
4. Blesser, B. and J.M. Kates, "Digital Processing in Audio Signals," in *Applications of Digital Signal Processing*, A.V. Oppenheim, Ed., Prentice-Hall, 1978.
5. Bauer, B.B., "Stereophonic Earphones and Binaural Loudspeakers," *JAES* (Volume 9, Number 2), April 1961, p. 148.
6. Schroeder, M.F., "Models of Hearing," *Proc. IEEE* (Volume 63, Number 9), September 1975, p. 1332.
7. Cable, C. and R.C. Enerson, "Those Early Late Arrivals! Mr. Haas, What Would You Do?" *JAES* (Volume 28, Number 1/2), January/February 1980, p. 40.
8. Madsen, E.R., "Extraction of Ambience Information from Ordinary Recordings," *JAES* (Volume 18, Number 5), October 1970, p. 490.
9. Clark, D.L. and B.F. Muller, "An Ambience Reproduction Speaker System," *SB* 1/80, p. 12.
10. Small, R.H., "Closed-Box Loudspeaker Systems Part II: Synthesis," *JAES* (Volume 21, Number 1), January/February 1973, p. 11.
11. Small, R.H., "Vented-Box Loudspeaker Systems Part III: Synthesis," *JAES* (Volume 21, Number 7), September 1973, p. 549.
12. Ashley, J.R. and A.L. Kaminsky, "Active and Passive Filters as Loudspeaker Crossover Networks," *JAES* (Volume 19, Number 6), June 1971, p. 494.
13. Small, R.H., "Constant-Voltage Crossover Network Design," *Proc. IREE Australia* (Volume 31, Number 3), March 1970, p. 66.
14. Thiele, A.N., "Air-Cored Inductors for Audio," *Proc. IREE Australia* (Volume 36, Number 10), October 1975, p. 329.
15. Thiele, A.N., "Optimum Passive Loudspeaker Dividing Networks," *Proc. IREE Australia* (Volume 36, Number 7), July 1975, p. 220.
16. von Bekesy, G., "Auditory Backward Inhibition in Concert Halls," *JAES* (Volume 27, Number 10), October 1979, p. 780.

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

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

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

WHY YET ANOTHER COMPUTER MAGAZINE?

Dear Publisher:

I think the *Audio Amateur* is the best hobby magazine published. You should be very proud of your accomplishments over the past decade. I thank you for your effort!

I wish a similar magazine existed for computers. *Microsystems* and *Doctor Dobbs Journal* are very good for software, but there is no magazine devoted to computer hardware. This is very unfortunate. Currently IBM and Intel seem to have a strangle hold on the hobbyist marketplace. It is not very difficult to build a computer which makes the IBM PC look like a 4-function calculator.

We are in a stagnant holding pattern using hardware concepts which are history. The 8088 and 8086 are not much more than stretched 8-bit processors. (Compare the register names and instructions to the 8085 if you doubt this statement.) There are several very interesting 16-bit alternatives which have not received the attention they deserve—the Motorola M68000, the Zilog Z8000 and the National NS16032. Very soon production quantities of 32-bit processors (Motorola's M68020, Zilog's Z80000 and National's NS32032) will be available. These, along with new VLSI coprocessors and peripheral chips, will make it very easy and reasonably inexpensive to build very advanced and useful computers.

If you ever decide, against the odds of all those "me too" microcomputer magazines, to publish the *Analytical Engine Works Amateur*, please let me know. I want a lifetime subscription. One 64-page issue a year of a computer magazine at the level of the *Audio Amateur* would be a 100% improvement over what is currently available.

I look forward to many years of reading your useful, interesting and very enjoyable magazine. Please keep up the outstanding work!

Sincerely,
Ed Scott
Glendale, CA

Tools, Tips & Techniques

Two Tips For Builders

I have two tips for fellow speaker builders. First, when building with particle board (or plywood), use ringed paneling nails. They are tempered so they do not bend the way that finishing nails do. They hold much better in end grain, and they are thinner for their length.

Second, use construction adhesive (panel adhesive will do) in place of glue and caulking. Although it is not as strong as glue, it is strong enough for many applications. It is also fast and easy to use, if you clean your fingers with mineral spirits.

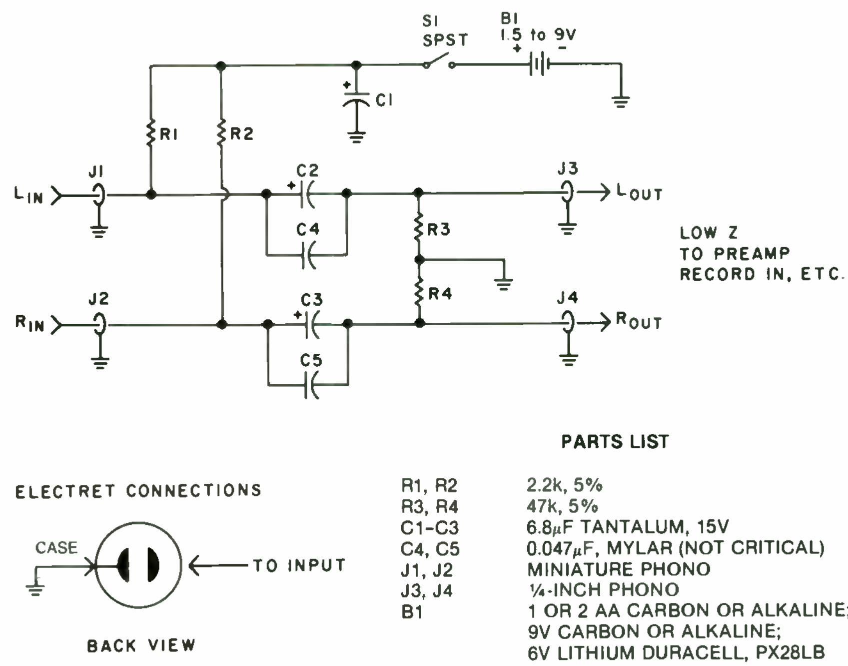
Robert White
Orlando, FL 32806

Economical Electret

During the course of my experimenting with and building speaker systems, I decided I needed a more accurate microphone. I discovered a new Panasonic omnidirectional electret condenser microphone capsule *(Fig. 1)* that offers a flat 20Hz to 16kHz, ±0.5dB response and is off no more than ±1dB at 20kHz. One of the more attractive features is the price—$1.95. (It is available from Digi-Key Corp., Highway 32 South, PO Box 677, Thief River Falls, MN 56701, part #P9932.)

I then built a stereo microphone set *(Fig. 2)* using two of these capsules. When soldering the electret, you must use care and very low heat. *(Most electrets are in a wax base that melts easily.—Ed.)* After connecting two small wires, attach the microphone cable. I used RG-175 coaxial cable. Then attach small connectors to the other end of the wires. For mounting, I used a hollow Plexiglas tube with silicone on the cartridge and the cable *(Fig. 3)*. The less silicone rubber used to hold the cartridge, the better the mechanical isolation will be. You may also use a small, donut-shaped piece of foam to seal the area around and to the rear of the capsule. This eliminates resonance effects. You need not use a wind screen for indoor speaker calibration, but I recommend using a piece of open-cell foam or felt to isolate wind pressure in outdoor use. You may use any tube, but a small front diameter produces the least reflection and refraction.

The power supply is housed in a mini-



PARTS LIST

R1, R2	2.2k, 5%
R3, R4	47k, 5%
C1-C3	6.8µF TANTALUM, 15V
C4, C5	0.047µF, MYLAR (NOT CRITICAL)
J1, J2	MINIATURE PHONO
J3, J4	¼-INCH PHONO
B1	1 OR 2 AA CARBON OR ALKALINE; 9V CARBON OR ALKALINE; 6V LITHIUM DURACELL, PX28LB
METAL ENCLOSURE	

FIGURE 2: Schematic diagram of the author's stereo setup.

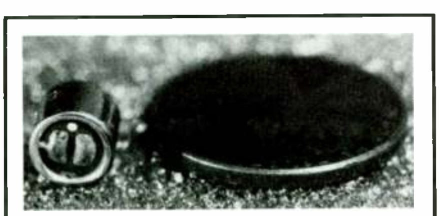


FIGURE 1: Front view of the Panasonic electret capsule, in comparison to a penny.

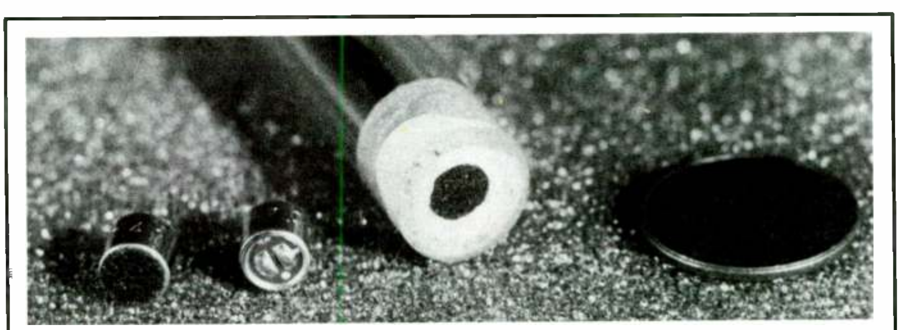


FIGURE 3: Front and rear views of the electret capsule (left) and the electret mounted in the microphone tube with silicone (center), in comparison to a penny.

FIGURE 4: Mr. Szekeres's finished stereo pair.

box with jacks to connect all the cables. For my stereo set *(Fig. 4)*, I used a small (6V) lithium battery, which should be good for a number of years of intermittent use. You may incorporate any number of microphones. Compare two or more mikes by checking for any response variances. To do so, just tape them together and run a frequency sweep test. There should be no variance, except perhaps around 3dB above 15kHz, due to the positioning. The only other consideration is coupling capacitors C2 and C3. The microphone is sensitive to air-pressure changes down to the hertz level, so you may change C2 and C3 for other roll-off frequencies.

Greg Szekeres
Pittsburgh, PA 15236

A Bar Graph Voltmeter

"Why should I have to pay $50 to $100 for a digital multimeter?" I asked myself one day. All I needed was a device to indicate when my function generator was putting out 100mV while using the constant-voltage method for measuring Thiele/Small parameters for drivers.

I already had a digital multimeter, but the constant switching of leads to measure milliamps and millivolts was not only putting wear and tear on the equipment, but it was also time-consuming and a nuisance. After looking for a cheaper alternative in several magazines, I came across a circuit in *Popular Electronics* (Forrest Mims III, September 1978, pp. 92–97) that looked promising *(Fig. 1)*. It is based on the LM339, a quad comparator on a single IC chip, and it uses LEDs for voltage indicators. I changed R1 from a 100k to a 5M pot, as recommended in the article, to allow the circuit to measure down into the millivolt range. Using a volt-ohmmeter, I hand-matched R2–R5 and R6–R9. This might not have been necessary, but I figured it wouldn't hurt. I also added another pot, R10, because I could not get the range I wanted between the LEDs. After some juggling between R1 and R10, I was able to adjust the cir-

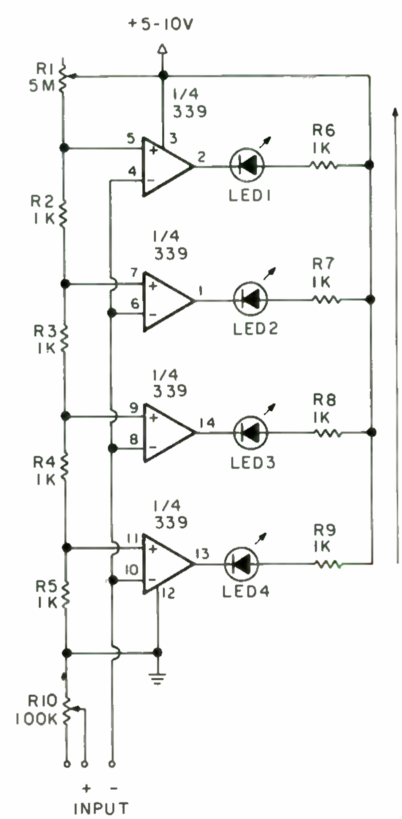


FIGURE 1: The bar graph voltmeter is a cheaper and more practical alternative to buying a digital multimeter.

cuit so that each LED represented 25mV. LED 1 equaled 125mV; LED 2, 100mV; LED 3, 75mV; and LED 4, 50mV.

You can adjust the circuit so that a 100mV input (or whatever standard you want) will turn off LED 1 and leave the remaining three on. Thus LED 2 will equal 100mV. The reason for doing this is that the transition from turning the LED on to off is quicker and more accurate visually than turning it off to on. When the circuit is calibrated and in use, set your signal source so that all the LEDs are illuminated, then back off the input until only the LEDs you want on are left on.

Frequency response of the circuit is flat from 5Hz to 100kHz. Total cost is about $10, not counting the power supply, which you can "steal" from another piece of equipment as I did.

David Long
Dalton, GA 30720

Crossover Check

After designing and building a passive, two-way, parallel crossover network, I like to check its operation with the drivers connected. I apply voltage to the network at selected frequencies, then measure the input voltage and the voltage across the woofer and tweeter. I convert this data to decibel values and plot them on semilog paper. This is not only a tedious business, but it shows only how the individual low-pass and high-pass filters are acting. It does not show how these will combine, considering the *phase* angle.

A quicker way to evaluate the system, so as to simulate its acoustic behavior with perfect speakers, is to measure *across* the hot woofer and tweeter network output terminals, with the speakers connected *(Fig. 1)*. This measurement corresponds to the acoustic sum with the tweeter (or woofer) phase inverted. This connection must be used with second-order networks because the in-phase connection will produce a null (outputs are 180 degrees out of phase) at the crossover frequency. First and third-order networks produce output voltages that are 90 degrees apart (quadrature), so at least insofar as magnitude is concerned, relative driving phasing is not important.

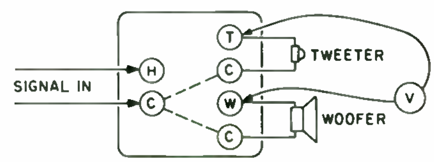


FIGURE 1: One way to evaluate a crossover system is to measure across the hot woofer and tweeter output.

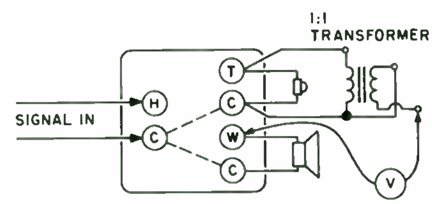


FIGURE 2: You may also fuse a 1:1 audio transformer to measure crossover network performance with the drivers connected in phase.

If for some reason you want to check operation with the drivers connected in phase, reverse the polarity of the low-pass or high-pass section with a 1:1 audio transformer and measure from one transformer's secondary to the hot terminal of the other *(Fig. 2)*. Try a Radio Shack CAT 273-1375 or a Universal Sound PAT.

David J. Meraner
Scotia, NY 12302

SB *Mailbox*

CORRECTION: BULLOCK FORMULAS

Two errors in Part II of my passive crossover series (*SB* 2/85, p. 26) have come to my attention. First, formula 16 (p. 30) is incorrect as it stands. The correct version is as follows:

$$A = \left(R + \frac{1}{R} \right)$$

The computer program listing uses the correct formula.

Second, the equation b = 6.667 in example 1 (p. 37) should read H = 6.667. I apologize for any inconvenience this might have caused.

Robert M. Bullock III
Contributing Editor

CORRECTION: SHADOW LETTER

A small error crept into my letter in *SB* 2/85 (p. 46). A line of text apparently was dropped in the fourth paragraph, beginning "Contrary to the article. . . ." The first sentence of that paragraph should read as follows: "Contrary to the article, power supply decoupling is via polystyrene capacitors; polypropylene caps are used for the filtering functions."

Thanks for the opportunity to clarify this point.

Neil Shattles
Shadow Engineering
Lilburn, GA 30247

T/S PARAMETER INFO WANTED

Can anyone provide me with measured (or calculated) Thiele/Small parameters for the Hartley 218HS low-frequency 18-inch driver? The Hartley Corporation claims that because of the unusual configuration of the magnetic suspension voice coil, they are unable to provide credible Thiele/Small parameters. Thanks in advance for any help.

D. Lee
3863 S.E. Madsen Ct.
Hillsboro, OR 97123

INFINITY INQUIRY

I need help. I own a pair of Infinity Column II speakers, which I would like to upgrade for greater efficiency, wider dispersion and controlled/tighter bass response. Does anyone have such a modification? If so, please write to me at the address below.

Robert Owak
6049 Kimberly Drive
Tinley Park, IL 60477

ADVANCING PASSIVE CROSSOVERS

Until recently, most investigation into three-way crossovers has been aimed at active crossovers. All that has changed, however, with Contributing Editor Bob Bullock's work[1-3] on passive-crossover design.

Although I knew the "on-paper" advantages of first-order crossovers, my individual driver tests usually showed the slow roll-off rate to be unacceptable. I had empirically developed charts for second-order crossovers, so I used them in my speaker design. The math indicated that I should invert the midrange driver, but my ears told me that this was unacceptable on voice and music with major transient content. I was forced, therefore, to "play" with the crossover points to smooth overall performance.

Mr. Bullock's third-order all-pass with positive-polarity bandpass caught my eye because the midrange did not have to be inverted. Last winter, with Mr. Bullock's help, I programmed my computer to design and plot responses for this crossover. The first system I built this spring fit the crossover's requirements (I believe that a ratio of crossover frequencies greater than 4:1 is desirable, i.e., a two-octave spread). Using the computed component values with no tinkering, I have found the performance to be excellent. I am sold on this crossover configuration. For my 800Hz and 5kHz crossover frequencies, the computer predicts only 1.5dB anomolies in the power response of this all-pass crossover. I used the transposed midrange topology[1] for impedance reasons.

Anyone using passive three-way crossovers should consider the third-order all-pass with positive-polarity bandpass and thank Bob Bullock for bringing passive crossovers into the 1980s.

G.R. Koonce
Liverpool, NY 13088

REFERENCES

1. "The Analysis and Synthesis of Passive Constant Power and All-Pass Three-Way Crossovers," *AES Preprint No. 1995 (F-1).*

2. "Passive Three-way All-Pass Crossover Networks," *JAES* (Volume 32, Number 9), September 1984, pp. 626–639.

3. "Passive Crossover Networks: Part II," *SB* 2/85, pp. 26–39.

RADIATION RAMIFICATIONS

I enjoyed Bob Bullock's article on passive crossover networks (*SB* 1/85, p. 13). Before reading the article, I was aware that the radiation patterns of loudspeakers have lobes, but I did not realize the extent to which the crossover network affects the pattern. In the process of casting light on the subject, however, Mr. Bullock's article left a few shadows.

I am intrigued by the D'Appolito satellite system (*SB* 4/84, p. 7). *Figure 5* in Mr. D'Appolito's article shows the difference in radiation patterns between Linkwitz-Riley

(L-R) and odd-order Butterworth filters. But it was not until I read Mr. Bullock's article that things began to make sense.

Looking at Mr. Bullock's *Fig. 12,* I noticed that the listening axis (L) was not centered in the primary lobe. Is this because of the constant 90-degree phase shift inherent in odd-order filters? I took a thin sheet of paper and traced *Fig. 12.* Then I flipped over the paper and placed my tracing over *Fig. 12* with the axes coinciding. This produced the hemispherical pattern shown in *Fig. 17.* But Mr. Bullock says that using a L-R crossover with this configuration will not produce the hemispherical pattern. What pattern will it produce? I am tempted to say the pattern of *Fig. 13,* but surely the presence of the second woofer would have some influence.

My sketch *(Fig. 1)* shows the arrangement of a hypothetical three-way system using two 8-inch woofers wired in parallel and two 4-inch midranges with a 1-inch tweeter in the D'Appolito configuration. The woofers have the same sensitivity as the midranges. The desired woofer/midrange crossover frequency (f_C) is in the 300 to 500Hz range. Even though the physical location of the woofers/midranges does not follow a true D'Appolito configuration, there are similarities. Using the tracing paper again on Mr. Bullock's *Fig. 14* and superimposing its mirror image over the original, it seems I can get a hemispherical radiation pattern at the lower f_C. Is that true? If so, is it due in part to the drivers' physical location or entirely to the fact that the lobes will be wider at a lower frequency? If the physical layout has some effect, will it make any difference whether L-R or odd-order Butterworth filters are used?

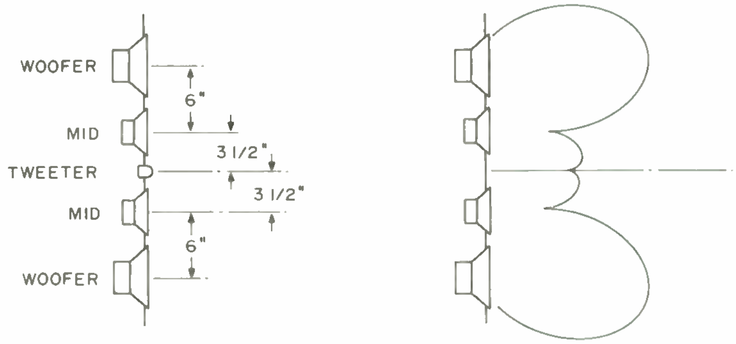


FIGURE 1: Mr. Miller's hypothetical three-way system in the D'Appolito configuration would produce the desired radiation pattern of the woofer/midrange at an f_C of around 300 to 500Hz.

I have several related questions. First, what kind of radiation pattern results when asymmetric filters are used? For example, what happens if you use a midrange rolling off at a rate of – 12dB/octave while the tweeter is rolling (up?) at 18dB/octave? Second, several authors have mentioned growing evidence that even-order filters have "problems" not encountered with odd-order types. Judging by the radiation patterns in Mr. Bullock's *Figs. 12* and *13,* the even-order filter has the advantage of symmetrical radiation. What are these authors talking about?

Finally, I have a question about driver time alignment. As I understand it, the drivers are considered time aligned if their acoustic outputs are in phase and the leading edges of the propagated waves lie in the same plane. In other words, the information from each driver reaches the listener's ear at the same time. Are time aligned and phase coherent the same thing? If my understanding of time alignment is correct or close, then it should not be possible to time align drivers arranged in the D'Appolito configuration and retain the uniform hemispherical radiation pattern. Am I correct?

Answers to these questions will help me sleep at night.

Kenneth P. Miller
Mexico, MO 65265

Mr. Bullock replies:

Your guess about the polar pattern of an L-R crossover in the D'Appolito configuration is essentially correct. The only difference is that the lobes are somewhat narrower.

In general, lobing from two sources occurs when the separation between them is comparable to the wavelength being radiated. In the D'Appolito configuration with an odd-order crossover, the tweeter fills in a pattern that would otherwise have lobes. At very low *crossover frequencies, this action is not necessary because the two-source radiation will already be spherical. Your three-way D'Appolito configuration will have spherical patterns at both crossover frequencies, either because the crossover frequency is low or because fill-in takes place. There is a possibility of lobing between the two crossover frequencies if the separation between the two midrange sources is comparable to the wavelengths being radiated. This is because the other drivers may not fill in enough in this band.*

I have not studied radiation patterns from "asymmetric" crossovers. I imagine the pattern would also be asymmetric. It is true that "even-order filters have 'problems' not encountered with odd-order types," but the converse is also true. The choice of a crossover involves unavoidable trade-offs in performance characteristics. Right now, I prefer even-order crossovers.

Finally, the term "phase coherent" is widely used, but I have never been able to figure out what it means. "Time aligned" appears to mean that two sources are in phase at their common crossover frequency.

HIGH-ORDER CROSSOVERS

Bob Bullock's article in *SB* 1/85 (p. 13) is probably the most useful crossover design article I have seen. Since he solicited reader response, here is mine.

Recently I have noticed that two of the best-sounding new commercial speakers I have heard share a unique design concept. The speakers are the Goldmund Dialogue and the JSE Model 2. They image very well and play cleanly at very loud volumes. Indeed, I think the Goldmund is the best speaker I have heard, next to the Infinity IRS. The unusual design concept involves the use of extremely high-order crossovers; I believe one is 36dB/octave, while the other is 60dB/octave. I have not seen useful design information for this type of crossover, but it makes sense that if you want to play extremely loud music, you must be able to suppress the out-of-band portions by quite a few decibels. Does anyone know of a reference to such designs? If not, I would appreciate a discussion of these crossovers.

Vytenis Babrauskas
Bethesda, MD 20814

Mr. Bullock replies:

I have used a 36dB/octave three-way active crossover network for about three years and have been very pleased with it. I plan to describe how to build it in Part IV of my SB *series on crossover networks. As far as I know, that will become the first published information on the subject.*

SB Mailbox

There are problems with very high-order networks, not the least of which is cost. My active crossover network came to about $300, including power supply and enclosure. A passive version of the same network would require 15 inductors and 15 capacitors, which I suspect would run more than $100, even if I raised the low crossover point an octave or so to reduce inductor sizes. I believe sensitivity problems might also arise with long ladders or many cascaded active sections.

I certainly will look into the idea and see whether I can come up with something useful.

PASSIVE CROSSOVER QUESTIONS

Reading Bob Bullock's article on passive crossover networks in *SB* 1/85 (p. 13) helped clarify a question I have, but I am still confused about the relationship between transient and phase response in two-way passive crossovers. If fourth-order slopes are used, I understand that the woofer and tweeter will be in phase with each other at all frequencies (ideally), but out of phase with the input at frequencies near crossover.

My first question is, does their common phase shift in the crossover region involve differing transient responses so that the woofer and tweeter do not respond simultaneously to a transient input? Second, what are the woofer's and tweeter's responses to the first few cycles of (transient) sine waves at various frequencies? And third, does the "speed" or rise time of the woofer affect its response and integration with the tweeter when crossed over well below the woofer's natural high-frequency roll-off point (and the associated inductive effects)?

Mr. Bullock also discusses all-pass crossovers (APCs) and constant-power crossovers (CPCs). I assume that a flat magnitude response—as provided by an APC—corresponds to a flat sound pressure level (SPL) versus frequency response (and hence a flat on-axis acoustical power response). When would it be desirable to sacrifice this flatness to obtain a more nearly flat power (consumption?) response, as described in the section "Even-Order Compromise" (p. 19)?

Finally, does using two woofers as in Joseph D'Appolito's configuration (*SB* 4/84, p. 7) double your efficiency as well as raising sensitivity 6dB?

Ralph Gonzalez
Philadelphia, PA 19143

Mr. Bullock replies:

I believe Mr. Gonzalez's first two questions amount to asking whether a standard passive two-way crossover network affects the transient response of a loudspeaker system. The answer is yes for all orders except the first, which is a linear-phase crossover and hence is transient perfect. Lipshitz and Vanderkooy have studied extensively the problem of designing high-order linear-phase crossover networks [JAES (Volume 31), January/February 1983, p. 2]. The conclusion I have drawn from their work is that such networks require abundant circuitry and sophisticated design techniques. Even then, the results are not always satisfactory according to other desirable criteria.

I believe that rise time is a relatively unimportant parameter for a woofer, as long as the woofer can trace frequencies somewhat higher than the crossover frequency. I associate a short rise time with a signal's high-frequency components, so it should be a significant parameter for the tweeter.

In response to Mr. Gonzalez's fourth question, if we suppose that V_L and V_H are the output voltages of a two-way crossover network, the APC network satisfies

$$|V_L + V_H| = |V_{in}|$$

and the constant-power network satisfies

$$|V_L|^2 + |V_H|^2 = |V_{in}|^2$$

Condition (1) corresponds to a flat-summed sound-pressure response on-axis, and condition (2) to a flat-summed acoustic-power response on-axis. Conditions (1) and (2) are mathematically equivalent only if V_L and V_H are 90 degrees out of phase or if V_L or V_H is identically zero. Since this latter condition never holds in a two-way system, the only networks that are both APC and CPC are those of odd order—i.e., those with a 90-degree phase difference between their outputs.

I designed the compromise crossover network to allow a trade-off of magnitude flatness for power flatness, or vice-versa, in even-order networks. An even-order APC has a 3dB dip in its power response, and a CPC a 3dB peak in its magnitude response. You could design a compromise crossover to have a PdB dip in its power response for any value of $0<P<3$. Its corresponding magnitude response would then have a peak of $(3-P)$dB.

According to the experts, magnitude response is related to the direct-sound field and power response to the reverberant field. Thus, if you heard only direct sound at your listening position, you would want to use an APC, while if you heard only reverberant sound, you would want a CPC. In reality, the sound field is probably a mixture of both types, so a compromise network is reasonable.

The woofer efficiency in a D'Appolito configuration is twice the efficiency of one of the drivers. The sensitivity of the woofer drivers must be 6dB less than that of the tweeter for correct sensitivity matching if they are connected in parallel. Their sensitivity should equal that of the tweeter if they are in series.

DEVALUATION DOWN UNDER

In the land down under, the falling value of the dollar has really hit hard. Times must be tough when a pair of KEF B139B drivers retails for around $469 (Australian). If you think that is outrageous, then imagine paying $800 (Australian) for a single JBL 2245H 18-inch driver. And that's the discount price; $940 is the full price! Consider yourselves (in the US) fortunate to have such a diverse range of drivers (brands and types) available at such reasonable rates.

Meanwhile, on a lighter vane, and one echoing the appeals of my countryman John Kasowicz (*SB* 2/84, p. 42), is there any corporation or even individual willing to assist fellow enthusiasts in importing specific drivers to Australia (e.g., Janzsen electrostatic tweeter elements)? The range available here is quite restricted. Any help in this regard would be more than well rewarded. Please contact me directly or through *SB*.

Nigel Goodwin
33 Grasmere Crescent
Wheeler Heights
NSW 2098 Australia

TEA & SYMPHONY

While finishing up the last days of my recent holiday in England, I saw an ad for E.J. Jordan, Ltd., in a hi-fi magazine. Noting that the address indicated a mere 30-minute drive, my wife and I managed an unannounced visit with Ted Jordan, his wife and a friendly neighbor. The Jordans were very gracious with tea and biscuits, and we were lucky to audition some of Mr. Jordan's newest aluminum-cone woofers. Some information may be of interest to other *SB* enthusiasts.

Mr. Jordan's current system consists of four 2-inch modules and two 8-inch woofers per channel. The crossover is a passive first-order at 150Hz to separate the woofer and mid/tweeter enclosures. Dynamic range with a Jordan 70WPC amplifier was very good with plenty of transient response and low-end power. A compact disk of the 1812 cannon shots moved the woofer cones approximately 1 inch peak-to-peak with no bottoming or apparent distortion.

The new Jordan woofers were in prototype form in simple enclosures. Although they had stamped frames, the production units will be castings. The cones are driven via microtubes (similar to the smaller modules) with a quoted moving mass of 7 grams. Free-air resonance is about 28Hz with a Q_T of 0.62. Mr. Jordan's enclosures were tuned for unity-Q and had an f_4 near 30Hz.

The new woofers and the Jordan modules will now be distributed in the US by Mr. and Mrs. Brisacher, 301 N. Harrison Street, Princeton, NJ 08540. According to Mr. Jordan, the new distributers are competent and knowledgeable, and pricing should be better than in the past. Mr. Jordan also noted that his modules should be used (or purchased) unmodified—i.e., no ferrofluid. In his opinion, the modifications affect damping characteristics and ruin the system's musicality. The "unmodified" units sounded as good as anything I have heard, with excellent imaging and breadth of sound stage. You can sit anywhere in the room and get full appreciation of the sound.

Matthew Honnert
Carol Stream, IL 60188

DRIVER RESONANT FREQUENCY

In reference to Robert Bullock's article on sixth-order alignments (*SB* 1/82, p. 20), Mr. Bullock describes f_{SB} as the driver resonant frequency in the enclosure (p. 21). How do I find this frequency? Is it different from f_S (free-air resonance), f_B (box frequency) or f_L and f_H (impedance peaks of the driver in a reflex enclosure)? If it is none of the above, does anyone have a simple but accurate method for finding this frequency?

James Rosa
S. Fallsburg, NY 12779

Mr. Bullock replies:

Refer to my article in SB *2/81 (p. 22, second column) for a definition of the in-box resonant frequency, where it is defined in terms of the frequencies f_L, f_H, f_M of impedance maxima and minimum, respectively. It is usually quite close to f_S, but somewhat lower. The relevant formula is as follows:*

$$f_{SB} = \frac{f_L f_H}{f_M}$$

I hope this explanation is satisfactory.

TL SUBWOOFER MODIFICATIONS

Craig Cushing's article in *SB* 1/85 ("A Compact Transmission-Line Subwoofer," p. 6) is of great interest to me. I am considering building it, but I could use some clarification on several points.

Mr. Cushing lists only nine alternative 8-inch woofers as candidates for this proj-

SB Mailbox

ect. I was surprised to see the Dynaudio 21W54 missing from the list. Most of what I have heard and read suggests that this is one of the finest 8-inch woofers available. Would this driver be a good choice for this application? And if not, why not?

Mr. Cushing's only disclaimer for the project was a potential power-handling limitation. Since my preferred listening level is higher than most, I was considering using a 10-inch driver. Would this subwoofer design accept a 10-inch driver, or would it have to be scaled up? Stated more technically, are the dimensions of a transmission line related to the wavelength of the desired low-end response limit or to the dimensions of the driver?

Bob Davis
Belle Mead, NJ 08502

Mr. Cushing replies:

I am really pleased that Mr. Davis plans to build a version of my subwoofer design. I had no specifications on the Dynaudio 21W54 at the time I wrote the article, so I did not include it in the list of prospective driver candidates. I have since listened to a 21W54 in a replica of the LFT MK-IV, and if I were choosing one 8-inch driver now, I would probably choose the Dynaudio, based on performance and regardless of price. Sound and power handling are excellent, and I think Mr. Davis would be happy with it, even with the high listening levels he likes.

As for power-handling limitations, the cannon shots on the Telarc 1812 Overture *on compact disk do not readily bottom the Speakerlab W848P, even at hefty volumes. When I play the analog version on my turntable, I raise the cartridge tracking force from*

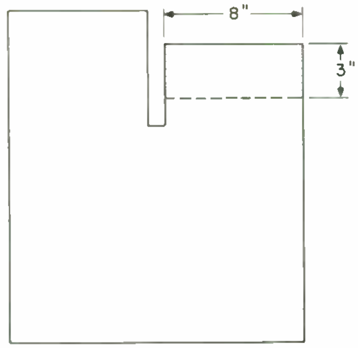


FIGURE 1: To use a KEF B139 or to implement my most recent design, remove a 3-by-8-inch piece from center divider I.

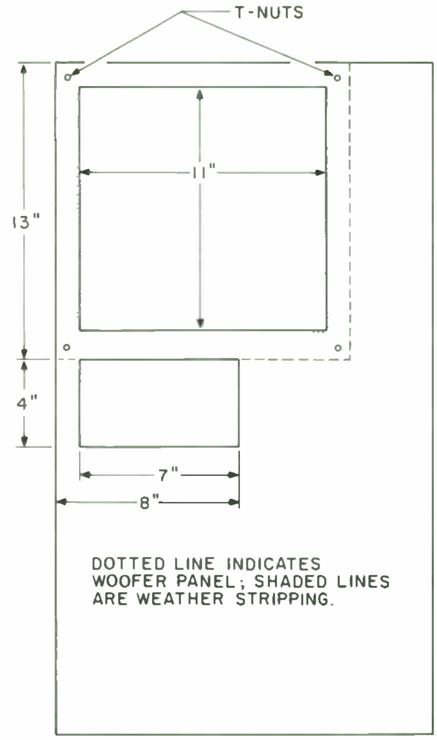


FIGURE 2: Mr. Cushing's redesign calls for a 90-degree clockwise rotation of the entire divider complex, including all cleats and angle baffles.

1 to 1¾ grams; the "bottoming" problem goes away unless I increase the volume to amp-clipping levels.

You need not enlarge the box for a 10-inch woofer. The line cross-sectional area is sufficiently large to allow use of a 10-inch driver, with modifications that I will outline later. The length of the line determines the f_3 cutoff, regardless of woofer size, as long as the woofer's f_S is as low as, or lower than, the desired cutoff point.

A friend of mine has slightly modified his LFT to use a KEF B139 woofer. He simply removed a piece 3 inches high by 8 inches wide from center divider I to the right of the notch (Fig. 1). Then he located the shelf (N) and line terminus 3 inches lower in the front baffle (D). He partly compensated for the loss of line length by reducing the vertical dimension of the line terminus from 6 to 4 inches. The enlarged woofer chamber nicely accommodates the KEF when it is mounted with the long dimension vertical. Of course, you would have to recalculate stuffing amounts.

This combination, to my ears at least, sounds smooth and powerful, with very solid bass in the bottom octave. If you want to take advantage of the 23Hz resonance of the B139, you can stretch the vertical dimensions of the cabinet and dividers 5 or 6 inches, but you will need more particle board than the one sheet noted in the article, and you will not be able to experiment with higher-resonance woofers. I am an inveterate tinkerer, so switching drivers easily is a plus for me, and I have recently redesigned the subwoofer to facilitate this.

Basically, the redesign (Fig. 2) calls for a 90-degree clockwise rotation of the entire interior divider complex, including all cleats and angle baffles. Cut center divider I as per Fig. 1, then make an 11-inch-square opening in the upper left of the front baffle panel (D), allowing a 1-inch border. Make a 4-by-7-inch terminus cutout below it, then glue ⅜-inch-square weather stripping around the outside face of the woofer opening. For woofer baffles, I simply use 13-inch-square pieces of particle board with woofer mounting holes sized as needed. T-nuts at the corners of the box opening and the baffle allow mounting and compression sealing against the weather stripping. I have also installed a 5-by-13-inch angle baffle vertically at the left rear of the woofer chamber (when viewed from the front).

With this mounting system, I have been able to experiment with a 10-inch Audax HD24B45 woofer, the aforementioned KEF and my present setup, which consists of four 5-inch Audax HD13B25H4C12s wired in series/parallel. The 10-inch Audax provides nicely detailed deep bass, as does the KEF, but the small ganged Audaxes seem to give the best combination of bass extension, sonic detail and power handling.

When you get ready to install stuffing in the line, you might try rolling the stuffing in nylon netting, which is available in fabric stores for less than 75 cents a yard (about one-fifth the cost of net curtain material). It is stiffer than the curtain material and slides readily into place.

Classified Advertising

TRADE

CUSTOM MADE POLYACOUSTIC foam speaker grilles. Any size to 48" x 120", thicknesses ¼", ½", ¾", 1", 1½", 2". Colors black, brown, blue, red, burnt orange, silver, gold, or ?. Any design or quantity. SPECIAL '85 *SPEAKER BUILDER* PRICES: price per grille (quantities to four)—multiply size in square feet times thickness times: 3.0 for plain grilles with 90° edges (minimum charge $8), 3.5 for 45° beveled edges (minimum $9), 4.0 for striped or checkerboard designs (minimum $12). Price includes Velcro tape and shipping UPS. Call or write for quantity discounts over four. Check, money order or bank card accepted. Dealer, manufacturer and rep inquiries invited. CUSTOM SPEAKER ENCLOSURES AND PARTS—call or write for quotations. Send 50¢ (refundable) for literature and foam sample. ***CUSTOM SOUND***, 8460 Marsh Rd., Algonac, MI 48001, (313) 794-5746, Monday–Friday to 7 p.m. T3/85

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

GOLD LION "INDIA" TELEFUNKEN tubes. Cramolin, Fidelity Research, Furman Sound, Hartley, Kimber Kable, PS Audio, Precision Fidelity, Sheffield, Sonex, Superphon, V.P.I. Visa, MasterCard, American Express. ***VECTOR ELECTRONICS,*** 1653 SE Marion, Portland, OR 97202, (503) 233-2603. T4/85

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

"YOUR CATALOG IS the best I've read," writes R. Bullock, contributing editor, *Speaker Builder*. More than 30 proven designs for home, car, subwoofer and professional applications. JBL, Audax, EV, SEAS, Hafler, polypropylene drivers and crossovers. Thousands of parts in stock! Send $2 to ***GOLD SOUND***, Box 141SB, Englewood, CO 80151. T4/85

"I RECENTLY HAD the opportunity to use a pair of Celestion SL600s, and the Vanguards proved to be superior in every respect for less than a third the cost. They handle incredible power without strain and play incredibly loud for any small system, let alone a two-way. They are an excellent value that would sell for much more at a retail audio dealer." Steve M., Pennsylvania. Steve is one of many satisfied owners of an ***AUDIO CONCEPTS*** loudspeaker kit. The Vanguards at under $400/pair including cabinets, are just one of the twelve outstanding kits that we sell. Of course we also stock a full line of drivers, cabinets, crossovers and accessories. 40 page catalog, $2. 1631 Caledonia St., LaCrosse, WI 54602, (608) 781-2110.T3/85

Madisound Speaker Components

8982 Table Bluff Road
Box 4283
Madison, Wisconsin 53711
(608) 767-2673

Audax, Dynaudio, KEF, Philips Loudspeakers

GOLD PLATED phono jacks, $1.20; plugs 90¢ each, $1.00 handling. Also custom 1% capacitors, 0.5% resistors. Details, SASE. ***REFERENCE AUDIO***, Box 368M, Rindge, NH 03461. T2/85

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

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

Classified Advertising

FIBERGLASS MIDRANGE TRACTRIX horn by Edgar-Rowe, smooth response 400Hz to 4kHz, 50W, 100dB SPL. Very clean and open sound. Great imaging, with cone driver, $100 plus shipping. California residents add 6½ percent sales tax. ***BRUCE EDGAR***, Box 1515, Redondo Beach, CA 90278. T3/85

EXCESS STOCK. Audio manufacturer liquidating unused wire, connectors, transformers, resistors, capacitors, semiconductors, heat-shrink tubing, multicables and mixing consoles at wholesale prices! Call for a complete list. ***CREST AUDIO***, 150 Florence Ave., Hawthorne, NJ 07506, (201) 423-1300. T3/85

SPEAKER ENCLOSURES, stock sizes or custom made, wood veneers, plastic laminates, grille cloth, Peerless and SEAS drivers. Call, write or visit: ***SOUND CABINETS***, 3 Tasker St., Trenton, NJ 08638, (609) 882-3069. T1/86

CLUBS

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

CENTRAL FLORIDA AUDIO SOCIETY meets monthly in Orlando. Come and meet others who share your interest in music reproduction as we audition equipment and recordings or discuss audio related topics. Contact Ron Deak, 2404 S. Conway Rd., #162, Orlando, FL 32806, (305) 894-6784.

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., Palo Alto, CA 94303 for newsletter.

THE AUDIO SOCIETY OF HONOLULU cordially invites you to attend one of our monthly meetings and meet others like yourself who are interested in the how's and why's 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.

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, WA. Be our guest, write Box 435, Mercer Island, WA 98040 or call Bob McDonald (206) 232-8130.

THE BOSTON AUDIO SOCIETY INVITES you to join and receive the monthly B.A.S. SPEAKER with reviews, debates, scientific analyses, summaries of lectures by major engineers. The BAS was the first to publish info on TIM, effects of capacitors, tonearm damping, tuner IM distortion, Holman's and Carver's designs, etc. Sample issue $1, subscription $16/yr. PO Box 7, Boston, MA 02215.

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

LONG ISLAND AUDIOPHILES: The Audio Syndrome is a Nassau/Suffolk county club dedicated to the pursuit of sonic excellence. Monthly meetings. Fred Masters, (516) 589-4260 or (516) 271-4408.

A CLUB FOR FM AND TV DXers, offering antenna, equipment and technique discussions, plus updates from FCC on new station data. Monthly publication "VHF—UHF Digest." Annual convention in August. For more info: Worldwide TV-FM DX Association, PO Box 97, Calumet City, IL 60409.

THE VANCOUVER AUDIO SOCIETY publishes a monthly newsletter with technical articles, humor and news of interest to those who share our disease. We have 50 members and meet twice every month. Dues are $15/year, which includes 12 newsletters ($15 US outside Canada). Write to the VAS c/o the Secretary-Treasurer, Box 4265, Vancouver, BC, Canada V6B 3Z7. We would like to be put on your mailing list.

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. Membership fee is $10 per year. For more information, send SASE to: CAS, 4506 Osceola St., Denver, CO 80212.

AUDIOPHILES in the Riverside-San Bernadino areas to form an audio club. Contact Frank Manrique, 1219 Fulbright Ave., Redlands, CA 92373, (714) 793-9209.

AUDIOPHILES interested in starting Yugoslavian audio society contact JAS, Likob Matjaz, Stefanovaga, 61000 Ljubliana, Yugoslavia.

SERIOUS AUDIOPHILES interested in a central Colorado group (Denver, Boulder, Ft. Collins, Greeley area) contact James S. Upton, 2631 17th Ave., Greeley, CO 80631.

AUDIOPHILES in the New Jersey and Staten Island areas interested in forming a club please contact Joe at PO Box 1588, Staten Island, NY 10314, (212) 698-2280.

SAINT LOUIS AUDIO SOCIETY meets monthly for discussion and equipment audition. For information sheet send a stamped, self-addressed envelope to SLAS, 7435 Cornell, Saint Louis, MO 63130.

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, recording studio visits, design analyses, digital audio, AB listening tests, equipment clinics, annual picnic and audio fun. The club journal is *LC, The SMWTMS Network*. Corresponding members' subscriptions $6 per year. Call (313) 477-6502 (days) or write SMWTMS, PO Box 1464, Berkley, MI 48072-0464.

ORGAN MUSIC ENTHUSIASTS: If live recordings of fine Theatre Organ Music are your thing, SFORZANDO has room for a few new members. We lend you the music on reels or cassettes. All operation is via the mail. SFORZANDO, c/o E.A. Rawlings, 5411 Bocage St., Montreal, Canada H4J 1A2.

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 ESOTERIC AUDIO RATING SOCIETY (usually known as EARS) is San Antonio's premier audio club. Its members consist of audiophiles and music lovers who share a mutual interest in enhancing their enjoyment of recorded music. EARS meets bimonthly and has been fortunate to offer interesting presentations on audio, recordings and music. The club also has an on-going project of recording local concert activities for radio broadcasts or other purposes. Additionally, EARS is currently trying to launch a club newsletter. Anyone interested in finding out more about EARS should write to the following address to obtain information on the next meeting date and location. EARS, PO Box 27621, San Antonio, TX 78227.

THE NEW YORK AUDIO SOCIETY meets monthly with prominent guest speakers, discussions and demonstrations of the latest equipment. Its $20 annual membership dues includes a subscription to S/N, the society's quarterly publication. For a free invitation to our next meeting, call (212) 544-1222, (212) 289-2788 or (201) 647-2788 or write us at PO Box 125, Whitestone, NY 11357.

SARASOTA AUDIOPHILES interested in forming a club—write to Mark Woodruff, 5700 N. Tamiami, Box 539, Sarasota, FL 33580.

AUDIOPHILES INTERESTED IN FORMING an audio club in the Washington, D.C. area please contact: Joseph Kmetz, PO Box 914, Lanham, MD 20706 or call days (301) 794-7296, eves. (301) 585-3186.

MINNESOTA AUDIO SOCIETY. Monthly programs, newsletter, special events, yearly equipment sale. Write Audio Society of Minnesota, PO Box 3341, Traffic Station, Minneapolis, MN 55402.

NEW JERSEY AUDIO SOCIETY meets monthly with the emphasis on construction and modification of electronics and speakers. Individuals at any level of electronics expertise are invited to join. Contact Bill Donnally, (201) 334-9412 or Bob Young, (201) 381-6269, 116 Cleveland Ave., Colonia, NJ 07067.

TORONTO AREA AUDIO SOCIETY formed. Serious audiophiles contact Neelam Makhija (416) 842-2606 or John Sloan (416) 532-4387.

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 at (404) 253-6419.

WASHINGTON AREA AUDIO SOCIETY and CONSTRUCTORS CLUB (DC, MD, N. VA) in the process of formation. If interested, please contact Horace J. Vignale, 1540 Northgate Sq., #31B, Reston, VA 22090.

HI-FI CLUB OF CAPE TOWN, South Africa, issues newsletter monthly for its members and subscribers. Since our audio problems are the same as yours, we'd like to hear from you. Send 2 I.R.C.s for next newsletter ($16/year) to PO Box 6685, Roggebaai 8012 South Africa.

WOULD LIKE TO CONTACT experienced home builders of audio equipment in the southern Ontario area. I have built the Marsh preamp and head amp...they sound great. T. B. Palmer-Benson, RR 1, Ariss, Ontario N0B 1B0, (519) 837-3964.

FOR SALE

Fujitech A502 preamp kit and two A501 power amp kits. $600. Joe, (503) 629-5948 or (503) 649-9482.

Vintage JBL speaker components: C34 mahogany bass corner horn, $200; 150-4C 15″ woofer, $140; 175 DLH mid-horn, $130; D208 8″ mid/woofer, $36; D280 8″ mid/woofer, $36; N600 crossover, $60; N7000 crossover, $27; Altec 601-A duplex, $100. Ron Bauman, (202) 965-5320.

Nikko Alpha III power amp, $225; NAD 2155 power amp, $165; Dyna ST-70 with mods, $90; BIGG model S-400 EL-84 power amp, $25; EICO HF12 preamp, $20; PMC switching power supply, 5V, 20A, $35; Dynaudio 30W54 12″ woofer, low hours, $50; SWTPC .01 amp, chassis, power supply and circuit board, $25; Dynaco Mk III, $10; Sonus Blue cartridge, $5. Floyd Andrews, (415) 552-7807 days.

SME series 2 improved tonearm with removable headshell. Reference arm of *High Fidelity* magazine. Like new in original box, $50. Matthew, (312) 260-1628.

Dynaco ST-150 power amp, bench checked 90W RMS/channel, with manual, $150; STACO 260W panelmount Variac, new in box, $22; boxer fans 3″ or 5″, $8 each; Dynaco Quadaptor, funky cosmetics but works, $12; power supply filter caps for Dynaco FM-3, $8 each. Ben Poehland, 14 Carol Lane, Malvern, PA 19355, (215) 644-3677.

Series 20-M25, $375; Hapi 2-597, $200; Met-7, $195; 9 ft. trans. subwoofers with KEF 139, $150 each, $275 pair; Soundcraftsman SP 4001 preamp/equalizer, $295; dbx 400, $195; dbx 228, $295; Decoursey Q-8, $100; custom poly electronic crossover, $195; Pioneer TV tuner, $95; 50m roll Mogami speaker wire, $220; PAS-3, $50; Hafler 220, $295. WANTED: Strathearn ribbons. Joe, (206) 588-1417, (206) 474-9068.

Thorens TD126 Mk 3/SME 3009 S3, both in excellent condition, $375/$100. Gary, PO Box 5883, Roswell, NM 88201, (505) 625-1830.

Fisher X-100-B master control amplifier ready for POOGing by a tube freak, includes wood cabinet, operating manual, delivery, but no schematic, $55. R. Jensen, 8 Heather Lane, Reading, PA 19601.

Creek 4040 integrated amp, $175; mint Sanyo Plus C55 preamp, $140; biamp-stereo rackmount electronic crossover, two-way adjust, 100Hz to 10kHz, phase control, $200; *TAA's* Rod Cooper tangential tonearm kit, $90; Kmal arms, $60. Will trade for good three-way electronic crossover, 400Hz and 4500Hz or close, Dyna SE-10 equalizer. David Landrie, 1714 E. 56th, Tacoma, WA 98404, (206) 472-9930.

Classified Advertising

Recently purchased Superphon preamp, $290; Audax HD12.9D25 tweeters, $12 pair. WANTED: Hafler DH-200 or DH-220 in good condition. Rick Kuroda, 112 S. School St., #410, Honolulu, HI 96813.

Harman Kardon Citation 2, new tubes, no cage, excellent, $350; Magnavox FD1000, $200; Dynaco 410, Rollins mod, $350; Sherwood 6040CP, 100W/channel, dual mono, MOSFET, $250; Dynaco 70, stock with WonderCaps, $120; Fisher 500C receiver, $125; Hitachi HCA8500 Mk 2 preamp, a sleeper, $250; Pioneer SF850 electronic crossover, $175; Dynaco FM3, $50. Roger Artman, Box 2463, Grass Valley, CA 95945, (916) 272-8118.

Used Mallory CGS263UO75X5R 26,000μF 75V–95V surge caps, $10 each, 12 for $100; two Technics EAS-10th400A leaf tweeters, $50 per pair; four Morel MDM-75 3″ dome mids, new, $32 each. Jim Katchis, 3016 NE Oregon St., Portland, OR 97232.

Gold Sound 8″ black polypropylene woofers, dual voice coil, $50 pair; Morel MDT-27 tweeters, $25 pair. All barely used. Dan Greene, 216 Auburn Ave., Santa Cruz, CA 95060.

Pristine Yamaha NS1000M loudspeakers, $750 pair; pair Southwest Technical Products Tigersaurus 250W mono amps, great for subwoofers, $325; Advent 100A dual process Dolby unit with a walnut case, $150; dB Systems pre-preamp and power supply, $115. Michael Marks, 81 Grove St., West Roxbury, MA 02132, (617) 327-6784.

Famous British Dalesford D50/200 midbass Bextrene 8″ driver, better made than KEF B200, 3 lb. magnetic construction, never mounted, $29 each; Dynavector DV23RS (MR), unused, $149. S. Sun, Box 6312, Long Island City, NY 11106, (718) 784-2939 evenings and weekends.

Pair of EV Patrician 800s, $4500. FOB Louisville, KY, (502) 895- 3501.

ADS model 10-01 digital time delay system, rack-mount, original carton, $300; Sansui AU-717 amp (for use with model 10-01), $150; Walker CJ55 turntable, Linn LVX arm, Basik cartridge, dust cover, $350; McIntosh MC-225 amp, retubed, $250; Marantz 120 FM tuner, integral oscilloscope tuning, $150. Patrick, (916) 885-3163.

McIntosh MC-75 amp, MC-60, trade either for other model or sell. Welzer ACE (with electronics) dual-triode 350W tube amp, excellent; GSI ST-70 input boards (latest); pair Altec/Ampex 6516 60W theater amps; pair Dukane super-fi 60W tube amps; IR2200 dynamics restorer, new. Dennis Fraker, (406) 222-7404 evenings or early mornings.

ARC SP-6B, mint, $595; Counterpoint SA-2, mint, silver, $545; classic ARC D75, $475; Citation II, 60/60W, very clean, $175; Oracle Alexandria, as new, $545; Citation 12 MOSFET, $125; EICO HF30 amp, excellent, $40. John, (916) 791-2153.

Audio Research SP-5 preamp, excellent condition, asking $450; Watson model 10 speakers designed by Mike Wright, asking $900; Mitch Cotter Mk II moving-coil transformer, $200; Fulton gold speaker cables, 14′, $40; 57″, $18. Al Halstead, (607) 565-8438.

Pair Rauna Tyr loudspeakers. Concrete satellites using SEAS 6.5 and 1″ drivers (see *Stereophile*, Volume 7, Number 4). Excellent condition. Paid $450 in 1984, sell $375. I will ship. Pair Dynaco A-30s and EV 635A microphone, make offer. Harlan, Box 61171, Jacksonville, FL 32236-1171, (904) 388-4483 evenings or weekends.

Heathkit SM-4100 frequency counter, $65; Heathkit 1M-2202 digital multimeter, $115. Call Sam, (206) 367-6548 after 6 p.m.

Phase Linear DRS 400 amp, P3600 preamp, T5200 tuner, 400W/channel peak, 6 months old, with boxes and balance three-year warranty, list $1595, sell $650. Phase Linear 8000A turntable, less than 100 hours, list $650, sell $325. Phase Linear model 1000 NR unit, $150; Stax SR-44 electret headphones, $80. Call John, (904) 641-5830 evenings.

Bogen CHS-100A, 100W PA amplifier, $100; Citation Twelve power amplifier, working, $150; Soundcraftsmen 20-12 equalizer, $150; four KEF B139 woofers, matched FAR, mint, $150/pair; four Sanders transmission line enclosures, braced and damped, no cutouts, $200 pair. Canadian prices. Nick Mastrobuono, 50 Copperfield, Sarnia, Ontario, Canada N7S 5K8, (519) 336-9160.

WANTED

Acoustics book by L.L. Beranek (McGraw-Hill, New York, 1954). Please contact Frank L. Pyle, Pyle Industries, 501 Center St., Huntington, IN 46750.

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

I own a pair of original Advent speakers and would like to know what a good replacement would be for the low-range drivers and/or does anyone repair the original drivers? David Lucas, 924 Hulton Rd., Oakmont, PA 15139.

Horn plans for the following: Carver 960, 980, 1330; JBL 4550Bka, 4560 Bka, 4520, 4530; EV, SH1502; GLI Model 3, DB-50; Altec 828C, 815A, 817, 816A, 210; Klipsch, Belle, La Scala; Sunn 9115, 9215, 9118, 9218; Cerwin Vega D-32, B-36A, V35B, V30X, V31X. Will pay $5 per plan. Al, (212) 569-8858 after 6 p.m.

Can anyone provide information or experimental data on the Thiele/Small parameters for the Hartley Model 218HS low-frequency driver? D.H. Lee, 3863 SE Madsen Court, Hillsboro, OR 97123.

Schematics or technical manual for my H.H. Scott 477 receiver. Will copy and return or pay for same. Please write or call. Jimmy Poole, 8 Wedgewood Dr., Russellville, AR 72801, (501) 968-3981.

Marantz and McIntosh tube amps/preamps; JBL Hartsfield, Paragon; EV Patrician; Row speaker components. Call with anything of interest. Charles Dripps, 4331 Maxson Rd., El Monte, CA 91732, (818) 444-7079.

Advertising Index

McGEE RADIO HOUSE of SPEAKERS

THE LARGEST MAIL ORDER SUPPLIER OF WHOLESALE RAW SPEAKERS IN THE COUNTRY

FOR OUR ALL *NEW FREE* CATALOG, CLIP OUT COUPON BELOW AND RETURN TO US.

McGEE RADIO & ELECTRONIC CORP.
ATTN: SPEAKER DEPT.
1901 McGEE
KANSAS CITY, MO. 64108-1891

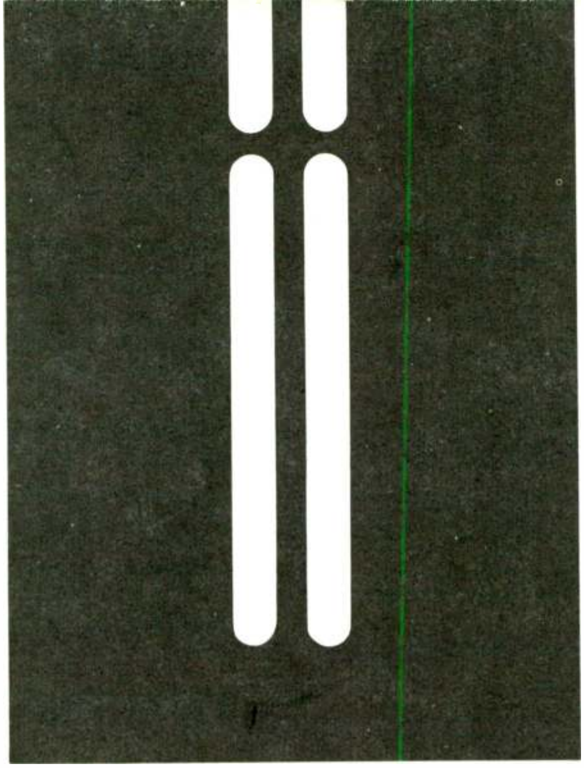
FREE CATALOG COUPON

Name ______________________________

Address ______________________________

City/State/Zip ______________________________

FAST REPLY #HH44



GOLD 3.0 SATELLITES

The Gold 3.0 ribbon satellite represents the most subtle refinement of a reference quality audiophile loudspeaker. It is a phase coherent line source capable of handling the most complex musical transients effortlessly and without coloration. Why use 99.9999% gold? Experiments with aluminum, titanium, platinum (to name a few), as voice coil materials did not yield the very natural, transparent quality of gold. Though good test specifications could be obtained with many different metals, pure gold was unanimously chosen after countless hours of listening tests. Scientific specifications only approximate the effectiveness of any high end audio component. The consummate qualities of the GOLD 3.0 satellite become apparent in a loudspeaker's most exacting analysis—the subjective listening test.

THE BEAUTY OF GOLD IN SOUND

Sensitivity
load dependent

Max. Distortion
.3%

Power Handling
20 amps maximum/side

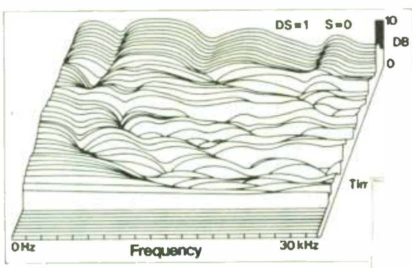
Load
+/− .1 ohms 2 or 4 ohms

Polar Response
> 30 degrees

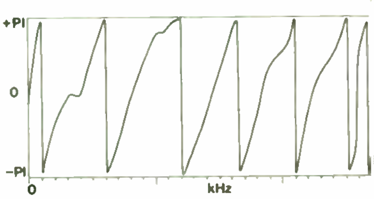
Freq. Response
200 −30 khz

Nextel Suede Coatings
Grey b21; Black c10; White a10.

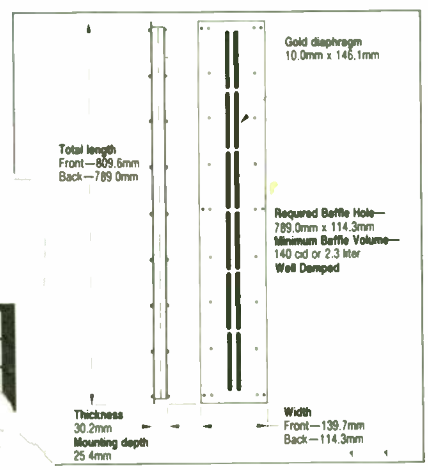
GOLD RIBBON CONCEPTS
211 East 11th Street
Iowa City, Coralville, IA, 52241, USA
(319) 351-9144/1-800-841-GOLD



GOLD 3.0
Time Delay
Spectrum Analysis



GOLD 3.0 Phase Response — Graphic Plot Delay



FAST REPLY #HH915