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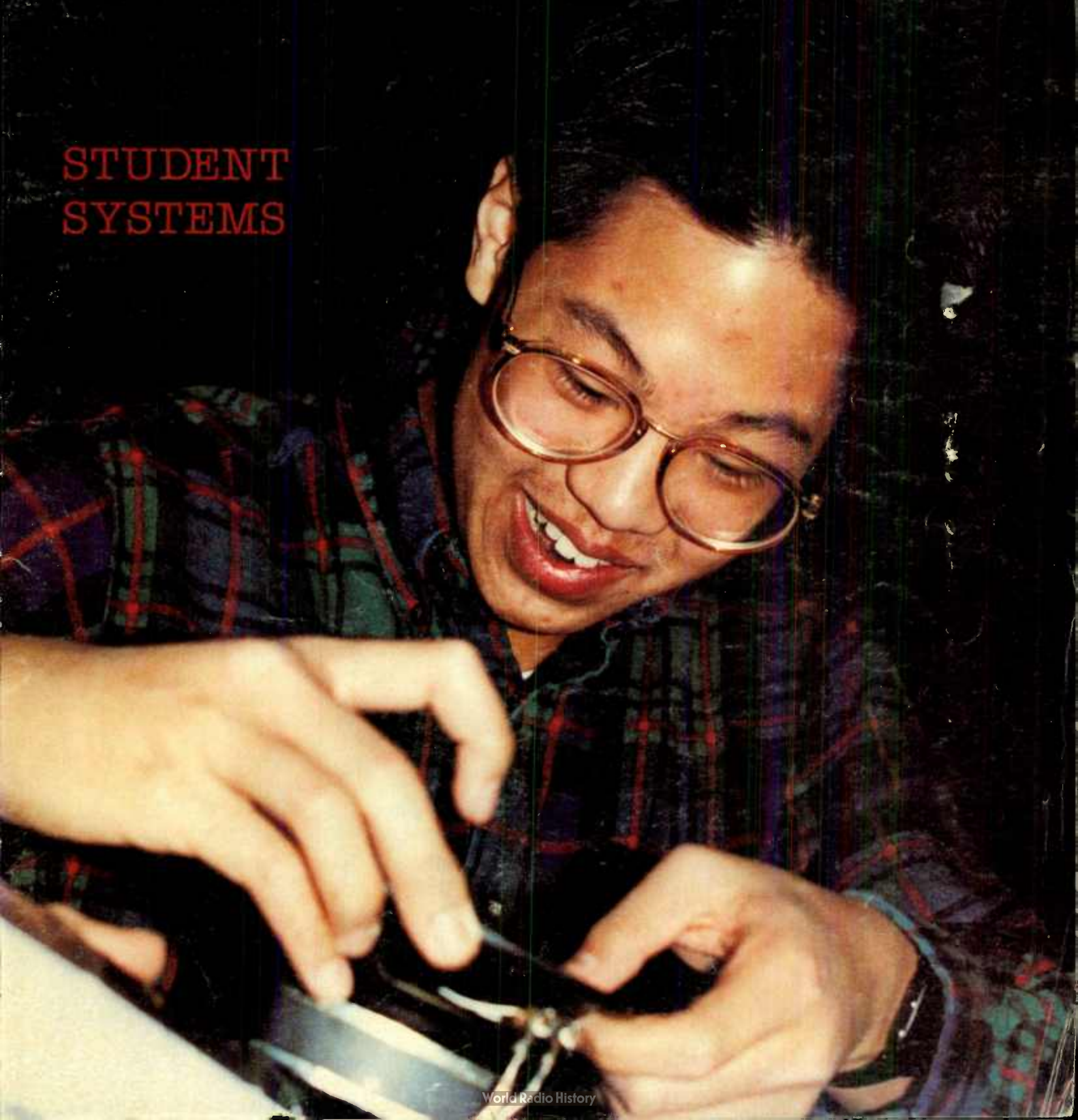
Speaker Builder

THE LOUDSPEAKER JOURNAL

STUDENT SYSTEMS



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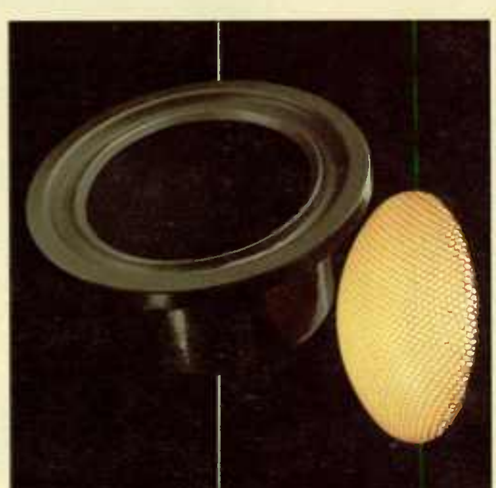

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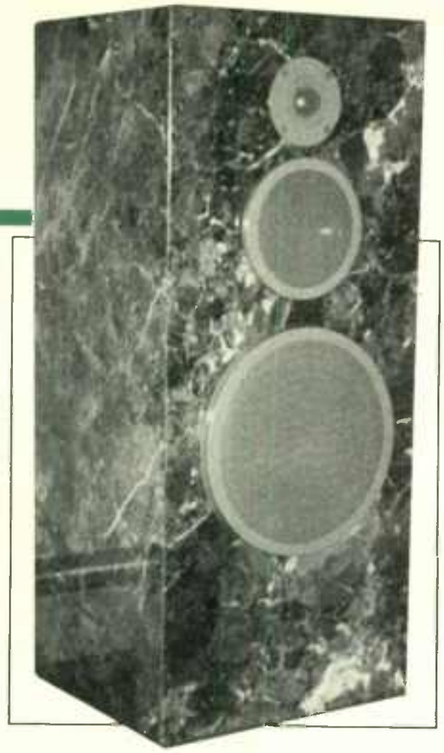
Good News



STONECRAFT SPEAKER SYSTEMS builds marble and granite speaker enclosures. They are available complete using all Focal drivers or without components. Each enclosure is handcrafted with ¾" stone cut at a 45° angle. All electrical connections are silver-soldered using 16-gauge polarized copper-stranded wire. Binding posts are gold-plated through the marble, then silver-soldered to the copper wire. The midrange is housed in its own enclosure within the cabinet.

For more information, contact Stonecraft Speaker Systems Co., 8690 SW 10th St., Pembroke Pines, FL 33025, (305) 436-8582.

Fast Reply #EF253

SOLEN ELECTRONIQUE imports speaker driver units from Davis Acoustics. These units feature carbon fiber, Kevlar fiber, Aramid fiberglass, and a high-power magnet. They are colorationless and have low distortion and dynamic high-power handling.

For more information, contact Solen Electronique Inc., 4470 Thibault Ave., St-Hubert QC J3Y 7T9, Canada, (514) 656-2759, FAX (514) 443-4949.

Fast Reply #EF197

A number of US companies, including **CRYSTALLUME** of Menlo Park, California, are finding uses for diamond thin films: x-ray windows for scanning electron microscopes, diamond heat sinks for high-power electronics, and possibly wear-resistant tools. Also, Japanese electronics companies are planning to manufacture a line of diamond-coated loudspeaker tweeters.

PRECISE ACOUSTIC LABORATORIES has unveiled two additions to their BETA line. The 200BL and the 300BL bookshelf systems join the floor-standing 400BL and 600BL speakers. The 200BL is a two-way system consisting of a 6½" woofer and a 1" soft dome tweeter. The system has a sophisticated crossover network, computer-grade components, gold-plated binding posts, and double-cabinet construction.

The 300BL is a two-way system that uses a newly designed 8" woofer and a 1" soft dome tweeter. The ¾" MDF cabinet includes a crossover with gold-plated binding posts.

Cabinet design integrates the front baffle with the speaker's stand, which can be removed. Cabinet finishes include black satin or hand-rubbed and oiled North American walnut. Special-order custom combinations and finishes are available as are handmade grilles in ebony or graphite.

Suggested retail price is $500/pair for the 200BLs and $700/pair for the 300BLs. For more information, contact Precise Acoustic Laboratories, Suite B, 200 Williams Dr., Ramsey, NJ 07446, (201) 934-1335.

Fast Reply #EF103

The Director™ from **AUDIO CONTROL** is a speaker switcher and system controller capable of handling two different sources (like a TV and a stereo system) at the same time. It is a full-sized unit, and other items in a stereo system, like a compact disc player, can be stacked on top. Since it uses speaker wires and speaker level, it is easy to install and can run all speaker systems in the house on one amplifier. The Director contains automatic impedance protection to safeguard the amplifier from problems of connecting too many speakers.

The Director controls up to six zones of speakers. Five are dedicated to one source (like the stereo system); the main zone may be playing the same music or a different source (like the TV). The main zone has a separate volume control.

The suggested retail price of The Director is $299 and is available at specialty audio stores. For more information, contact Tom Walker or Mark Eshom at Audio Control, 22313 70th Ave. W., Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166, TX 3711409.

Fast Reply #EF123

Fast Reply #EF564

Unlike a fuse, when a PolySwitch PCT from **RAYCHEM** stops overcurrent, it can reset itself. When a current overload hits its conductive polymer compounds, the PTC (positive temperature coefficient) device heats up and turns nonconductive. When you lower the current, the device's conductivity automatically resumes. PolySwitch PTCs are small, monolithic, board-mountable, and testable.

For a free sample and more information, contact Raychem Corp., PolySwitch Division, 300 Constitution Dr., Menlo Park, CA 94025-1164, (415) 361-6900.

Fast Reply #EF372

RAPID SYSTEMS announced its R381, a turnkey FFT spectrum analyzer for Mac 512e, XL, Plus, SE, or II. It features two independent 14-bit A/D channels, an 85dB dynamic range (1mV resolution on ±8V range due to the 14-bit A/D convertors), and cut and paste and autosave capabilities. R381 connects to the serial port of your Mac.

You can set the exact trigger level and conditions you need with on-screen tools, or use the external trigger for remote control. You can also make spot measurements of voltage, time difference, and frequency and zoom in on your signal to check glitches.

The hardware output bit automatically toggles on trigger or start of sampling; you can use it to start an experiment or measurement. You can also read a signal from disk into either channel and compare it with the other channel. Also, you can save your signal complete with setup information or as a text file.

R381 sells for $2,495. For more information, contact Susan Conley, Marketing, Rapid Systems Inc., 433 N. 34th St., Seattle, WA 98103, (206) 547-8311, FAX (206) 548-0322.

Fast Reply #EF948

Fast Reply #EF1342

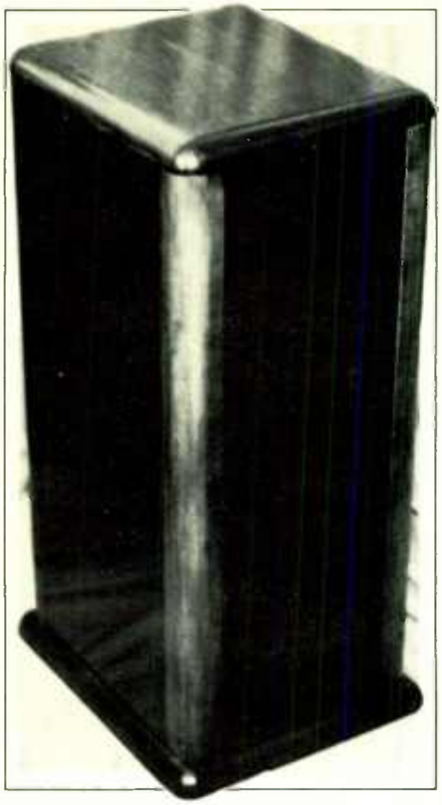
AUDIO VIDEO TECHNOLOGIES has published a guide for those planning to install a Surround Sound system. This fact-filled booklet covers all aspects from sound theory to the best Surround Sound format for your listening room. It also describes how Dolby, Lucas THX, Sony SRS, and Omnisound technologies work and explains the dos and don'ts of high definition Surround Sound.

High Definition Surround Sound sells for $6 alone and for $10 with room templates. With each purchase, a $3 contribution will be made to the Illinois Institute of Technology to support HDSS research. You receive a $10 discount certificate on your first Omnisound Home Theater Surround Sound order. A dealer list is included. For more information, contact Audio Video Technologies, Inc., 60 E. Ida, Antioch, IL 60002, (312) 395-6321.

Fast Reply #EF407

BENCHMARK MEDIA SYSTEMS has moved due to an increase in production and the need for more design and engineering space. The new facility, located at 5925 Court Street Rd., Syracuse, NY 13206, includes a dedicated electronics lab for use in product development and custom design work. For more information on Benchmark products and dealer inquiries, call (315) 437-6300 or FAX (315) 437-8119.

Fast Reply #EF247



GREGENG offers the Pre-stressed Enclosure as a new concept in loudspeaker enclosure design, one which eliminates sound coloration caused by panel vibration. Panels are fabricated of aluminum-faced honeycomb, which is more rigid than typical panel materials such as fiber board or particle board. Tension cables pre-stress the panels internally, which also enhances rigidity.

Unassembled kits are available from Gregeng, 9737 Macleod Trail, PO Box 42206, Calgary, Canada T2J 7A6.

Fast Reply #EF699

SuperStuff™ is now being marketed, a **MARRS** spokesperson announced. These modules fit inside a loudspeaker, conveying high performance by cushioning the bass waves. Low frequency performance can be doubled, according to MARRS, resulting in a superior compact speaker.

The MARRS modules also increase acoustical volume as much as 100 percent by exchanging thermodynamic energy for kinetic energy in the sound waves, according to Ralph Marrs, the inventor of super compliant technology.

SuperStuff modules, while interacting with acoustical energy, are totally passive and automatic, requiring no connections or support. MARRS has offered to respond to all communications with advice on applying this technology in advanced designs.

Call or write MARRS, 6809 Chateau Ct., San Jose, CA 95120, (408) 629-8520.

Fast Reply #EF389

THE BINAURAL SOURCE publishes a catalog of binaural recordings. This is a special recording method designed primarily for stereo headphone playback. Its aim is to place the listener where the sounds were created, rather than trying to bring the sounds into the listening room. No special equipment is required for playback of binaural other than the recordings and a pair of stereo headphones.

For a catalog and more information, contact The Binaural Source, Box 1727, Ross, CA 94957, (415) 457-9052.

Fast Reply #EF410

CELESTION INDUSTRIES of Holliston, Massachusetts, has reported growth in European and American markets. They show a 21% increase in sales, a 56% increase in worldwide operative profits, a 33% increase in production capacity, as well as increased warehouse capacity.

Headed by Peter Wellikoff, president of Celestion's US operation, the company reported excellent results for the first half of the year. He anticipates continued growth through the second half of the year due to the introduction of the Celestion 5.

Fast Reply #EF197

Speaker Builder

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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 (six issues) **$25**, two years (twelve issues) **$45**. All sets of back issues are available beginning with 1980. Canada add **$4** per year for postage. Overseas rates available on request. Subscribers residing outside the US and Canada are served by air.

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Speaker Builder (US ISSN 0199-7920) is published bi-monthly at $25 per year; $45 for two years, by Edward T. Dell, Jr. at 305 Union St., Peterborough, NH 03458 USA. Second class postage paid at Peterborough, NH and additional mailing office.
POSTMASTER: Send address changes to **SPEAKER BUILDER**, PO Box 494, Peterborough, NH 03458.

About This Issue

How many high-school students do you know who enjoy math? **Rich Alden**'s do. His hands-on teaching method of speaker building shows them how math can be practical and fun. Read about his non-traditional technique beginning on page 10.

Arthur Brown's at it again. He's been experimenting with servo subwoofers this time. His approaches begin on page 20.

To overcome the problem of shared walls, floors, and ceilings in apartment living, **Scott Ellis** created a pair of speakers that prevented bass from escaping into neighbors' apartments. Turn to page 32 to see how he accomplished this feat.

First he completed his version of the Swan IV, now **V.H. Estrick** has built a second-order crossover for it. His project begins on page 34.

Mark Gadzikowski offers a construction project of a 20W amplifier. Turn to page 36 for the details.

The Precise Monitor 10 is not so precise, claims Contributing Editor **Vance Dickason**. See page 50 for his review.

On our cover: Steve Chan, a former student in **Ray Alden**'s math/speaker building class at Stuyvesant High School, holds the midrange on his Park Avenue speakers. Photo by the author.

Speaker Builder
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VOLUME 12 NUMBER 1 FEBRUARY 1991

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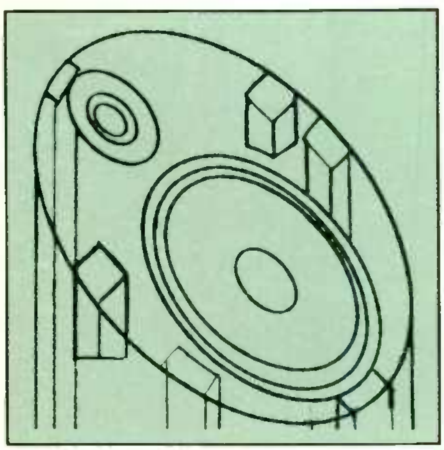

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FEATURES

DEPARTMENTS

Editorial

LATE NIGHT MUSINGS ON SPEAKERS AND BRAINS

Those who read my offerings in these pages know I keep a large table of hobby horses. Like many others I have become keenly interested in the emerging research regarding the human brain. The left brain/right brain studies began it all. Since that topic was first broached over a decade ago, other avenues of interest have developed.

Dr. Oliver Sacks, whose work I have cited in earlier editorials and author of *Awakenings* as well as *The Man Who Mistook His Wife for a Hat*, offers the information in the latter that the brain has the capacity to record permanently every sound a human has ever heard. The brain, in other words, is a recorder of gargantuan capacity beside which even the most advanced memory storage devices from Silicon Valley are dwarfed by comparison. If this is so, what does it say about the claim that most people have poor aural memory. Perhaps what needs investigating is the mechanism of recall and methods of retrieving past experience. Dr. Sacks' *Awakenings* has been made into a movie starring Robin Williams as the doctor and Robert DeNiro as the patient. Well worth seeing according to much of the critical comment.

Last year we reprinted a fascinating article by Andrew P. Stiller entitled "Toward a Biology of Music" (*TAA* 1/90, p. 38). Dr. Stiller's main point was that the universality of music in all cultures indicates its biological necessity to human evolution. Along the way, he also cited vital evidence garnered in extensive experiments at New York University which indicated that persons untrained in music process it primarily with their right brains, while those trained in music or in extensive music listening, process it with their left brains. The data was obtained from brain scans for activity during music listening sessions.

L.B. Dalzell, one of our authors, called my attention recently to the work of Dr. Doreen Kimura at the University of Western Ontario in London, Canada. Dr. Kimura has done extensive work on the right/left brain topic and has published results of research that strongly indicate humans hear music slightly better with their left ears and speech a bit better with their right ears. Of course, our two ears connect to opposite sides of the brain. Thus, your left ear hears music a bit better, presumably because the right brain is, for most of us, the primary processor for music, the left brain for speech.

Dr. Kimura's work has many critics, but the reports on her research seem to confirm the growing knowledge about the ways we use our brains and how that relates to our listening. She has also done a lot of work on the differences between male and female brains. One finding is that women's brains are more symmetric and their verbal skills more equally distributed between the hemispheres.

PBS recently aired an hour-long special on the brain and sexuality. It now appears very likely that the brains of women and men function very differently simply because the differentiating chemistries that make us male and female also tend to cause the brain to develop differently. It is possible that all humans begin existence as female but that males develop on a different emotional track which affects how the two halves of the brain interact. To put it too simply, the connections between the halves of the female brain are more well developed than those in males.

Our evolutionary development as humans is suggested as the reason for this difference. Until very recently in our human time line, males were hunters while females were gatherers. The male was primarily combative which in the canons of natural selection means that the best hunter is likely to be the best survivor— and reproduce himself more often than less combative males. This might well increase the male's use and development of the left brain.

What, if anything, has this to do with audio and loudspeakers? I think we need to pay a lot more attention to how we listen and also to the possibility that men and women hear differently. Few readers of this magazine are women. Nearly every serious audiophile realizes early that women respond differently to music systems than do men. They seem far more sensitive to high frequency distortion than males. Has any attention been paid to the question of male and female distribution in listening tests?

It certainly should be clear by this time that the qualifications of those chosen to participate in listening tests should have their capabilities quantified in some way. It ought also to be clear that random choice of participants from the general populace in any sort of listening test whether by questionnaire, ABX, or whatever form are nearly meaningless unless some care has been taken to determine whether the participants have developed evaluative skills by either training or experience.

The human being is not a scientific instrument. We know far too little about how we process sonic data to have full control over all the parameters that matter in the listener—even if we knew what those parameters were. If the results of listening tests are going to be regarded as scientifically reliable, then the only variable must of necessity be the sound. The human participant is a bundle of variables most of them undefined.

One of the strangest results of Dr. Kimura's research is evidence that brain activity and capability is affected by hormonal levels. Variations in estrogen level in women affect motor skills in a measurable way and similarly, testosterone level variations in men affect their spatial ability. Maybe it is not absurd to check your hormonal levels before deciding that your loudspeaker system is not giving you adequate spatial clues.

I raise these questions not in any attempt to resolve the old golden ear/meter reader controversy. Since very little in the way of basic research appears to be underway sponsored by the audio industry, we are not likely to have any new objective, analytical tools for making deeper analysis of what it is about sound that must be kept intact when it is reproduced. Those tests we have do not account for the differences we hear in equipment which measures the same.

All of these ramblings are meant only to remind us that we do not listen to our speaker systems in a vacuum. Many more elements enter into whether or not we derive the full measure of satisfactions we seek. This magazine has been a miraculous medium for bringing together your very diverse skills from a truly amazing variety of disciplines. Just as those technological skills have affected how we build and measure the hardware, so too other skills in even more diverse fields may well hold the clues to even better results. And some of those answers may well lie within our bodies, and more especially our brains.—E.T.D.

BUILDING SPEAKERS AT STUYVESANT

BY RAY ALDEN

PHOTO 1: Completed speakers: the Ave. A, the 14th St., and the Park Ave. (photo by Luke Tone)

Have you ever returned to your high school to sit in on a class you took years before? More likely, if you came back at all, it was to exchange a few pleasantries, feel a nostalgic twinge, and quickly get on with your busy life. In 1982, when I created an elective course on speaker building, open to juniors and seniors, I never realized how those casual visits of past students would change into continued involvement with the course.

One or two students typically come back during their winter break or in late May or June to work alongside the current students to complete another pair of speakers. Often this former student's new project is for a relative impressed with the sound and handiwork of the original. Sometimes I receive a letter from far away, a cry for help: "My roommate blew up my tweeters when I was away for the weekend. Can you tell me how to get a new pair?" With each of these contacts, I always sense some sort of excitement, a desire to forge ahead and build another pair of speakers or solve a problem. Never in this course do I hear the cliche so common in the ordinary math courses: "When am I ever going to use this stuff?"

ABOUT THE AUTHOR

Ray Alden teaches math at Stuyvesant High School, one of three schools in New York City specializing in science and mathematics. His interest in sound comes from the old-time traditional music of the South. He plays banjo and mandolin in that style, occasionally at festivals with the North Carolina Round Peak Band. Mr. Alden has recorded and produced several award-winning anthologies such as the double LPs "Visits" and "The Young Fogies." His fascination with getting good sound from his musical instruments provided a natural extension to that of speaker building.

HOW IT ALL BEGAN. Recently, some creative science and mathematics teachers have moved toward a hands-on approach, believing lectures teach students the words but not the meanings. Norman Ornstein of the American Enterprise Institute voices corporate concerns when he states: "The Japanese and the Germans seem more disciplined, with a greater work ethic and better technological capabilities. On a basic level of math and engineering, we're clearly behind." A number of reports, starting in 1983 with the National Science Board's Commission on Precollege Education in Mathematics and most recently with the American Chemical Society's "Education Policies for National Survival," emphasize that, particularly in science, students learn best when actively doing.

To be honest, none of those concerns started me designing the course. The speaker building bug bit me in the late seventies while I was working on a record of the old-time country music of Virginia and North Carolina. The sound mix came out of a pair of speakers built by Bobby Patterson (from McGee Radio designs and parts) for his small Heritage Records company. When I returned to New York, I slowly gathered information on the design of sealed and vented enclosures and crossover filters.

It became apparent many basic equations (such as those designed by D.B. Keele, Jr.) could be understood by high school students who had a knowledge of exponents and logarithms and (if they had also studied physics) had some exposure to resistors, capacitors, and in-

PHOTO 2: Delivery day—the speakers have arrived from Zalytron. (photo by Howard Chang)

PHOTO 3: Eddy Hsu carrying Ave. A cabinets. (photo by Howard Chang)

ductors. With my fledgling enthusiasm for speaker building, I instantly forgot the adage: "Fools rush in where angels fear to tread."

My initial idea was to offer a two-period course in math (to teach the theory necessary to design the speakers and build the crossovers) and wood shop (for building the cabinets and assembling the speakers). When the chairmen of the mathematics and the industrial arts departments approached the principal in 1981, the idea was rejected. One year later, multidisciplinary courses were encouraged, and the speaker building course had come alive.

SOME EARLY PROBLEMS. The chairman of the industrial arts department, Frank Wright, was my co-teacher on the maiden voyage. He was supportive and willing to try new ideas. Unfortunately, my expectations about what we could accomplish in the course turned out to be unrealistic.

One major error was having the students wind all the inductors they needed for their crossovers. After we tracked down a source for enamled wire, Frank and the class set up an old lathe and began cranking out the large number of inductors needed. At the same time, they were building hexagonal enclosures for the midrange and rectangular ones for the woofers and turning small oval frames for the round tweeters. Trying to finish the speakers by the end of the course turned out to be a nightmare. About 75% of the class finished their speakers that year, and I was determined this situation would not be repeated.

For six terms, I worked with the shop department. In the final one, Richard Realmuto, a wonderful woodworker, ensured that the students finished their cabinets. At that time, however, a bureaucratic issue arose dealing with how credit for the course was to be given, and it threatened the existence of the course. After a joint conference, the principal decided to permit me to teach the course alone in one period—in an ordinary classroom with one clothing closet, two electrical outlets, and one teacher's desk.

Fortunately, I had been in touch with many supportive driver manufacturers. Ted Jordan in England answered questions, Evan Struhl sent Polydax catalogs full of driver specifications, and Ed Hanson of Philips-Amperex supplied books on speaker theory. Ed, a technical coordinator instrumental in the development of the cassette and the dome tweeter at Philips and a former student of Stuyvesant, steered me toward a person who would prove to be essential to the course—Elliot Zalayet, whose family owns Zalytron Industries.[1] Elliot not only supplied drivers, custom inductors, and capacitors, but also agreed to build the cabinets, enabling the course to survive.

THE COURSE BEGINS. About a month before the beginning of the term, I visit Zalytron and find out about

PHOTO 4: Sung Sen, David Polyak, Adam Jaffe, Kahlia Fisher, and Gideon Cohen get a lesson from Ray Alden. (photo by Josh Pilner)

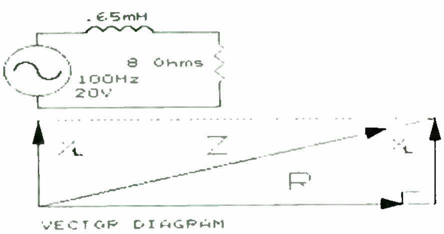

FIGURE 1. Circuit and vector diagrams for a typical problem-solving session.

good deals on drivers so the students can get the most affordable speakers. Elliot and I also explore cabinets, reviewing those in stock or ones he could custom-build. I narrow the possibilities down to three or four, with prices ranging from inexpensive (low-cost drivers in a two-way Boston Acoustics A-40 cabinet) to moderately expensive (a three-way 30- or 40-liter cabinet, veneered in a choice of colored formica, using drivers such as the TPX plastic-coned Audax TX1125FSN midrange and the now extinct Precision TX205F 8″ polypropylene woofer).

By the third day of the new term, students are shown the choices and given box dimensions and costs. They must make choices quickly so we can get supplies in time to complete their speakers. One of the nicest touches is that Elliot provides scholarships for students who could not otherwise afford to take home a pair of speakers.

The first month and a half of the five-month course is devoted to the theory and design of speakers. This whirlwind tour leaves me breathless but allows enough time in the course to build the speakers.

The students are initially concerned with *power*. "How many watts can this speaker take?" is their most common first question. My mind forms an image of a pool of melted tweeters as I inform them of electrical devices that prevent damage by permitting the correct frequencies to go to the appropriate drivers.

After introducing the capacitor and the inductor, I give them a sense of "storage" in the capacitor and the understanding that applying an AC current to an inductor creates a changing magnetic field and a back EMF. I then introduce the two basic reactance formulas: $X_C = 1 / 2\pi FC$ for capacitance and $X_L = 2\pi FL$ for inductance (X in ohms, F in hertz, C in farads, L in henries).

PROBLEM SOLVING. When dealing with reactance and resistance, voltages and impedances are not necessarily added arithmetically. For example, the voltage drop across a resistance is in phase with the current; the voltage drop across an inductive reactance is 90° ahead of the current. Introducing vectors to represent these phase differences, along with Ohm's law ($I = E/R$) and the equation for power in watts ($P = I^2R$), I enable the students to see the power supplied to an 8Ω speaker with an inductor in series with it. They are asked to solve in class the following type of problem using the circuit and vector diagrams in *Fig. 1*:

(A) *Find X_L.* They immediately change the inductance formula to work in mH.

$$X_L = .0062832FL = .0062832 \times 100 \times .65$$

$$X_L = .4084\Omega$$

(B) *Find the total impedance (Z) of the circuit.* The 90° phase difference is

Tips for Starting A Speaker Building Course

I advise only those who really enjoy speaker building to attempt this course. Only with a certain amount of passion will you have the fortitude to get through the initial maze. Once through, however, the rewards are many. Students will tell their friends how enjoyable the course was—a necessary condition for any elective to survive—and they will be more involved and cooperative than in the ordinary courses. And parents will compliment you and appreciate the course.

Conditions for beginning such a course will vary from school to school. Eckhard Lutz, a teacher at Robinson High School in Ontario, Canada, said his school is providing broad support to start a speaker building program. They will be able to design enclosures using CAD programs and then build them in the shops. I suspect the school will also be supportive in getting CACD and CASD. On the other hand, Jeff Levin, who teaches at the Manhatten Center for Science and Math in New York, has had to push reluctant administrators to allow digital and audio courses using the hands-on learning style. Even with these difficulties, he helped a student who plays the keyboards build his own power supply, amp, and speaker.

Funding is a potential problem for any course requiring supplies other than books. Some students may want to take the course but be unable to afford the materials. You may be able to arrange a scholarship with a manufacturer. Or perhaps your school has a parents' association that provides funds. Also, you could build up a supply of drivers, electronic parts, and enclosures that could be built, disassembled, and used again.

Some people will always say it cannot be done. It will remain for you as a creative problem-solver to show the nay-sayers how to give courses that not only teach mathematical and technical skills, but also give students the motivation to want to learn these skills.

PHOTO 5: Elizabeth Pan and Mike Gromm work on their speakers on the teacher's desk while John McGregor, Amy Baxt, and others help or observe.

recognized as the Pythagorean theorem of geometry.

$$Z = \sqrt{R^2 + X_L^2} = \sqrt{8^2 + .4084^2}$$

$$Z = 8.0104\Omega$$

Sometimes a student observes that $Z \approx R$ unless X_L takes on larger values.

(C) *Find the current flowing through the circuit using Ohm's law.*

$$I = E/Z = 20 / 8.0104 = 2.497A$$

(D) *Find the power to the 8Ω speaker.*

$$P = I^2 / R = 2.497^2 / 8 = .78W$$

After giving the class this problem, I assign homework: "Use the same circuit diagram, except allow the frequency to vary using the following values: 50Hz, 150Hz, 300Hz, 500Hz, 800Hz, 1kHz. Then recompute the power to the speaker." They are asked to plot a graph of these results, showing power on the Y-axis and frequency on the X-axis. They will see the inductor "choking" off the power to the speaker as the frequency increases.

Using the same basic circuit with a capacitor instead of an inductor, or later using both in the circuit, will show how various frequencies are permitted to play at full volume, while others are not. Much of this may be old hat to many of you, but it is a revelation to the students, many never realizing anything was needed to direct bands of frequencies to the various drivers. Other revelations will come when the students find out "8Ω" is a gross distortion of the real nature of the impedance of a driver.

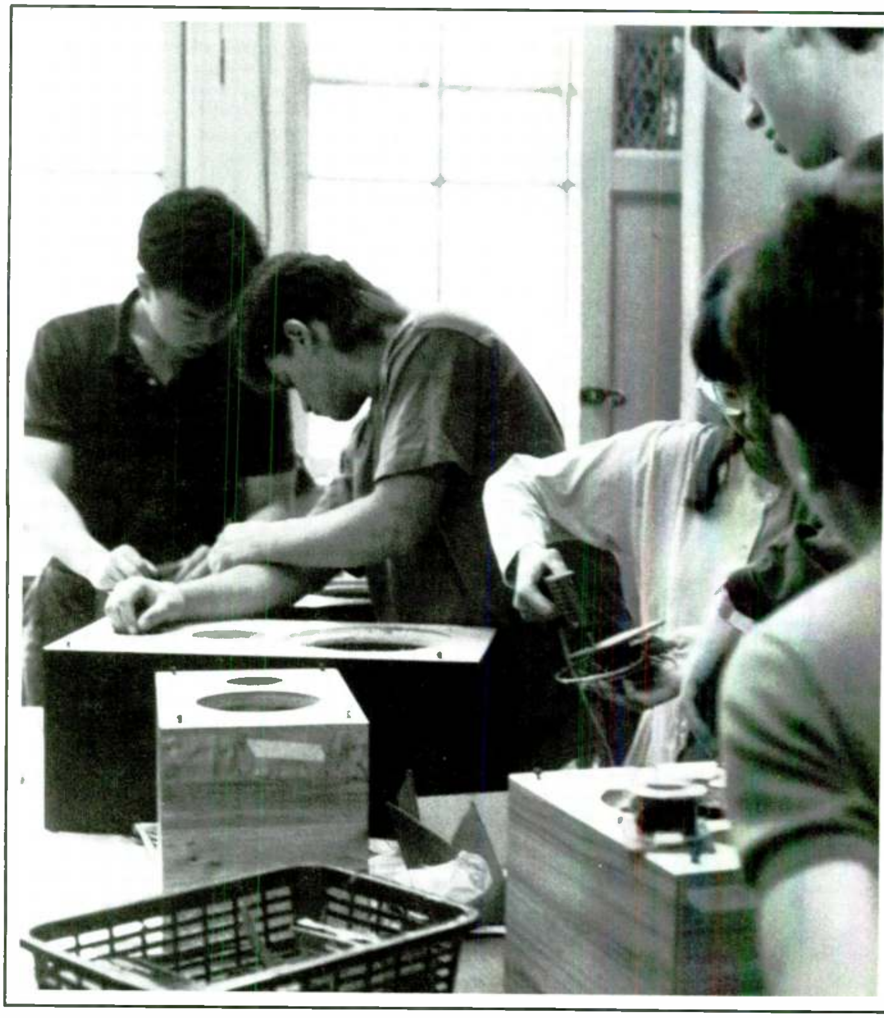

PHOTO 6: Left: Mike Gromm gets help on his tweeter assembly. Right: Elizabeth Pan hot glues the back of her crossover board while Paul D'Arcy helps.

Mathematical concepts are introduced at many places in the course so students learn they are valuable tools and not merely used to substitute numbers into equations. I'll give an example useful to those who design three-way crossovers using formulas such as those derived by solving the reactance formulas for capacitance and inductance above for C and L: $C = 1 / (2\pi FR)$ and $L = 2\pi FR$.

PHOTO 7: Sammy Pang soldering his Ave. A tweeter. (photo by Ray Alden)

Students learn that putting the appropriate capacitor and inductor in series will produce a bandpass filter useful for a midrange in a three-way speaker. However, one problem mentioned in books is that the interaction between the reactance of the bandpass components and the reactance of the components in the adjacent high- and low-pass sections causes difficulty in the straightforward calculation of the midrange components. That is to say, if you use the chosen low-frequency crossover point (F_L) and the high-frequency crossover point (F_H) in the formulas for C and L above to calculate the midrange components of a three-way, the midrange will not cross over at those points but at values below F_L and above F_H.

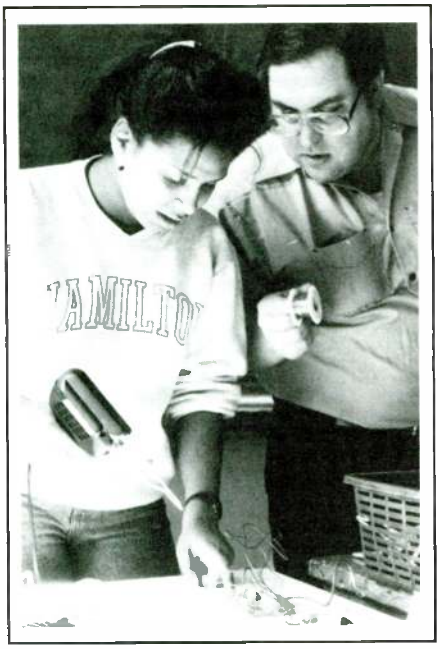

PHOTO 8: Barbara Gordon and Ray Alden discussing soldering techniques.
(photo by Amy Baxt)

One way to deal with this problem is to find two new frequencies (which I will call the shifted frequencies, F'_L and F'_H) that will correctly compute the midrange components so the midrange will behave as you wish it to.

Finding the shifted frequencies involves finding the means (the average) between F_H and F_L. To the students, and perhaps to most people, the average between F_H = 2,500Hz and F_L = 500Hz is 1,500Hz—the sum divided by two. However, something goes awry when I ask the students to solve the following:

"Neville Thiele rode for two hours out into the Australian countryside on a bicycle going 12 mph for 24 miles, at which point he had a flat tire and had to walk back for six hours at 4 mph to cover the 24 miles. What was his average rate of speed for the round trip?"

Again the students would take the sum of 12 mph and 4 mph, divide by two, and answer that his average rate was 8 mph. However, this is not the correct answer. Traveling 8 mph for the total 48 miles would take only six hours instead of the eight indicated.

The idea of means and averages needs to be enlarged to arrive at the correct answer and also to work with the average of the crossover frequencies in the bandpass situation. The correct answer for the round trip problem is obtained by taking the total distance (48 miles) and dividing by the total time (8 hours) giving 6 mph. Doing the same with symbols will give the correct formula for the average. Take the total distance (round trip = 2d) and divide by the total time ($d/r_1 + d/r_2$). This, when simplified, gives you $R_{av} = 2r_1r_2 / r_1 + r_2$.

This is the *harmonic mean* and it restores the proper balance between the two rates (Neville travels much longer at one rate than at the other). The original method, the *arithmetic mean*, does not give the proper "weight" to each rate.

Now that students realize more than one type of "average" exists, I introduce a third type of means between F_H and F_L, the *geometric mean*. Students are asked to recall a special proportion called a "mean proportional," such the geometric mean frequency (F_M), which falls between F_H and F_L:

$$F_H / F_M = F_M / F_L$$

Solving for F_M gives us:

$$F_M = \sqrt{F_H F_L}$$

The bandpass LCR series circuit resonates at F_M, acting like a low-pass filter above F_M and a high-pass filter below F_M. *Figure 2*, a trapezoid with upper base a and lower base b, illustrates the relative size and position of all three means. The inequality

$$2ab / a+b > \sqrt{ab} > a+b / 2$$

indicates that the harmonic mean (HM) is less than the geometric mean (GN), which is less than the arithmetic mean (AO). Also, the geometric mean *is* the geometric mean between the harmonic mean and the arithmetic mean:

$$GN = \sqrt{HM \times AO}$$

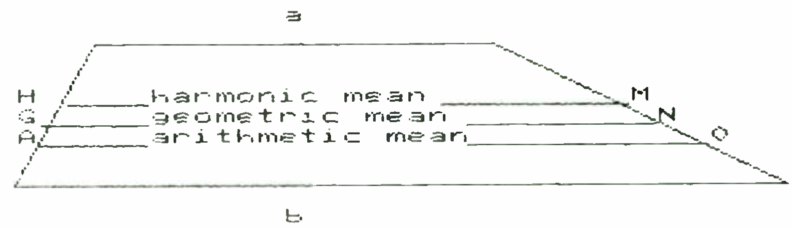

FIGURE 2: Relationship of the three types of means.

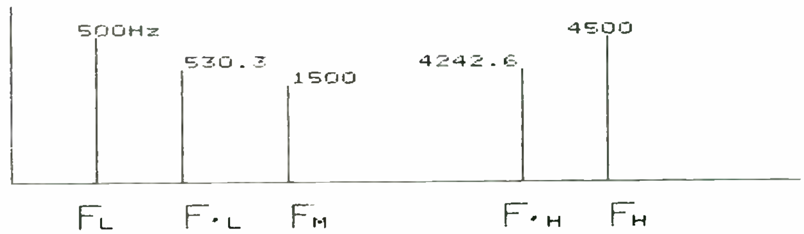

FIGURE 3: Diagram illustrating the relative position of all frequencies in the homework.

(AO is the median of the trapezoid, GN creates two similiar trapezoids, and HM passes through the intersection of the diagonals.)

Since the shifted frequencies have the same geometric mean as the actual crossover frequencies, you can write:

$$F_M = \sqrt{F'_L F'_H} = \sqrt{F_L F_H}$$

This means:

$$F'_L F'_H = F_H F_L$$

The ratio of the shifted frequencies can be made one unit smaller than the ratio of the crossover frequencies, giving the following equation:

$$F'_H / F'_L = (F_H / F_L) - 1$$

Solving for these two equations gives you the following:

$$F'_L = \sqrt{F_H F_L / ((F_H / F_L) - 1)}$$

$$\text{and } F'_H = F'_L ((F_H / F_L) - 1)$$

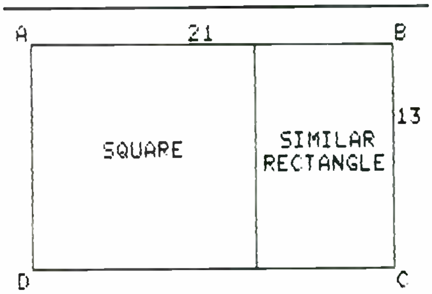

FIGURE 4: Using the Fibonacci sequence and the golden ratio determines the dimensions of a room that will produce quality sound.

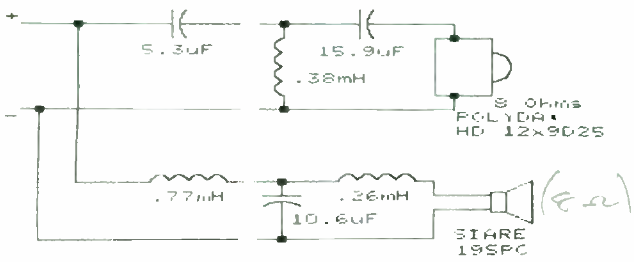

FIGURE 5: A crossover—using a Polydax HD 12X9D25 (8Ω) and Siare 19SPC—designed by formula alone.

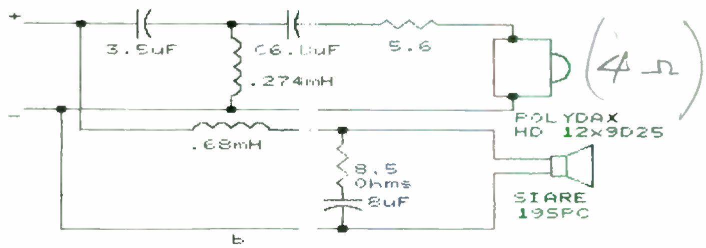

FIGURE 6: The optimized crossover using a Polydax HD 12X9D25 (4Ω) and a Siare 19SPC (8Ω). The 8.5Ω resistor in series with the 8μF capacitor provides impedance compensation for the 19SPC woofer.

The students would then be assigned a homework problem something like this: "Find the inductor and the capacitor to use in the midrange section of a three-way first-order filter so the midrange will cross over to the woofer at 500Hz and then to the tweeter at 4,500Hz."[2] (*Figure 3* shows the results.)

(A) *Find F'_L first.*

$$F'_L = \sqrt{4{,}500(500) / ((4{,}500/500) - 1)} = 530.3\text{Hz}$$

(B) *Find F'_H.*

$$F'_H = 530.3((4{,}500/500) - 1) = 4{,}242.6\text{Hz}$$

(C) *Find the inductor needed for the midrange section (midrange = 8Ω) using the shifted frequency.*

$$L = R/2\pi F'_H \text{ (in henries)} =$$

$$159.1549R/F'_H \text{ (in millihenries)} =$$

$$159.1549(8)/4{,}242.6 = .3\text{mH}$$

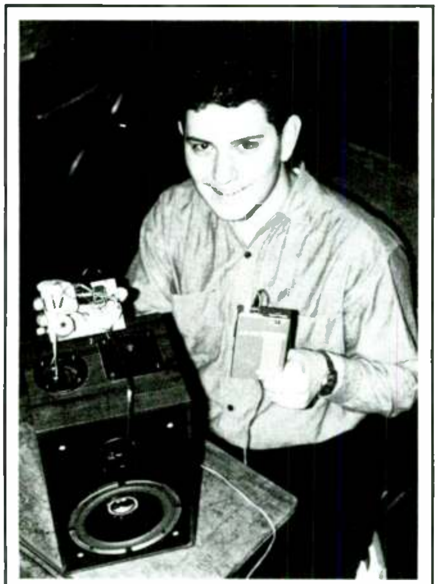

PHOTO 9: Roman Frenkel about to test his crossover on the "testing" speaker.

(photo by Ray Alden)

(D) *Find the capacitor needed in the same series circuit.*

$$C = 159{,}154.9/F'_L \times R \text{ (in microfarads)}$$

$$= 159{,}154.9/530.3(8) = 37.5\mu\text{F}$$

THE GOLDEN RATIO. Mathematical ideas are woven into the course in other places as well. The relationship between the Fibonacci sequence of numbers and the golden ratio[3] are introduced after asking, "Are some box shapes better suited for speaker enclosures and for rooms in which to listen to music?" The Fibonacci sequence {1,1,2,3,5,8,13,21,...}, in which each number is formed by the sum of the two preceding numbers, would give an approximation to the floor plan of one of the "better sounding" rooms if you picked two consecutive numbers in the latter part of the sequence.

For example, a 21- by 13-unit room (*Fig. 4*) has the ratio 13 / 21. This equals .619047619, a close approximation of the golden ratio (an irrational number ≈ .61083...). In other words, it is the ratio you would get if the rectangle were divided into a square and a smaller rectangle similar to the original. The golden ratio and its reciprocal (phi ≈ 1.61803...) can be used to figure out the optimal dimensions of a room or a speaker enclosure.

The students are given a problem along these lines: "Find the internal dimensions of a speaker to enclose a volume of 1,423.83 cubic inches using the golden ratio." The solution would be obtained as follows. The width would be $\sqrt[3]{1423.83} \approx 11.25$ inches. To get the depth, multiply the width by the golden ratio: 11.25 x .61803 = 6.95 inches. Multiply the width by phi to get the height: 11.25 x 1.61803 = 18.2 inches.

When these three dimensions are multiplied, you will get a close approximation of the required volume. Further, at least two ratios between the three dimensions will be close to the golden ratio.

Some people might think my hidden agenda is teaching mathematics, with speaker building thrown in on the side. However, students have made comments such as this: "It seems a lot of education these days is theory and memorization. In this course, we learn the theory involved and then apply it to a tangible product, and can see how what we learn is used." (Joe Mancuso, May 1990)

CASD AND CACD. The speaker systems we build usually vary from term to term. This is the reality of a fluid speaker world where the Precision driver or the Audax Bextrene driver ceases to exist. Finding what is available, in what quantity, which are the best value, and which will perform best in stock cabinets and which in custom-designed cabinets are issues resolved before the students appear in the class. Also, a crossover must be worked out in a manner more sophisticated than could be derived using a few simple formulas.

Leisurely experimenting with designs one month before the completed ones are needed is impossible. The solution came from Ken Kantor's articles, "Speakers by Design," in *Audio* magazine (Nov. 1988 and Dec. 1988). It was there I read about Scientific Design Software's (PO Box 3248, Chatsworth, CA 91313) two speaker-building CAD programs: CASD for enclosure design and CACD (reviewed by Bob White in *SB* 1/89, p. 42) for crossover design.[4] I wrote to Ken about relying on published data to use in the CAD programs. Ken answered, "In general, you can trust published response curves, but are better off if you can test your own Thiele/Small parameters." Slowly, I am getting the equipment needed to measure these parameters. The most recent acquisition is a B&K 3011-B function generator with a built-in frequency counter.

We give the speakers names reflecting the streets near the school (the Ave. A, the Park Ave.). I'd like to highlight a two-way design we call the 14th St. This model is based on a stock cabinet from Zalytron, the Micro-Acoustics box, and has dimensions of 17″ x 8″ x 10″. Its internal volume is approximately 15 liters, making it suitable for a Siare 19SPC, stocked by Zalytron. The published parameters for the 19SPC are as follows:

$$f_s = 45\text{Hz}$$
$$V_{AS} = 20 \text{ liters}$$
$$Q_T = .8$$
$$\text{Sensitivity} = 89\text{dB/1W/1M}$$

An EBP of 43 suggests that the Siare 19SPC is most suitable in a closed box environment. Using the data in CASD verifies that, in a 15-liter box with filling, the 19SPC is −3dB at 50Hz, deviating from the reference 89dB by only 1dB in the vicinity of 100Hz. Next, Elliot suggested a Polydax HD 12X9D25 impregnated fabric dome tweeter. It has a sensitivity to match the Siare 19SPC and, with an f_s = 800Hz, should cross over nicely at 2,500Hz. If you use an 18dB/octave filter, the tweeter should not be excited audibly at resonance. Elliot said he had the Polydax tweeter in stock only in 4Ω, but he and several customers had used it in series with a 5.6Ω resistor with excellent results. This completed the choices and I proceeded on to CACD and design of the crossover.

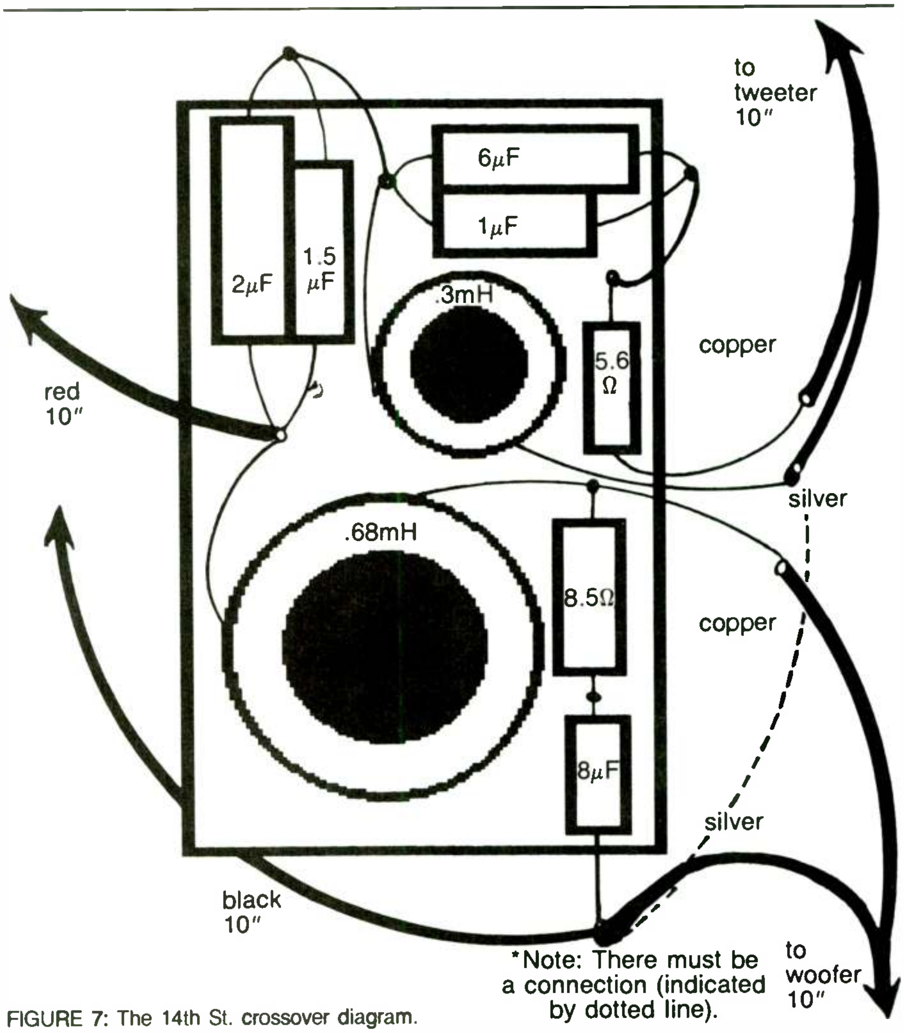

FIGURE 7: The 14th St. crossover diagram.

PHOTO 10: A student twisting wires on a pair of speakers.

If the students calculate the parts needed for a two-way, 18dB/octave filter, *Fig. 5* would be the result. However, once reminded the termination to the woofer is not into an 8Ω resistor but into a complex impedance curve that rises in the higher frequencies due to the voice coil inductance, they can be guided to introduce an impedance compensation circuit.[5] They are further reminded the natural falling response curve of the woofer in the higher frequencies, when combined with the filter, may produce a curve falling at a rate greater than 18dB/octave. They then feel the frustration many of us have felt when trying to produce a viable filter.

CACD is introduced to approximate a target curve of an ideal 18dB/octave filter. The students are shown a computer printout of a crossover filter designed by formula alone and CACD printouts with the addition of impedance compensation and circuit optimization. The refined crossover (*Fig. 6*) shows that with impedance compensation, you need only a 6dB/octave filter in the low-pass section to achieve the desired 18dB/octave filtering.

The students are given schedules of days to work on their crossovers; on the other days, they help others complete their crossovers. This same tactic is used when they integrate all of their parts with the enclosure. This way no one feels stranded if a problem arises, and the students learn to work as a team. When nobody has to call for help, I believe I have been successful in giving the students independence and putting myself out of business as the sole instructor.

TO BUILD A SPEAKER. On the first day of construction the students are given a Masonite board, inductors, capacitors, resistors, and a gob of "Liquid Nails" adhesive caulk (affectionately called "the schmutz") to stick it all on with. As an aid in transferring the schematic to a real-life visualization, they are given the 14th St. crossover diagram (*Fig. 7*).

The next time they work on their crossover boards, "the schmutz" has

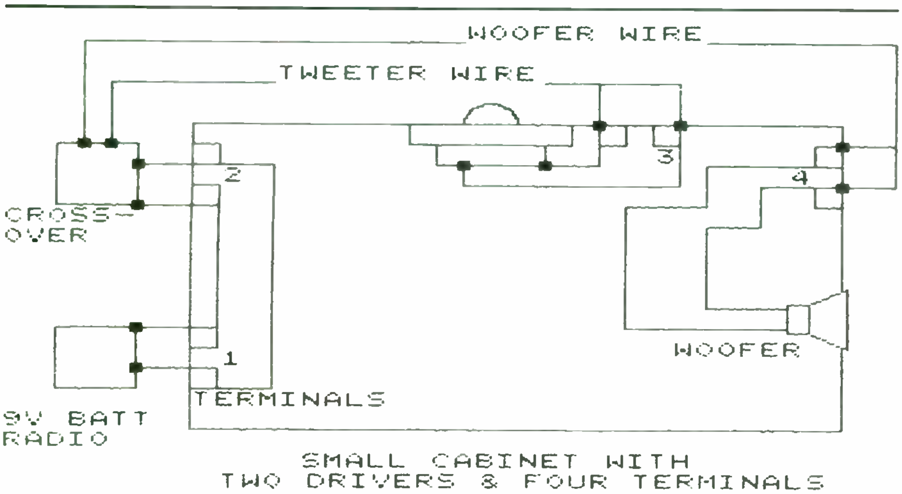

FIGURE 8: Schematic of a two-way speaker used to test the crossovers.

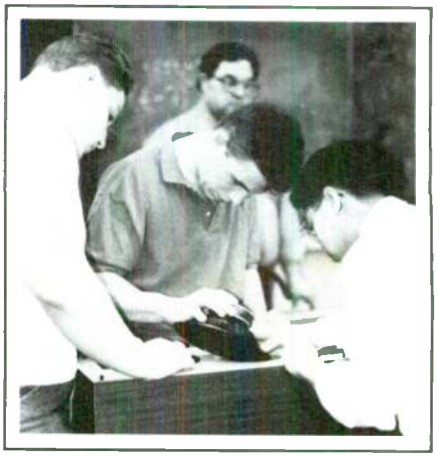
PHOTO 12: Steve Chan solders 18-gauge wire to his woofer from the crossover board while getting help from fellow students.

dried so the components are firmly stuck on. Using their hands and needle-nose pliers, the students wrap adjacent capacitor wires together, along with any inductor wire, hookup wire, or speaker wire that terminates at that particular junction. Some of the students produce crossovers approaching jewelry making in terms of the beauty and tightness of the wrapping.

With everything secured, soldering is next. I find it difficu't to get the students to produce a good solder joint at first; many do not realize the joint must be quite hot before they apply the solder. Finally, this idea and the solder sink in, giving the students a completed crossover.

Occasionally, even though the crossovers looked fine visually, we had problems once everything was in the enclosure. So, over the past few years, I have used a device to test them—a two-way speaker without an internal crossover (*Fig. 8* and *Photo 9*).

The wonderful part about this testing device is the students finally see all the mathematics, the electrical theory, and the cold parts come alive. Putting their ears to the tweeter to determine if the high-pass section is working gives them their first visceral feeling in the course—an exciting moment for them and me. At that point, we use an old elementary school device and stick a gold star on the crossover indicating it has passed muster.

The second phase of construction is completion of the speakers. On the first day of assembly, the students clean the dust from the side walls inside the enclosures. This is important since the hot glue used on the rear of the Masonite crossover boards does not adhere well to dust.

After that, they run the red and black hook-up wires out through the terminal hole, solder them to the correct color-coded terminal part, and hot-glue the terminal in place. Students who have finished become teachers by helping


PHOTO 11: Siliconing the hole prior to putting on the woofer in a ported two-way.
(photo by Venu Pilarshi)

TABLE I

COMPARATIVE RESULTS—THE 14TH ST. vs. COMMERCIAL SPEAKERS

Student	Speaker	Recording/Comments
Mirian Liss	Emerson Model 34	*Pavarotti at Carnegie Hall*: Notes sounded clear and precise on the 14th St. Sometimes the bass seemed constrained but, at certain frequencies, it became impressive. The Emersons played a little louder, but quality and precision of notes were sacrificed. The bass was considerably weaker and sounded tinny.
Eddy Hsu	Altec-Lansing 101 (titanium tweeter, 6½″ carbon-fiber woofer, 91dB output)	*Forever Young*, Alphaville: The Altecs have a smaller cabinet but more power. The tweeters sounded brighter. The 14th St., however, had a clearer midrange sound. Voices came out clearly. The bass was also much deeper. For less expensive speakers, they held their own.
Christine Liu	Kenwood JL-760 (12″ woofer, 4½″ midrange, 2″tweeter, totally sealed three-way with dimensions of 14¼″ x 28″ x 10½″)	*Right Here Waiting*, Richard Marx: The piano solo in the beginning was more distinct and clear on the 14th St. than on the Kenwoods. Sometimes, I couldn't tell which were playing. My parents were surprised at the great sound quality and professional look of the speakers I built.
Ganga Nair	Fisher STV-884 (15″ woofer, 4″ midrange, 2″ tweeter. Speakers stand about 3 feet tall)	*The Time of My Life*, Bill Medley & Jennifer Warnes: The bass on the Fishers was deeper, but sounded good on the 14th St. The balance between bass and treble on the 14th St. was excellent and gave the song more clarity.

out others. This reinforces what they have learned and gives them a can-do attitude. This is particularly satisfying to the female members of the class, many of whom initially lack confidence in putting together an electrical device.

Next, the students stuff the box with a polyfiber called "stuff and fluff," purchased at a local fabric store. Then the 18-gauge speaker wire for the woofer and the 20-gauge wire for the tweeter are run out of their respective holes and soldered to the drivers. The students are careful to cover the cones with a strip of paper so no solder drips on the cone or surrounding material. Then they use a bead of silicone caulk around the holes as a seal and as an adherent for the drivers. They try not to compress the interface, but let the driver sit on a layer of silicone that will absorb some of the driver's vibrations.

The next day the students "power test" their dry, sealed speakers. Our equipment consists of a 1W/channel Radio Shack amp and a Walkman or a portable radio with a 9V battery.

The final assignment was to gauge their accomplishment against another pair of speakers (usually their parents' larger, more expensive speakers) using their own testing device—their ears. I further asked them to use several dif-

Continued on page 97

REFERENCES

1. The story of Zalytron is fascinating. Started by Elliot's father, Nat, in 1955, its emphasis was on electronic tubes for hobbyists. Nat, who spoke eight languages, had worked for the British Intelligence Service and had ties in Europe and communication with Eastern Bloc countries during the cold war. At that time, hobbyists purchased tubes from large corporations such as RCA, Westinghouse, Sylvania, and GE. With their virtual monopoly on these products, the companies could charge high prices. Nat began to import tubes from Europe and Japan and sold them under the Zalytron name at excellent prices. Nat even managed to get Eastern European tubes, among the best, through Belgium. Elliot continues this tradition of giving reasonable prices to the hobbyist by searching out surplus supplies of excellent drivers. He is currently putting the finishing touches on the Zalytron line of drivers, which will offer high quality at reasonable cost.

2. In the May 1985 issue of *Speaker Builder* (2/85, p. 26), Robert Bullock solves the difficulties of the three-way passive crossover. You will recognize his equation #3 as the geometric mean we have used to find the shifted frequencies. The article will help you deal with these crossovers in a far more comprehensive way than anything presented here.

3. The golden ratio surfaces in many places: in art, architecture, nature, and speaker building. While looking through Ted Jordan's *The Jordan Manual*, I noticed he used it as the solution to the question "What value of speaker Q will damp all three resonances in a bass reflex speaker simultaneously?" His answer required the solution of a quadratic equation of the form you would get by multiplying the proportion created by a golden rectangle and its similar rectangle (let the short side be X and the long side be 1). This would give the quadratic equation $X^2 + X - 1 = 0$. Ted Jordan found the Q required is .618, the golden ratio. You can read more about Ted Jordan in *Speaker Builder* 2/84 (p. 6) and 3/84 (p. 26). Further reading on eliminating the resonances in a bass reflex speaker can be found in G.R. Koonce's "The QB_3 Vented Box Is Best" (*SB* 5/88, p. 22).

4. Particular thanks to Al Tarendash, Sandy Newman, and Mike D'Alleva, colleagues without whom I never would have obtained CACD or set up a computer.

5. They may use an impedance compensation circuit such as that mentioned by Vance Dickason (*The Loudspeaker Design Cookbook*, p. 62) or David Weems ("Make Your Speakers Behave Like Resistors," *Hands on Electronics*, December 1986, p. 68).

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You have to understand, Focal's stamped frames were beautifully made (1.5 mm steel) and no joke at all. In fact far superior to quite a few third party cast frames.

Well, you would have thought the same: why not take some of these brand new stamped frames and put them together with some brilliant moving assembly from Focal's park of technology at a close-out price! Focal did not resist long and here it is: the 8VO1DBL, dual voice coil woofer

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Total Driver Q	Q(ts)	=	.425
Driver Electrical Q	Q(es)	=	.48
Driver Mechanical Q	Q(ms)	=	3.733
Equiv. Volume	V(as)	=	1.63155 cubic fit
Nominal Impedance	Z(nom)	=	4 Ω
DC Resistance	R(e)	=	3.1 Ω
Max. Thermal Power	P(t)	=	70 Watts
Sensitivity (1W/1m)	SPL	=	91.05 dB
Magnet Weight	Magnet	=	1.4 lbs.

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DEVELOPING SERVO SUBWOOFERS

BY ARTHUR E. BROWN

I am very impressed when I run into something completely different from previous experience. Hearing a *good* stereo recording on my newly developed stereo system ("A Stereo System Odyssey," *SB* 1/89, p. 16) was one such experience. I had another when I listened to my stereo with my new servo subwoofers. The servo subwoofer development appears to make major improvements in sound reproduction technology.

I have my fourth servo subwoofer on line and continue to be delighted at the improvement in clarity and staging and the absolute quiet when no signal is present. I now hear the timbre of bowed strings and percussion instruments. The notes flow in a seamless fashion from satellite to subwoofer as a bass fiddle is plucked up and down the scale. Organ notes breathe. The locations of instruments are specific and stable. I will describe my servo subwoofers and outline the approach I used to develop them.

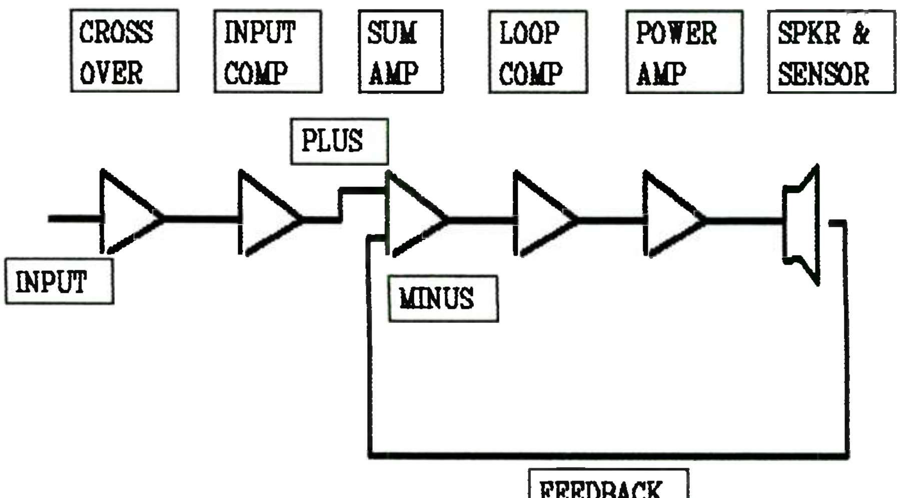

FIGURE 1: Servo system diagram.

SERVO SYSTEM APPROACH. A servo speaker system measures the output of the speaker and compares it to the input signal at the summing amplifier as shown in *Fig. 1*. Differences between the input and feedback signals are amplified to drive the speaker more like the input signal. This design approach corrects speaker overdrive and oscillations muddying the sound reproduction of open-loop speakers. It provides linear frequency response over the system's frequency range and significantly reduces distortion. A servo speaker system's elements include a speaker, sensor and feedback amplifiers, a power amplifier to drive the speaker, a loop compensation amplifier for stable servo operation, and a summing amplifier.

The electronic crossover frequently is included in the input amplifier, and an input compensation amplifier may be required depending on the sensing method.

Let's consider each of these components as they apply to the various servo systems. First I follow these steps in developing a servo speaker. This requires being able to build and develop electronic circuits and the instruments necessary to obtain data for the development.

THE DEVELOPMENT PROCESS.
1. I select a driver and an acoustic suspension enclosure with a sound pressure response about 15dB down at 20-30Hz. This generally results from having a resonant frequency of 50-60Hz for the driver in the enclosure. *Fig. 2* shows the sound pressure data (close miked) for one of the drivers I chose to develop into a servo speaker.

The speaker should be able to produce the desired sound pressure level with a single driver. Using two subwoofers gives you more power but requires two sets of

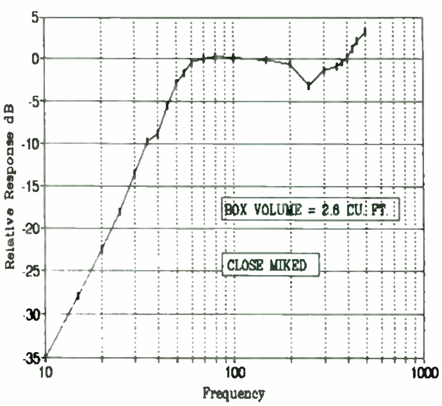

FIGURE 2: The sound pressure at a constant input voltage for a PP10.2 in a 2.6 cu. ft. box.

ABOUT THE AUTHOR
Arthur Brown has built loudspeakers since 1948, though inactive from 1960 to 1986. He is also a hobbyist in electronics, primarily audio and instrumentation, and photography. He graduated from Purdue in 1948 with a BS in Aero Engineering and worked in the aerospace and automotive industries, in controls, instrumentation, and product development. Now retired, he has added the personal computer to his many activities.

electronics. In a servo system, an average input signal causes the speaker drive voltage at some frequencies to be at least four times greater than the average drive voltage. At those frequencies, the speaker drive power will be 16 times (12dB) the average, if not more. The peak-to-average signal in recorded music is reported to be 10dB. In my experience, these high peak signals do not occur at low frequencies. The amplifier and driver, however, should be selected to handle the added power.

2. Select the sensor you wish to use for this system. Install it on the driver and place the speaker in the enclosure. Measure the sensor output volts at various frequencies at a constant sound pressure. I do this with the calibrated mike located within an inch of the cone. I normalize the curve by calculating the decibel deviation from an arbitrary voltage and use this deviation to indicate the sensor's fidelity in representing the sound pressure of the speaker.

If the fidelity is not acceptable, you need to compensate to make it satisfactory or use another sensor. With constant sound pressure, acceleration sensors produce a nearly constant signal at frequencies below the sensor resonant frequency. A velocity sensor has a 6dB per octave attenuating curve with rising frequency; a position sensor has a 12dB attenuating curve.

3. I like to measure the sensor voltage in response to increasing speaker drive voltage. If it follows the speaker linearly, the sensor will be an accurate representation of the speaker as a feedback signal at all power levels. If the linearity is unsatisfactory, reconsider the design. Do not burn out your speaker by driving it with high-power sine-wave signals for a long period of time. The speaker RMS (sine wave) wattage rating is different from the usually published music power rating (also see the sidebar "Travel Limits of Drivers").

4. Once you accept the sensor, you need to develop data showing the speaker/sensor gain and phase shift. Measure the speaker drive volts and the sensor volts over the range of sine-wave frequencies. Also obtain the phase relationship between the two voltages.

5. Determine the loop compensation required for loop stability. An appropriate loop gain curve has a peak (10-20dB) in the center of the frequencies over which you will servo the speaker. The gain curve falls off through 0dB at higher and lower frequencies. I plot the phase data from 150° to −150° as the phase shift from the central frequency, usually

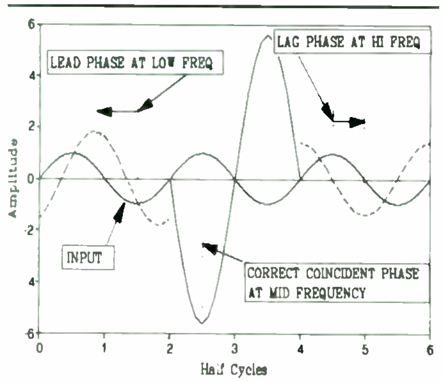

FIGURE 3: Relationships of the lead, coincident, and lag phases.

0° or 180°. The loop compensation amplifier, an active filter, provides the shape of the loop gain/phase curve that is not present in the driver/sensor curve.

Once the compensation is established, adjusting the loop gain will change the frequencies where 0dB occurs without changing the phase shift curve. Higher loop gain increases the spread of the 0dB frequencies and lowers distortion. The 0dB frequencies should be the same as the 130° phase-shift frequencies. Too much loop gain produces undesirable regeneration.

6. Build a breadboard of the loop amplifiers and hook it up to the power amplifier and driver/sensor. Run the open-loop gain and phase data to determine whether it satisfies the requirement (step 5). I input the signal where the sensor is normally connected, reading the output at the sensor. If it is unsatisfactory, redo steps 5 and 6 to refine the curve. Do not connect the sensor to the summing amplifier until later (step 7).

7. You must verify and set the phase

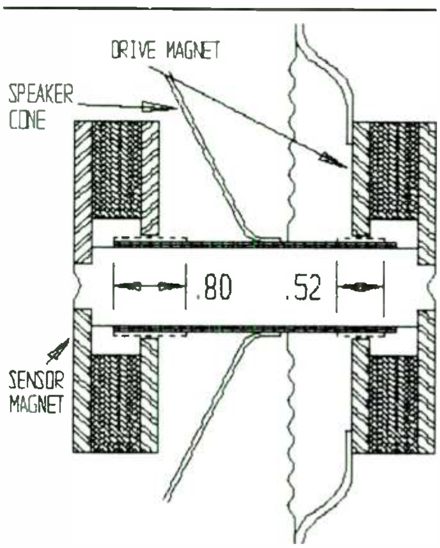

FIGURE 4: Diagram of a dual-magnet speaker.

of the loop signals. *Figure 1* indicates the two signals have opposite signs—180° out of phase. *Figure 3* shows the correct phase relationships of the input and sensor signals at the summing amplifier. If the phase is incorrect, you must change it somewhere in the loop—usually by switching the leads at the speaker. *Figure 3* also illustrates how the amplitude and phase change as frequency varies from the maximum gain (central) frequency. Once the requirements of gain and phase are satisfied, you can connect the sensor to the summing amplifier to close the loop.

Record closed-loop data by feeding a signal to the loop at the point where the input signal normally goes to the summing amplifier. Measure the sensor voltage at the summing amplifier input (it represents the sound pressure). The response curve of the servo (20 * LOG(Sensor/Input)) should be a reasonably flat curve. If the rise at either end is more than 3dB, lower the loop gain. If only the low or high frequency has no rise, the loop compensation may be incorrect. Consider revising the compensation amplifier. The two designs I detail show examples of these criteria.

8. The input amplifier compensation depends on the sensor selected. An acceleration sensor needs no input compensation, a velocity sensor requires a 6dB attenuating curve, and a position sensor needs a 12dB attenuating curve. (You will see these curves in my description of each system.) You also should consider your crossover requirements and the means of balancing the drive between satellites and subwoofer. I provided gain adjustment in the input amplifier for this balancing operation.

9. At this point, I package everything into an integrated system for easy interconnection and control. I always re-run my open-loop gain and phase data on this package to ensure everything is correct before I close the loop. I have found wiring errors, and once I blew several ICs because I failed to check the phase. You could even destroy a power amplifier or speaker. It is also desirable to record the open-loop gains and phase for future troubleshooting.

10. Finally, I set the balance of the subwoofer sound pressure with that of the satellites. I adjusted the subwoofer input amplifier gain to obtain equal sound pressure readings at 12" in front of each speaker.

SERVO SPEAKER SYSTEMS. Sound pressure sensor method. I stated earlier I had built four servo speaker systems. I reported on the first in "Servo-

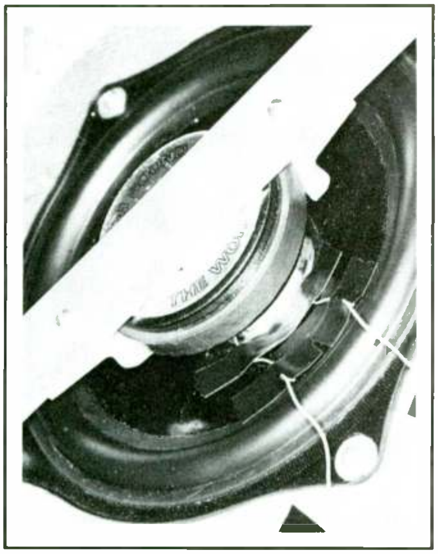

PHOTO 1: Dual-magnet speaker.

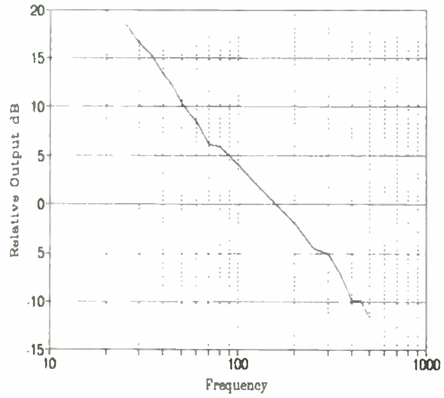

FIGURE 5: The characteristics curve of a PP10.2.

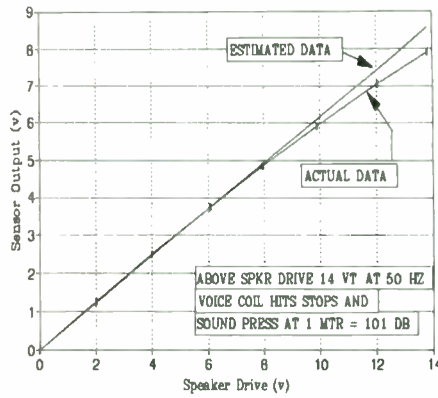

FIGURE 6: The sensor linearity of a PP10.2.

Controlling the AR-1" (*SB* 3/89, p. 24). This approach used the sound pressure generated by the driver and sensed by a microphone as the feedback method. Its principal benefit lies in using the parameter you wish to control—sound pressure.

One of its serious limitations, however, is the high sound-pressure level required by mounting the mike close to the speaker cone. Using a .3" (.76cm) mike distance, you can calculate a mike sound pressure level of 162dB for a maximum of 100dB at 1 meter from the speaker. The equation for this calculation follows: Maximum Sound Pressure (dB) = $(10 \times \log((1{,}000\text{cm} / .76\text{cm})^2)) + 100$. The way recorded material was reproduced on this system indicated the servo speaker technique worked.

Cone position sensing servo. Of the remaining three methods (measuring the cone position, velocity, and acceleration), the position sensor method demonstrated low distortion as one clear benefit of the technique. I used a Hall Effects Device (a sensor of magnetic field strength), a Texas Instruments linear sensing device, type TL173C. (This application requires a linear sensing device, not a switching device.)

I had hoped the Hall sensor would indicate the movement of the cone as it was exposed to the magnetic flux lines spreading between the poles of the magnet. My test, however, showed the magnetic flux varied with the position of the coil and with the speaker drive current. This was unsatisfactory.

I then mounted the Hall Device on the voice coil, but away from the driver magnet, so I could build a separate magnet structure. The test showed as major problems the low output of the Hall Device and its susceptibility to noise. Even though I had filtered the sensor carefully, the speaker system background noise was at an absolute level of 60dB in my living room—too high for a true high-fidelity system. Otherwise, its performance was clean and unobtrusive. I listened to this system for about a week.

Cone velocity sensing servo. I attempted to use the second coil of a dual voice coil/one magnet speaker (Madisound type 10204) to sense cone velocity. My tests showed that the sensing coil produced a voltage proportional to cone velocity *and* to the current in the driving coil. This was unacceptable.

My next configuration had a second coil on the driver coil form with its own magnet structure, located external to the driven coil magnet system (*Fig. 4* and *Photo 1*). I made the sensor coil longer (.8") than the driver coil to ensure I was sensing the cone velocity at all displacements. However, I was unable to eliminate the rubbing friction that destabil-

Continued on page 24

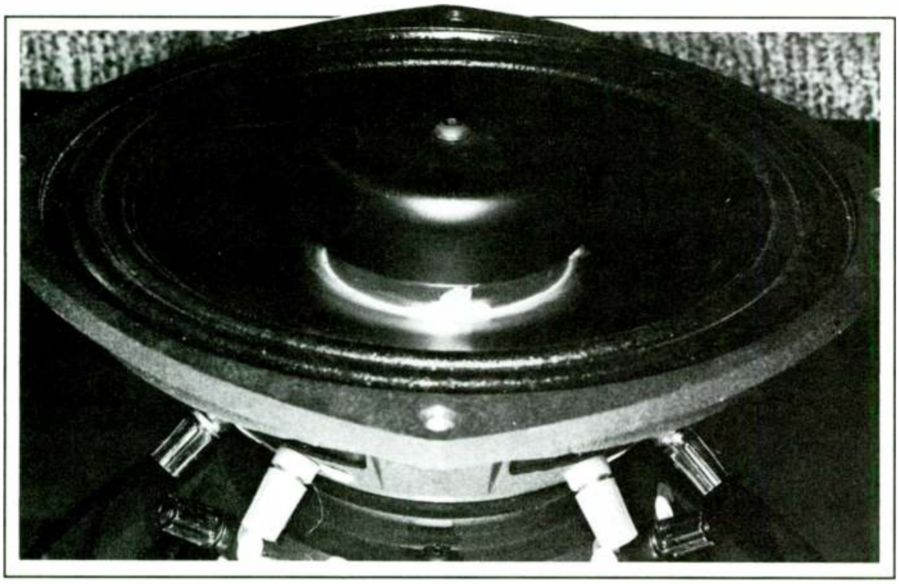

PHOTO 2: Morel PP10.2 dual-magnetic speaker.

TABLE 1

MOREL PP10.2 DUAL MAGNETIC SPEAKER SPECIFICATIONS PER COIL

	SPECIFIED	MEASURED
Power	100W	
Nominal Impedance	4Ω	
Sensitivity	92dB	
Freqency Response	25–1.5kHz	
RDC	3.7Ω	3.6Ω
Voice Coil Inductance	47mH	
Air Gap Length	6mm	
Voic Coil Length	14.5mm	
B x L	5.11WB/m	
Q_{MS}	1.46	1.708
Q_{ES}	1.15	.891
Q_T	.64	
V_{AS}	109 liter	50.176 liter
Effective Cone Area	380 sq. cm	
Moving Mass	43 gm	
R_{MEC}	6.13 ns/m	
Natural Frequency	33Hz	42Hz

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Watters Sound Wave Co. 4320 Spring Valley, Dallas, TX 75244 Tel: (214) 991-6994 Fax: (214) 991-5016
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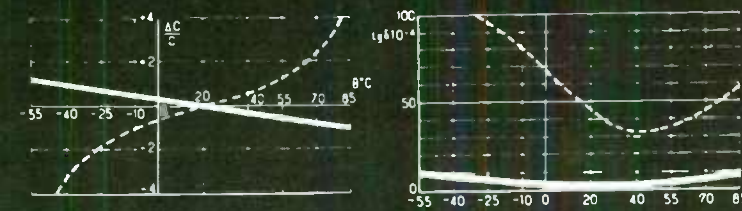
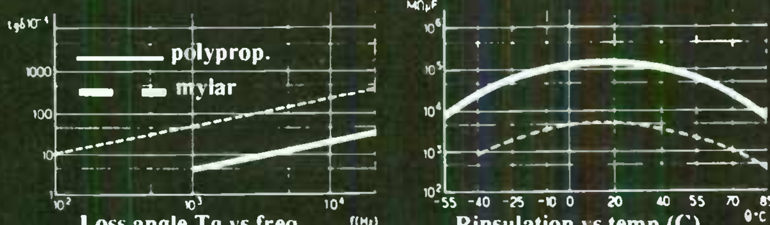
SCR metallized polypropylene capacitors
self-healing, non inductive and moistproof, these capacitors represent an exceptional value



Variation of capacitance vs temp.(C) **Loss angle Tg vs temp (C)**



Loss angle Tg vs freq. **Rinsulation vs temp (C)**

400 VAC/630VDC reinforced metal. polypropylene (tested at 1,000 VAC).
Faster rise time than lower voltage caps.
Ideally suited for X-overs as well as electronics.
5% precision
Ultra compact winding (stronger high voltage film allows a higher tension winding than lower voltage film) reduces microphonics. Polyurethane filled core and encapsulation.
Thick PVC heat shrunk envelope (not tape wrap).
14 AWG Teflon damped silver coated pure copper leads.
Dissipation factor at 20 C. 1 Khz : from .1 microF to 20 microF : Tg < 12. 10-4.

value	Diuameter	Length	retail $				$				$
010			1.45	3.0	.875	1.125	2.38	10.0	1.375	1.75	4.27
0.22			1.48	3.9	.937	1.5	2.67	12.0	1.375	2.5	4.83
0.33			1.53	4.7	1.0	1.5	2.86	15.0	1.375	2.5	5.49
0.47			1.58	5.6	1.0	1.625	3.06	20.0	1.5	2.5	6.10
1.0	.625	.875	1.94	6.8	1.0	1.75	3.37	Any special value upon request			
2.0	.75	1.125	2.16	8.0	1.125	1.75	3.74				

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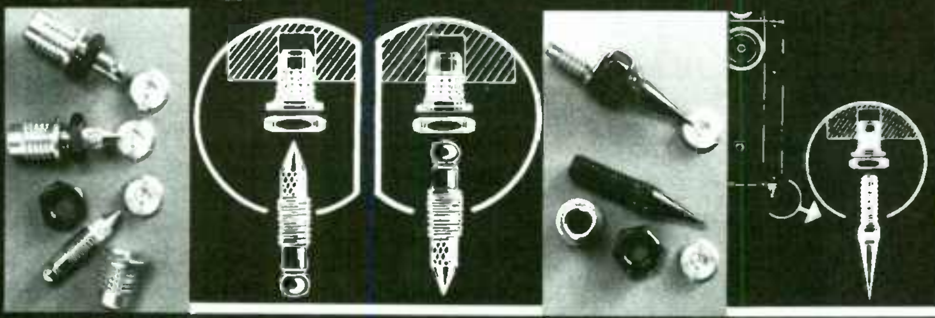

Part in bulk	suggested retail per part
Solid brass insert (hole diam.: 10mm = 3/8) (hole depth: 14mm = 9/16)	1.40
Black steel nut (8mm = 5/16)	0.15
Black steel single spike (8mm X 43mm)=(5/16 X 1.11/16)	1.25
Chrome steel double spike (8mm X 30mm)=(5/16 X 1.3/16)	1.85
Solid brass protector (diam. 13mm = 1/2)	0.85

FOCAL

Polished brass binding posts.
Accept RCA plug or up to 4 AWG naked wire.
Shaft is long enough for 3/4 in. wall thickness.
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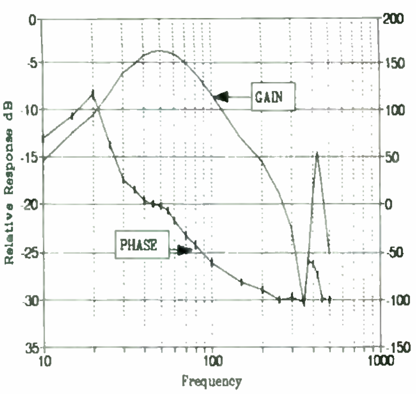

FIGURE 7: The PP10.2 speaker/sensor data.

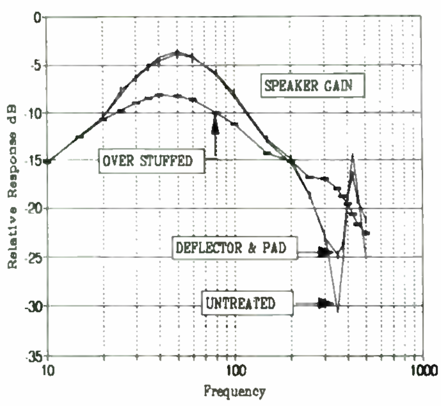

FIGURE 8: PP10.2 box configuration.

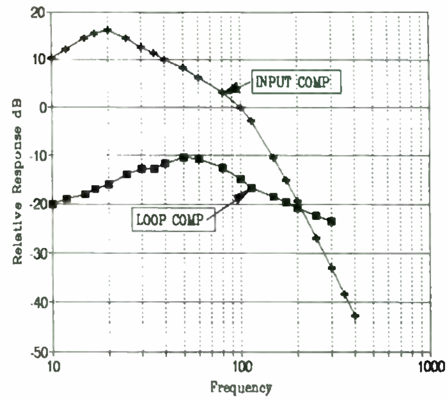

FIGURE 9: Loop and input compensation curves of the PP10.2.

Continued from page 22
ized the servo loop. I abandoned that buildup.

Morel makes speakers with dual voice coil/dual magnets and with single voice coil/single magnets. I chose the model PP10.2 from their dual coil/dual magnet line of speakers. *Photo 2* shows this speaker. In normal use, the dual coil/magnet functions the same as the dual voice-coil speakers—right and left stereo signals summing in the speaker. I used one coil/magnet to drive the speaker and the second to measure cone velocity.

Table 1 lists the published driver specifications and the parameters I measured after 24 hours of use. The values of V_{AS} and natural frequency varied considerably.

Figure 5 confirms the 6dB attenuation characteristics of a velocity sensor when the speaker generates a constant sound pressure. The linearity data shown in *Fig. 6* was run at 50Hz. Just above 14 drive volts, the speaker voice coil hit the stops of the magnet frame—one way to determine the travel limits. Fourteen volts into 4Ω equals 49W. The sound pressure at 1 meter was about 101dB at this input level. The deviation from the estimated linear data is smooth and should introduce little distortion—something I could live with.

The speaker/sensor gain and phase data (in a 2.6 cu. ft. enclosure) are shown in *Fig. 7*. The enclosure had no absorption material. I found the breakup in the curve was related to the enclosure dimensions, causing interference and reinforcement at 350 and 420Hz, respectively. I installed a temporary deflector to change the front-to-back reflection and absorption felt to change the lengthwise reflections. Then, after I recorded the gain data for this configuration, I filled the enclosure with long hair wool, polyester, and felt and repeated the gain data.

The data on all three configurations is shown in *Fig. 8*. The most significant change occurred when I stuffed the box. The 350 and 420Hz deviations are significantly smoothed and the gain curve is lowered and shifted. My first reaction was to use the stuffed configuration. I rejected the idea after I determined the required loop compensation. Using attenuation of the high frequencies to limit the reflection behavior of the untreated box was the remaining option.

I used the open-loop gain and phase data shown in *Fig. 7*, therefore, to develop the loop compensation—a second-order (53Hz) bandpass amplifier with damping of 1.3. The response provides 6dB per octave attenuating curves at each end of the frequency range (*Fig. 9*). This gave me the open-loop gain and phase data shown in *Fig. 10* where I obtained 0dB crossings at 15 and 180Hz. The phase shift from the central frequency of 55Hz was 130° at the low 0dB frequency and 125° at the high 0dB frequency. I closed the loop and obtained the loop (only) data in *Fig. 11*. A 1.5dB rise occurs at 12Hz and a 3dB increase at 200Hz.

The servo defined above is a velocity servo. The characteristic curve shown in *Fig. 5* is duplicated in the input amplifier with a second-order bandpass amplifier of 20Hz and a Q of 1. The response shown in *Fig. 9* between 20Hz and 100Hz is the 6dB per octave attenuating curve corresponding to the characteristic curve of *Fig. 5*. The response falloff above 100Hz of this amplifier results from the crossover compensation filter. The final system response curve (loop plus input amp) in *Fig. 11* is flat except for about a 1dB rise in the 18–25Hz range. The rumble filter effect is a result of the input bandpass amplifier where the curve below 20Hz is 12dB different from the curve above 20Hz. This attenuation, plus the inherent attenuation of the servo response, provides an effective rumble filter below approximately 18Hz.

The special electronics for this system are shown as a circuit diagram in *Fig. 12*. The circuit is built on a perfboard with point-to-point wiring and uses three operational amplifiers. I used Don Lancaster's *Active-Filter Cookbook* (Howard Sams, 1975) for all my designs and adapted circuits by selecting components for my specific design.

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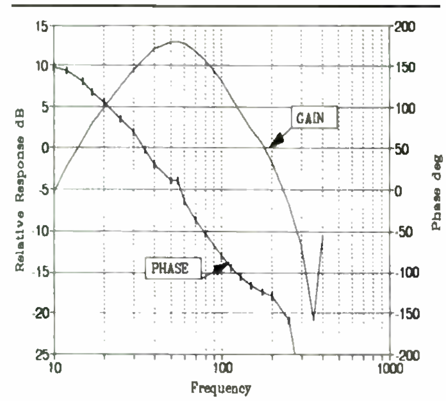

FIGURE 10: Servo loop performance of the PP10.2.

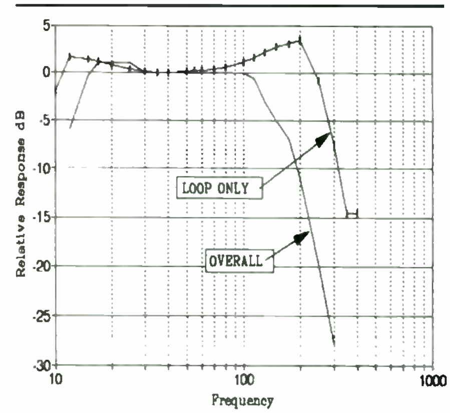

FIGURE 11: System performance of the PP10.2.

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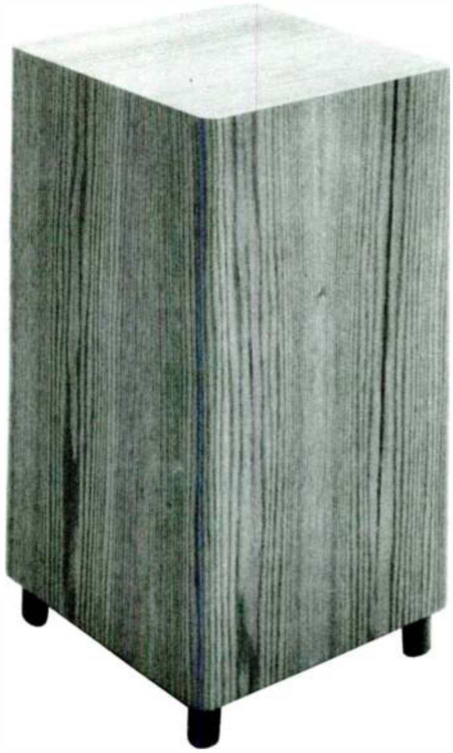

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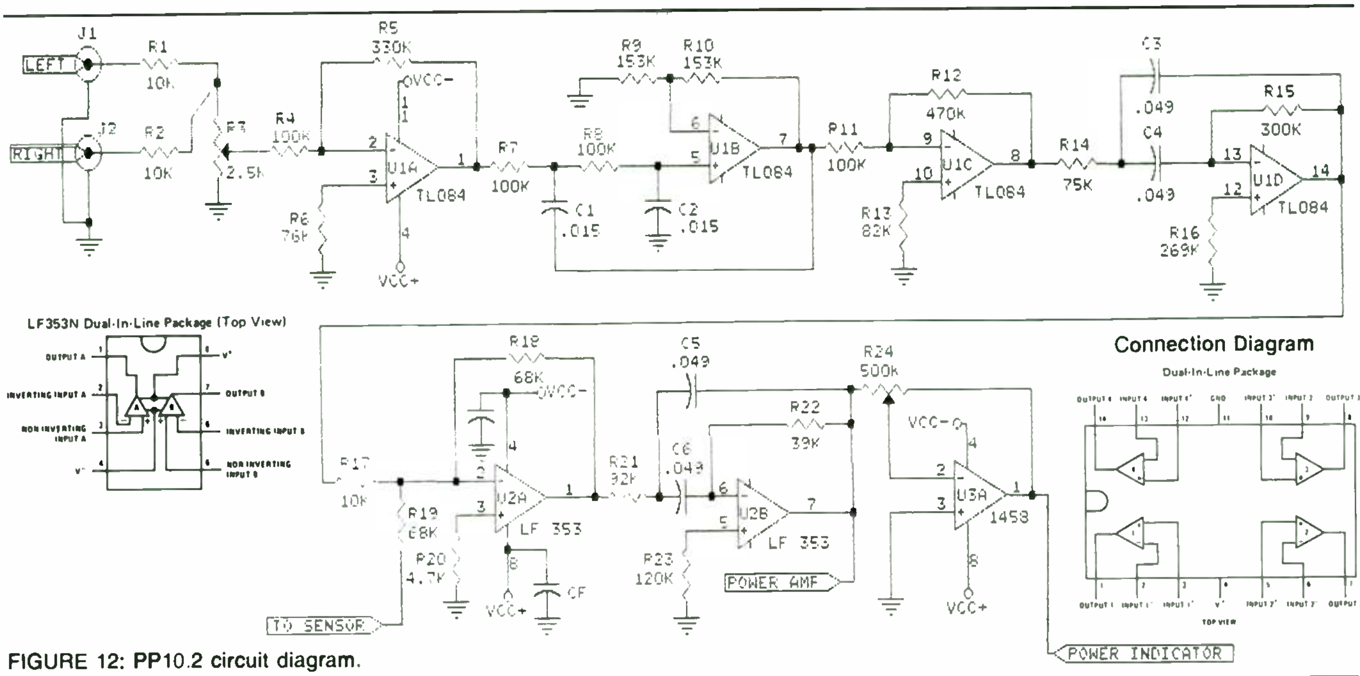

FIGURE 12: PP10.2 circuit diagram.

Continued from page 24

I used the LF353 operational amplifier for the compensation and summing amplifiers. The compensation amplifier is the second stage and is configured as a second-order bandpass amplifier with the break frequency and damping described above. The two capacitors and the input and feedback resistors determine the resonant frequency. The ratio of the two resistors determines the Q of the circuit (damping = 1/Q). The center frequency and Q were chosen to provide the gain and phase shift required by the specified open-loop response. In the summing stage, the sensor signal comes through the 68k resistor and the input signal comes through the 10k resistor. I chose the amplifier feedback resistor (68k) and the two input resistors to provide the relative signal levels and the loop gain I needed. The power amplifier has a gain control to provide the final loop gain adjustment.

Half of the 1458 dual-stage operational amplifier provides gain for the signal going to a power level indicator, a ten-segment LED with a LM-3915 driver for 3dB indicator steps. The power meter uses the signal going to the power amplifier. This signal represents the speaker drive voltage once the power amplifier

TABLE 2

PP10.2 CIRCUIT DIAGRAM PARTS LIST

ITEM	QTY.	REF.	PART
Resistors			
1	3	R1, 2, 17	10k
2	1	R3	2.5k
3	4	R4, 7, 8, 11	100k
4	1	R5	330k
5	1	R6	76k
6	2	R9, 10	153k
7	1	R12	470k
8	1	R13	82k
9	1	R14	75k
10	1	R15	300k
11	1	R16	269k
12	2	R18, 19	68k
13	1	R20	4.7k
14	1	R21	92k
15	1	R22	39k
16	1	R23	120k
17	1	R24	500k
Capacitors			
18	2	C1, 2	.015
19	4	C3–6	.049
Miscellaneous			
20	2	J1, 2	RCA jack
21	1	U1	TL084
22	1	U2	LF353
23	1	U3	1458

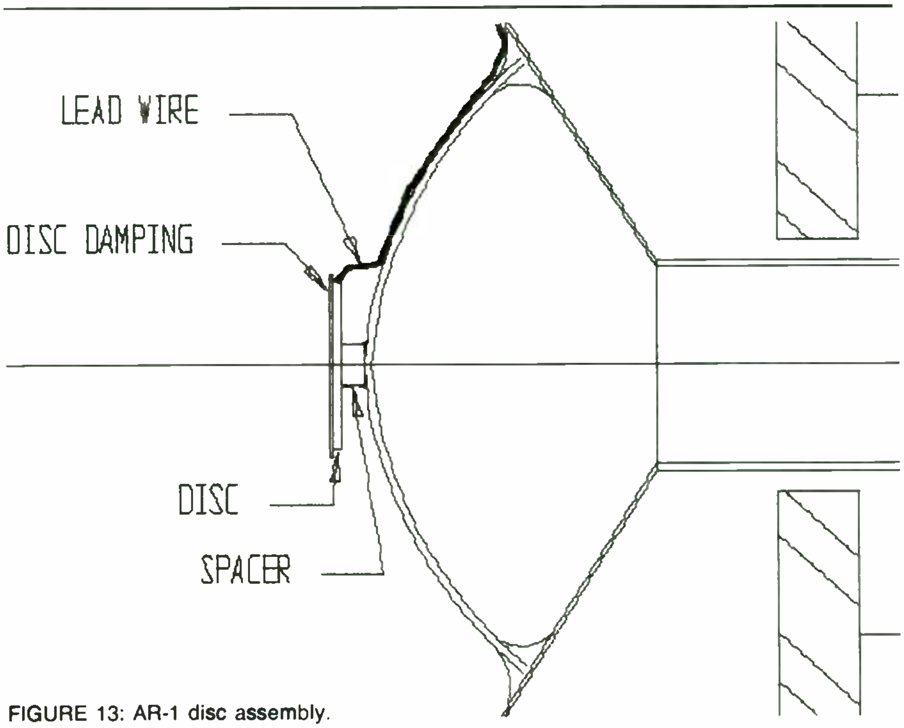

FIGURE 13: AR-1 disc assembly.

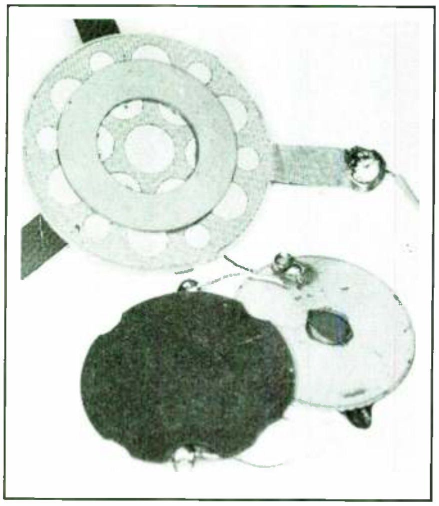

PHOTO 3: Disks from piezoelectric tweeters.

gain is set. The power meter was used in the absence of power-limiting circuits for the low frequencies.

The TL084 operational amplifier serves four functions. The first stage sums the right and left satellite signals and provides gain adjustment. The second stage second-order low-pass filter provides the 110Hz crossover function. The high-pass filter (110Hz) of the crossover function is the result of sizing the satellite enclosure (described in my "A Stereo System Odyssey" article mentioned earlier). The third stage provides gain and the fourth stage the integrating function needed to convert the input acceleration-related signal to velocity-related signal. It is a second-order bandpass network tuned to 20Hz and a Q of 1. This circuit also provides the rumble filter.

This velocity sensor servo speaker is a servo speaker and provides all of the performance benefits attributable to the servo approach discussed in the second paragraph of this article. I listened to this system for six months before I installed the acceleration sensor speaker system described below.

Cone Acceleration Sensor Servo. I undertook the acceleration-sensing system to prove an inexpensive acceleration sensor could be devised allowing any driver to be converted to servo action. The Morel PP10.2 cost me about $160, a significant investment especially if you have low-frequency speakers that could be modified. I was interested in developing something that did not need instrumentation accelerometers (normally

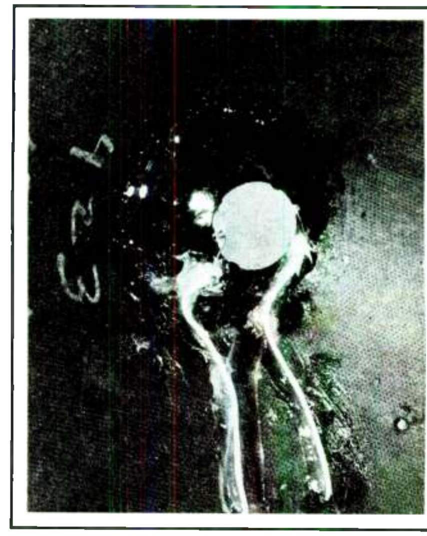

PHOTO 4: AR-1 disc mounted on the dust cover.

costing $500 or more). I was aware of the Velodyne servo subwoofers using the accelerometer as feedback. Other servo speakers were listed in the annual directory issue of *Audio* magazine, but with little information.

After I was well into this project, Hans Mortensen's article "An Acceleration

Travel Limits of Drivers

I often find myself looking at a new driver and wishing I knew the travel limits of the voice coil—the actual limit of travel or the power at the travel limit. The technique described here was suggested by my son, Jack L. Brown, of Irvine, California. I tested several drivers in this manner and feel it works well enough to set controls on speaker drive. My data on my AR-1 driver illustrates the idea.

The concept is based on the fact that a dynamic driver is a spring-mass resonant system. The displacement resulting from a driving force is mass-controlled above and compliance-controlled below its natural frequency. Below the resonant frequency, the mass term of the force equation is small compared to the compliance term. The following equation, therefore, provides an estimate of the behavior of the speaker at frequencies below its natural frequency (N = Newton, m = meters):

$$BLi\ (N) = F = x(m)\ /\ C(m/N)$$

The compliance (C) is essentially a constant. The force increases with current (i) until the coil reaches the displacement (x) where the coil begins to leave a constant magnetic field. Beyond this displacement, the BL product starts to fall off; it is no longer a constant. The force and the travel are no longer increasing proportionally to the current. The upper curve in *Fig. A* illustrates this. The illustration is simplified since the BL product does not change abruptly.

This modification in the travel pattern will show up in the sound pressure signal as a distortion, which you could observe with a distortion meter. We looked at the electrical output of the sound pressure meter and found a start of distortion similar to that in the lower curve in *Fig A*. The feedback sensor electrical signal should also show the distortion.

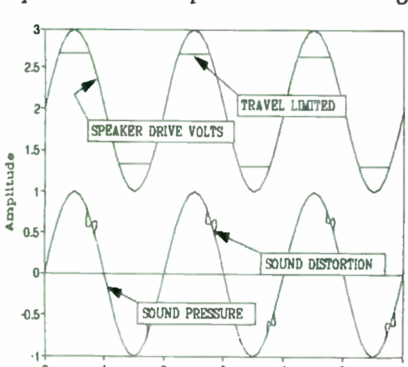

FIGURE A: Travel limits: typical response.

If you use the sensor signal, you may see the limit of the sensor capability or the drive coil travel. In either case, it is a limit you should observe in servo systems. I have not tested this sensor idea.

Using this concept, I drove the AR-1 low frequency driver in the enclosure with sine-wave signals. I looked for the start of this sound distortion at each frequency. I plotted on *Fig. B* the average of five voltage estimates of the start of distortion at each frequency. I visually judged the displacement of the cone to be the same at all the test points, about ½" peak to peak. I also plotted on *Fig. B* a 6dB per octave curve. The scale on the right represents the power to a 4Ω speaker. I used this data to set my power monitor.

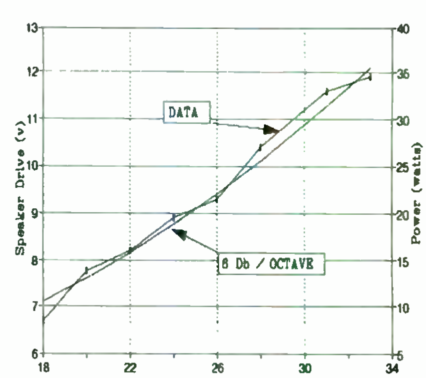

FIGURE B: AR-1 speaker: travel limits.

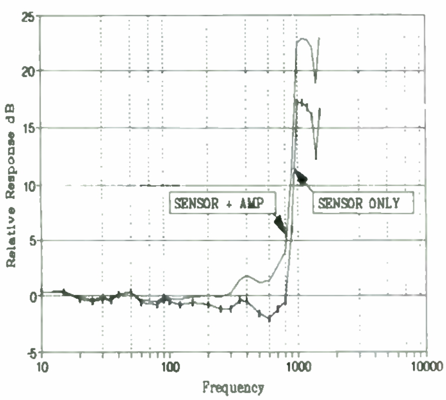

FIGURE 14: AR-1 sensor relative response.

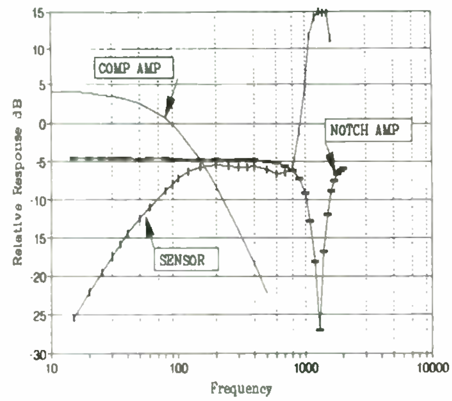

FIGURE 15: AR-1 Loop gain curves.

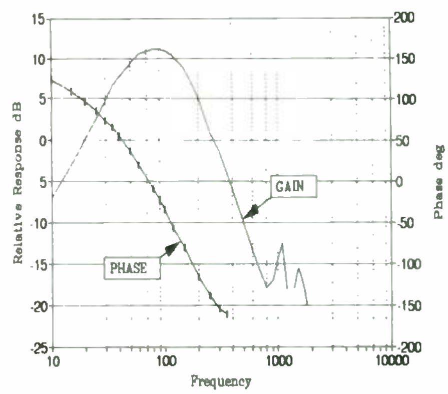

FIGURE 16: AR-1 open loop data.

Feedback System" (*SB* 1/90, p. 10) was published. His work covered the use of piezoelectric disc sensors taken from loudspeakers—just what I was doing. He also applied it to existing speakers.

I looked for commercial devices with usable piezoelectric elements. A piezoelectric phonograph cartridge was one possibility. Piezoelectric tweeters were another. The automotive industry was introducing accelerometers in their airbag systems suggesting another possibility. When I learned they were large and cost $160 as a replacement part, I gave up that idea. I tested a phonograph cartridge from my daughter's old record player. The speaker/sensor gain and phase data showed promise but varied considerably at 200Hz and higher. I attributed it to the mounting of the cartridge and to the behavior of the phonograph cartridge as a mass/spring system for measuring acceleration. I discarded it for the moment.

I purchased four piezoelectric tweeters from Radio Shack (part numbers 40-1397, 40-1379, 40-1396, and 40-1383) and removed the discs (*Photo 3*). All but one (40-1383) had rubber damping material attached to one side. I mounted one of the damped discs with a small spacer at the center to the edge of the voice coil of the 8N401 speaker. My test of this sensor showed good linearity, acceptable gain, and sensor response at constant sound pressure.

I then built up my selected speaker with a disc sensor. I used the AR-1 low-frequency driver in the AR-1 enclosure because I believed it had the long travel for low-frequency operation (see "Travel Limits of Drivers"). Since the driver had a 12" diameter, one should generate sufficient sound pressure for my application. I covered the voice coil dust cover with epoxy, making it a ridged dome. Then, I attached one of the small damped disks with a small spacer at its center to the dust cover.

Figure 13 and *Photo 4* show the disc mounted on the dust cover. As the cone moves, the disc edges flop back and forth, flexing the disc. The voltage generated by this flexing is proportional to the acceleration of the cone and thus the sound pressure generated by the speaker. The two lead wires were dressed with loops to allow disc flexing and then attached to the cone. They then passed through the cone to the charge amplifier I mounted to the speaker basket (*Photo 5*).

The sensor output at constant sound pressure (*Fig. 14*) determined the sensor would represent sound pressure reasonably well. I chose to add slight boost at the upper frequencies to adjust the response of the sensor (the sensor + amp curve in *Fig. 14*). The sharp rise at 1kHz results from the resonating of the disc at its natural frequency. In selecting a disc from a tweeter, it is desirable to choose

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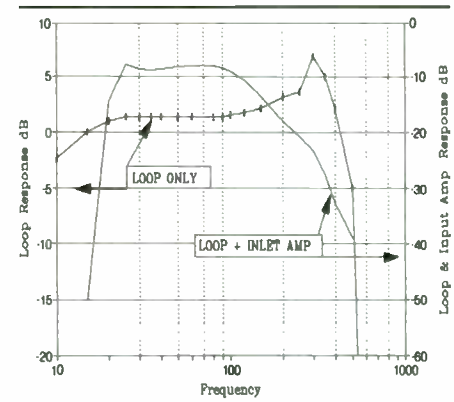

FIGURE 17: AR-1 closed loop data.

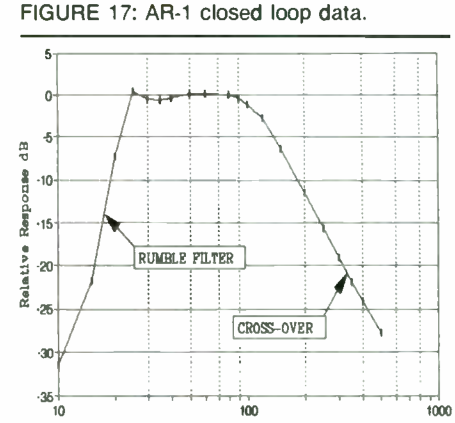

FIGURE 18: AR-1 input amp data.

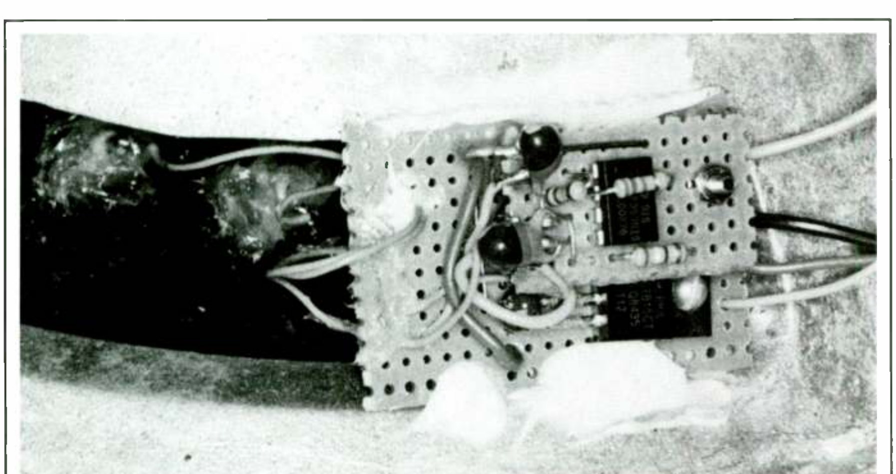
PHOTO 5: Lead wires attached to the cone and passed through the cone to charge amplifier.

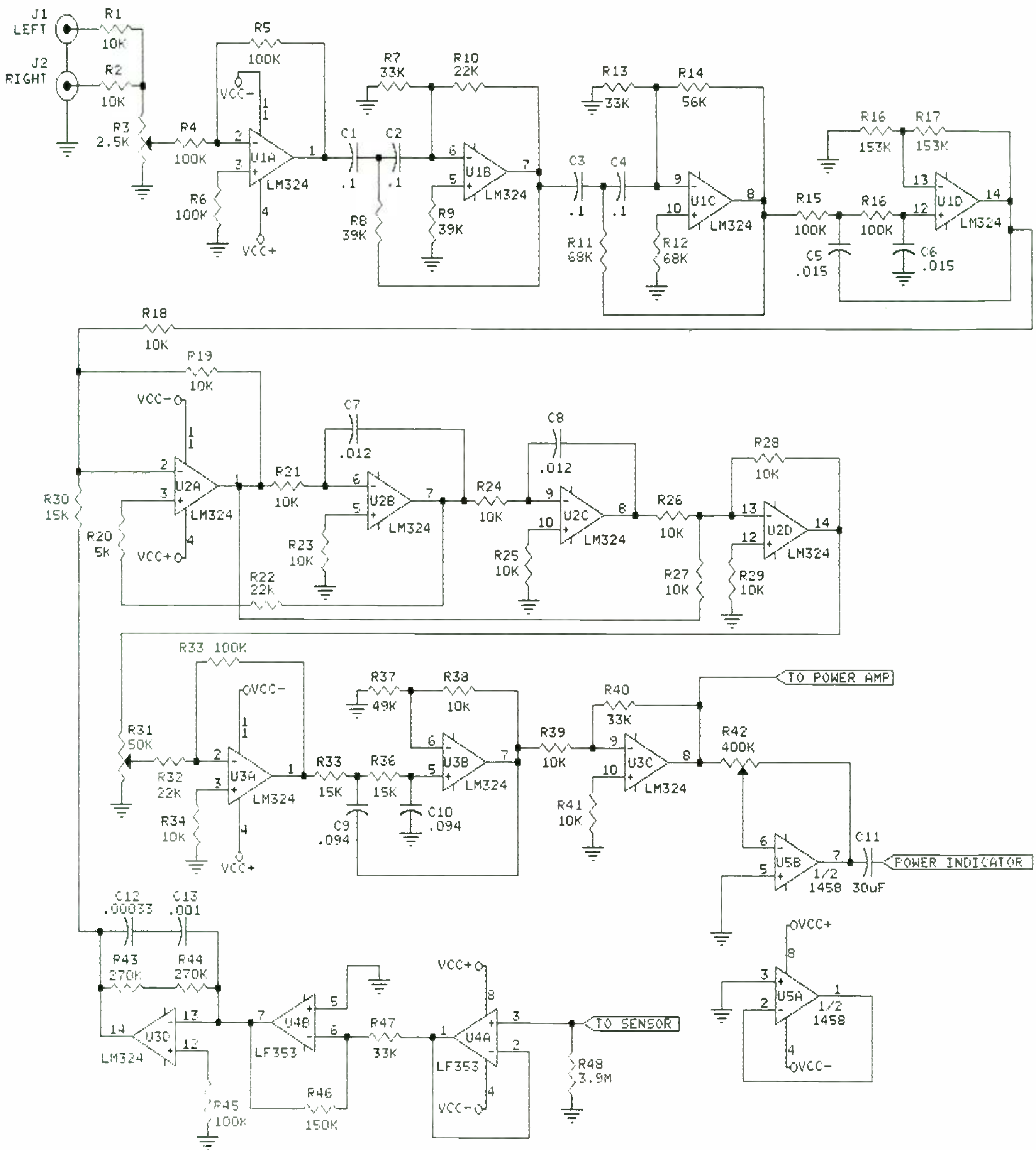

FIGURE 19: AR-1 circuit diagram.

Continued from page 28

a tweeter with the high published frequency response range of 5-20kHz rather than 2-20kHz. The natural frequency of the element is higher for the former. I used the disc from the Radio Shack tweeter (40-1397).

The speaker/sensor gain curve is shown in *Fig. 15*. I decided to use a notch filter to counter the gain rise at the disc resonance. The notch amplifier response also is shown in *Fig. 15* as is the compensation (comp amp) I designed for the loop. The open-loop gain and phase data resulting from combining the above loop responses is shown in *Fig. 16*. The notch filter took care of the resonant peak of the disc. The phase shift at 20Hz (the 0dB crossing point) is 100°. The 0dB phase shift at 280Hz is about 140°. I normally would recommend redesigning the loop compensation in this situation. I deviated from my rule and proceeded as below.

After verifying the correctness of the loop phase, I closed the loop with these settings and ran data. *Figure 17* shows this "loop only" data. The low frequency response is flat. I obtained a peak of about 5dB at about 300Hz. The combined response of this loop response and the input amplifier (*Fig. 18*) also is shown in *Fig. 17* as "loop + input amp" (scale

TABLE 3

AR-1 CIRCUIT DIAGRAM PARTS LIST

ITEM	QTY.	REF.	PART
Resistors			
1	16	R1, 2, 18, 19, 21, 23–29, 33, 38–40	10k
2	1	R3	2.5k
3	7	R4–6, 15, 16, 34, 45	100k
4	4	R7, 13, 41, 47	33k
5	2	R8, 9	39k
6	3	R10, 22, 32	22k
7	2	R11, 12	68k
8	1	R14	56k
9	2	R17, 49	153k
10	1	R20	5k
11	3	R30, 35, 36	15k
12	1	R31	50k
13	1	R37	49k
14	1	R42	400k
15	2	R43, 44	270k
16	1	R46	150k
17	1	R48	3.9M
Capacitors			
18	4	C1–4	.1
19	2	C5, 6	.015
20	2	C7, 8	.012
21	2	C9, 10	.094
22	1	C11	30μF
23	1	C12	.00033
24	1	C13	.001
Miscellaneous			
25	1	J1	left
26	1	J2	right
27	3	U1–3	LM324
28	1	U4	LF353
29	1	U5	1458

on the right). I concluded I could use this setup since the attenuation rate at crossover was close to the desired 12dB per octave before changing to 24dB per octave at 300Hz. In developing the input amplifier response (*Fig. 18*), I included a 24dB per octave rumble filter at about 22Hz as well as the crossover filter. The AR-1 speaker was uncontrolled at low frequencies.

I show my system circuit diagram in *Fig. 19*. The charge amplifier (an LF-353) shown at the bottom of the diagram is powered by a separate ±15V power supply mounted on the back of the speaker enclosure. The charge amplifier mounted on the driver frame uses a noninverting follower with a 3.9MΩ resistor at the input. This avoids losses due to the series capacitance (.13μF) of the sensor. The second stage provides gain for the sensor.

The remaining electronics were built on a perfboard with point-to-point wiring and placed in a separate cabinet with a ±15V power supply. The loop electronics consist of the notch filter, the loop compensation amplifier, and the feedback amplifier. These require the lower two quad LM-324 op amps. The notch amplifier was adapted directly from Lancaster, page 205. It required all four sections of one LM-324. I selected components to set the frequency and the Q of the notch as the best to handle the disc resonant peak.

The notch filter requires a summing stage (the first). I used this stage for summation of the input and feedback signals as well. The loop compensation amplifier uses three sections of an LM-324 and includes the input gain stage, the second-order low-pass filter compensation stage and another gain stage. The compensation stage is designed for 112Hz break frequency, with damping at maximum for the stage (D = 2). The fourth section of this LM-324 (bottom of the schematic) is used as the feedback amplifier. The compensation is a lead/lag circuit with break frequencies at 590 and 1,790Hz. The 1,790Hz lag is needed to avoid oscillation in the circuit.

The input amplifier (at the top of the schematic) uses all four stages of an LM-324. The first stage includes summing the input R and L stereo signals and gain adjustment. The second and third stages are the 22Hz high-pass fourth-order rumble filter. It was taken directly from Lancaster, with component adjustments for frequency and damping. The response curve in *Fig. 18* shows I chose a 1dB dip characteristic response. The fourth stage provides the 110Hz second-order low-pass crossover filter used on all my systems.

I used half of a 1458 amplifier to amplify a signal for the power meter. The maximum red LED is set to light at 50W. I chose this level because of the power limitation of the driver (see "Travel Limits of Speakers"). The power meter alerts me to possible damaging high power peaks. In my listening, I rarely notice power signals above −9dB of the max LED level. This power monitoring substitutes for automatic power limiting at low frequencies where you would expect to run into cone travel limits.

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AN APARTMENT TL

BY SCOTT S. ELLIS

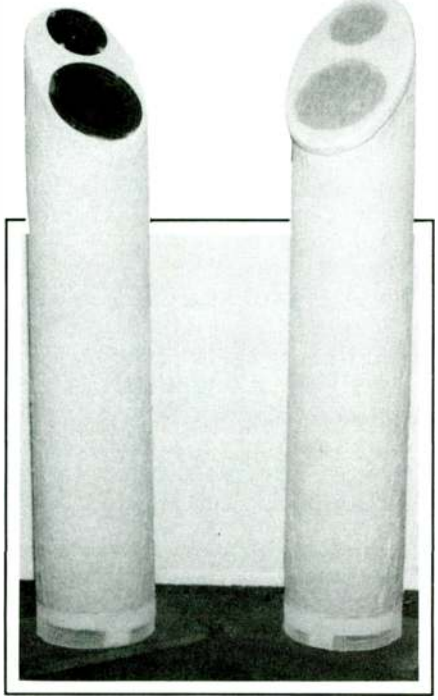

PHOTO 1: A pair of speakers suitable for an apartment.

Many urban Americans live in multi-family settings—apartments, townhouses, or condominiums with shared walls, floors, and ceilings. I am among them. When I moved to the Washington, DC, area, I left my monstrous-but-much-beloved curvilinear vertical arrays (CVAs, *SB* 2/85, p. 7) in the care of friends. I decided to build a pair of speakers that would satisfy me, conserve space, not irritate the neighbors, and be future ambience units.

The primary problem in apartment-noise management is preventing the bass from propagating through the walls and floors. Most apartments can easily contain sounds above 100Hz. Below that, the structure transmits sound fairly efficiently. High SPLs (arbitrarily defined as above 90dB at 1 meter) are also a problem. In my speakers, the lack of response below about 50Hz and the separation of the woofer from the floor mean little bass escapes to neighbors' apartments.

My design (*Photo 1*) is a transmission-line, with a nominal F_c of 60Hz and a two-way drive configuration. Opting for the lowest crossover, I chose a 2″ soft-dome tweeter (the VersaTronics PMK 50/130R) because its on-axis response purports to have a −3dB point at 14kHz, crossed-over at 1.2kHz.

The woofer is a Radio Shack 8″ type, (40-1016). I chose the paper cone version because most small polypropylene cones have severe resonances in the 1–2kHz region, the crossover passband. The paper version also has a slightly heavier magnet with the same voice coil. Treating the foam surround with Armor-All™ reduces the long-term damage from gas-heat fumes and seemed to make the surround more compliant.

Any problems at the response extremes can be managed as the associated amplifier has a 12dB/octave high-pass filter at 30Hz (low-level passive) and ±5dB control at 50Hz and 20kHz.

THE CONSTRUCTION PROCESS. The enclosure (*Fig. 1*) is a Perma-Form™ tube, 10″ in diameter, and the base is a limestone flagstone, 12″ x 18″ x 1″ thick. I glued three upright braces 120° apart between two stanchions epoxy-baked to the flagstone. Then, I coated the braces with Liquid Nails™ and slid the tube over them. The braces rise to the end of the tube, which I cut about 45° for proper temporal alignment.

I cut the ends of the braces to serve as cleats for the baffle (cut from ⅜″ plywood), and I added two extra "floating" cleats and glued them to the inside wall of the tube to secure the baffle better. Next, I cut the line exit from the tube base end and removed the sections between the brace stanchions to make the exit area equal to the woofer cone. I covered the exit with white plastic needlepoint "can-

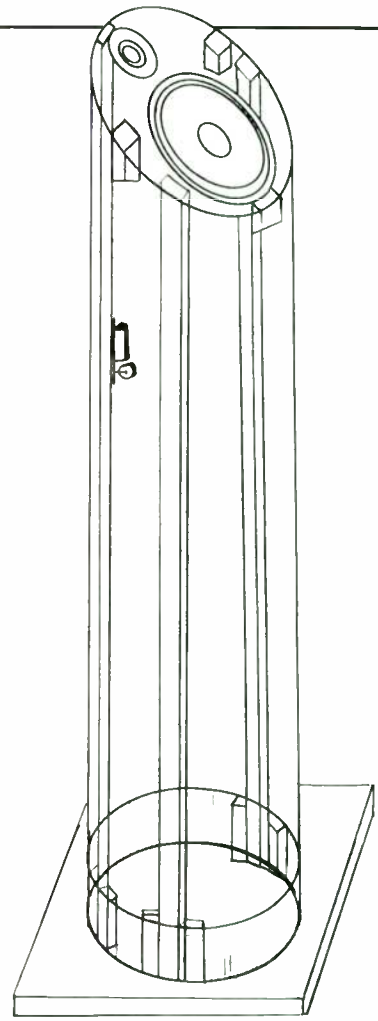

FIGURE 1: Outline of the enclosure (10″ tube diameter, 18″ x 12″ base, 56″ height).

ABOUT THE AUTHOR

Scott Ellis has been building speaker systems since 1972. He holds an Associates degree in electronics technology and is employed as a logistics consultant for government and industrial clients. Photography is his other hobby; this is his third article for Speaker Builder.

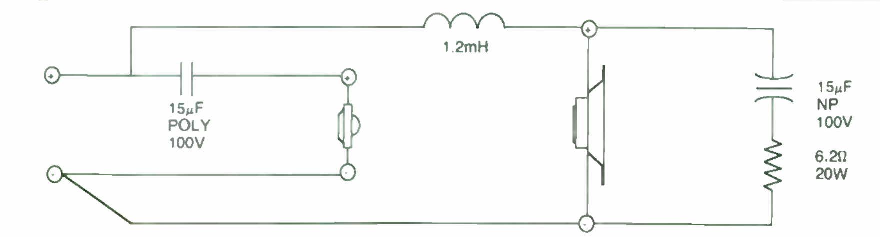

FIGURE 2: Diagram of the crossover.

vas" as a grille. The input is a two-terminal euro-style barrier strip glued into a notch in the back of the exit grille.

The tube walls are about ¼" thick and need damping to prevent flexure and transmitted resonances. I used roofing tar (messy but cheap) for the bottom two-thirds of the inside of the tubes and coated the top third of each tube, especially in the area behind the woofer, with glazing putty in a "blob and string" manner. The inside of the baffle uses a sheet of foam rubber carpet pad, and weatherstripping covers the woofer frame and magnet to suppress resonances further. I covered the back of the tweeter cup with tar-tape.

Next, I hardwired the crossover (*Fig. 2*) on a piece of masonite and secured it to a woodscrew driven into one of the uprights. I glued the Zobel network to the inside of the baffle. Then, I stuffed the tube with polyfil to a density of about ½ lb. per cubic foot, amounting to about 1.125 lbs. per tube. The next step was to cut grille frames from felt/foam typewriter pads and attach the cloth to the frames. The edges of the grilles were finished with iron-on cloth tape. I secured the grille assemblies with magnets mating with steel washers on the baffles.

Finally, I sanded the outside of the tubes to cut the wax coating, and then primed them with flat white spray paint. I finished the tubes with K-Lux™ texture wall covering—a thick, stonelike surface when dry—to help damp the tube. The K-Lux should be put on about ¼" thick; my first application was too thick and some areas were "starved" as the coat flowed down. As the K-Lux takes about 48 hours to dry completely, mistakes are easy to fix. After it dries to an off-white color, you can scrape off drips.

CONCLUSIONS. My biggest challenge was to keep from "gilding the lily" by adding features not included in my initial concept. In this way, I managed to complete the project for about $175.

In auditioning the system, I discovered more bass then I thought possible. Although it is not boomy or overbearing, I suspect the system is closer to a tuned pipe than to a true transmission line. The lack of bends in the line reduces the resistance the woofers see. The overall response is smooth; the top end lacks the crisp brilliance of ribbon tweeters but also minimizes source noises. Because of the angle and dispersion of the tweeters, a listening distance of 10–12 feet provides smoothest response and best imaging. The midrange is free of the resonances typical of small boxes.

I think it is better to have a single, broad resonance at the low end, where it can be rolled-off with a simple high-pass filter, than to have the usual midrange resonances typical of a rectangular box. The imaging is quite good, with a great deal of depth. This is largely due to the narrow dispersion of the relatively large tweeters. In sum, the performance is easy and unchallenging.

ACKNOWLEDGEMENT
The author wishes to thank Toutant Electronics (Cleveland, OH) for their assistance in tweeter selection.

A SECOND-ORDER L-R CROSSOVER FOR THE SWAN IV

BY V.H. ESTRICK

I find I learn a great deal (or at least uncover a lot of questions) when I design or construct a project, as do most *Speaker Builder* readers. After I built the Swan IV speaker system (*SB* 4/88, p. 9 and 5/88, p. 34), I queried Joe D'Appolito about the crossover (*SB* 4/89, p. 69). He replied that the redesign of the Swan IV Pedal Coupler accommodates the phase shift and amplitude roll-off of the satellite. The effect of the satellite response and the original single-pole crossover was to produce a dip in the on-axis response. The new design is an in-phase second-order Linkwitz-Riley (L-R) configuration using the satellite response to approximate one of the poles of the high-pass filter. D'Appolito said a commercial unit from Whale Cove Audio uses this arrangement. Joe Curcio's Pedal Coupler II is also available. However, I do not know what configuration this version uses. Either unit undoubtably would be satisfactory.

For those who like to "roll their own," however, I am offering my version of a second-order L-R crossover. It also uses the satellite response as a single-pole high-pass with electronics to complete the crossover. This combination, which approximates an on-axis acoustic response of the L-R, is possible because of the low Q (Q_{TC} = 0.35) of the satellite system.[1] Even though the theoretical response of this closed-box system is a second-order filter, the low Q makes it behave like a single-pole system in the 200Hz crossover frequency region. Since you can obtain a second-order L-R by cascading two identical first-order filters,[2] the combination of the satellite and a single pole can give reasonable results. The transfer functions for the second-order L-R are given by the following:

$$H_{LP} = 1 / s^2+2s+1 = (1/s+1)(1/s+1)$$

and

$$H_{HP} = -s^2 / s^2+2s+1 = (s/s+1)(s/s+1)$$

Therefore, you can obtain the low-pass function by constructing a two-pole filter with a Q of 0.5 or two cascaded one-pole filters, each having a 200Hz corner frequency. You obtain the high-pass function with the cascade of a one-pole high-pass filter and the satellite acoustic response.

I modeled the on-axis response of the complete system using a Microsoft Excel™ simulation. It uses the theoretical transfer functions of the crossover filters and the satellite, plus the effect of the 4mH boost coil on the satellite phase response. It also models the effect of the distance between the acoustic centers of the satellite and the bass drivers, a necessary consideration since the satellite needs to be near the front of the bass cabinet to ensure minimal diffraction. Therefore, the midrange and woofers will not be "time aligned."

Figure 1 shows the predicted frequency response of the system and the individual filter and satellite responses. The on-axis response is vector addition of the low-pass and combination high-pass outputs for the bass unit 3″ behind the satellite acoustic center or "zero delay" plane. Interestingly, a 3″ simulated difference had a more nearly flat response than a delay of zero, no doubt due to the satellite response only approximating a first-order filter.

Figure 2 compares the predicted

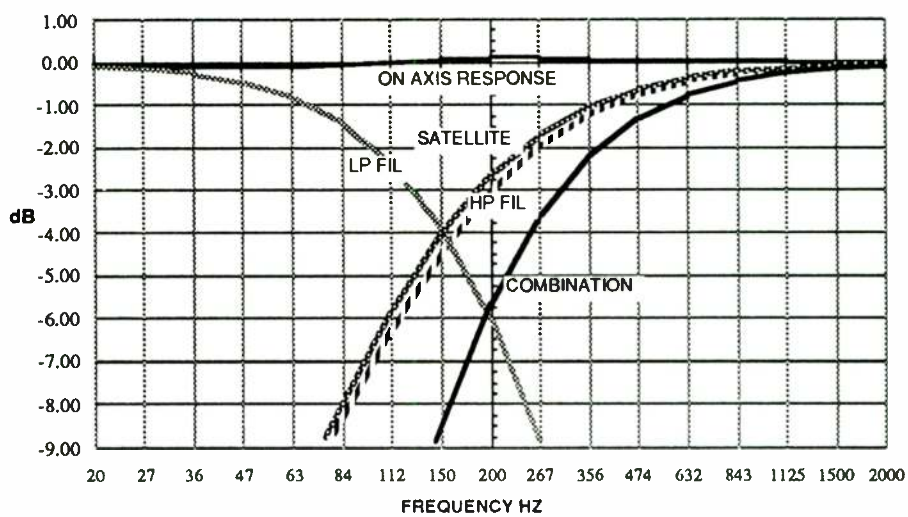

FIGURE 1: Second-order Linkwitz-Riley crossover. The high-pass combination is comprised of a single-pole filter and the acoustic response of the satellite.

ABOUT THE AUTHOR

V.H. Estrick is an engineer at Hughes Aircraft Co. in Fullerton, California in a radar design group. He has been an audio hobbyist since he built his first Heathkit in 1960 and has constructed several of the projects in Speaker Builder *and* Audio Amateur. *He has a wife and two children (plus a grumpy cat) who all quizzically allow him to indulge in his hobby.*

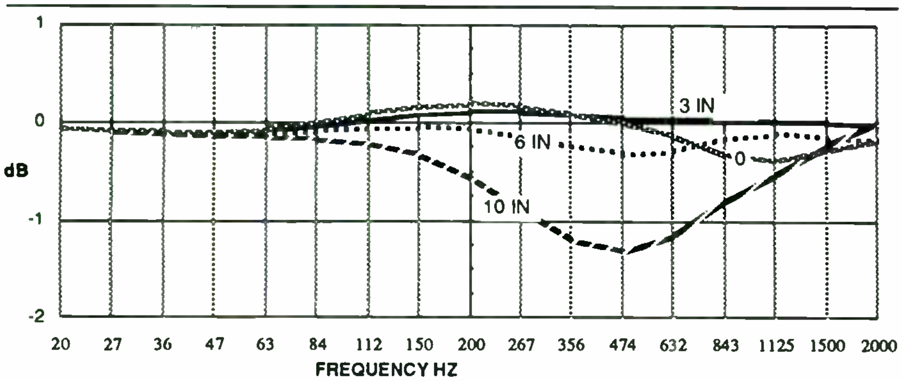

FIGURE 2: Response vs. driver offset.

response for 0, 3, 6 and 10 inches of driver offset and indicates that the ripple will be less than 0.5dB as long as the difference is between 0 and 6 inches. This probably means positioning the satellite relative to the bass unit is not critical when using this crossover.

I also investigated other crossover orders including a four-pole low-pass, three-pole high-pass to approximate a fourth-order Linkwitz-Riley using the satellite for the fourth pole. The second-order L-R gave the least ripple of any of the configurations and is easier to implement than higher order filters.

The schematic for the crossover is shown in *Fig. 3*. You obtain the low-pass function by the cascade of the 200Hz active one-pole with the 20k resistor and the 0.039μF capacitor. The second pole is obtained with the 200Hz passive filter formed by the 1.69k resistor and the 0.47μF capacitor. The second op amp is the bass equalizer similar to the original Pedal Coupler and has the boost reduced to about 4.5dB as recommended by D'Appolito for use with the 10″ Eclipse drivers. You get the high-pass one-pole at 200Hz with a single capacitor in series with the input to the satellite power amplifier. You can calculate the value of this capacitor using the formula

$$C = 1 / (2p \times 200 \times R_{IN})$$

where R_{IN} is the input impedance of the satellite power amplifier. Make certain you get the actual input impedance of the amplifier.

The Hafler XL280 amplifier in my system has a 47k specified input impedance. When I used this value to calculate the capacitor, the measured corner frequency was too high. Further examination of the XL280 schematic showed a DC-blocked 110k resistor in parallel with the 47k, making the actual AC impedance 32k. The required capacitor is 0.025μF for -3dB at 200Hz, a value obtained by paralleling 0.022 and 0.0003μF film capacitors. This calculation assumes the output impedance of the preamplifier or other circuit driving the crossover is much lower than the power amplifier input impedance. If the driving impedance is more than a few hundred ohms, its value should be added to R_{IN} prior to calculation.

This circuit is a little simpler than the original Pedal Coupler and should appeal to those who worry about adding extra ICs in the signal path. The high-pass filter adds only a single capacitor. It can be as expensive as desired for anyone believing this will improve the sound.

I constructed the crossover on the Lampton-Zukauckas high-level circuit board (*TAA* 1/79, p. 5) using locations A3 and A4 for the bass equalizer portion and A5 and A6 for the input filter. I have used this board for several projects because of its versatility; it has room for large active filter capacitors and has lots of ground plane. This latter feature is important; adequate grounding and careful EMI/RFI are often ignored in audio.

Use of this board requires additional jumper wiring to complete the circuit properly, but the changes are not difficult since the crossover is fairly simple. I used AD711s op amps, but any respectable device should work since this portion of the crossover covers the lower frequency of the audio spectrum. You can locate the high-pass filter capacitor on the crossover board or you can install it inside the satellite amplifier chassis. I used a Borbely preamplifier DC supply to power the circuit.

How well did it work? I have no way of measuring the on-axis response of the system accurately, but listening to music indicated the theoretical response was smoother. The only compromise for this crossover is the phase reversal required between the bass and satellite, plus the

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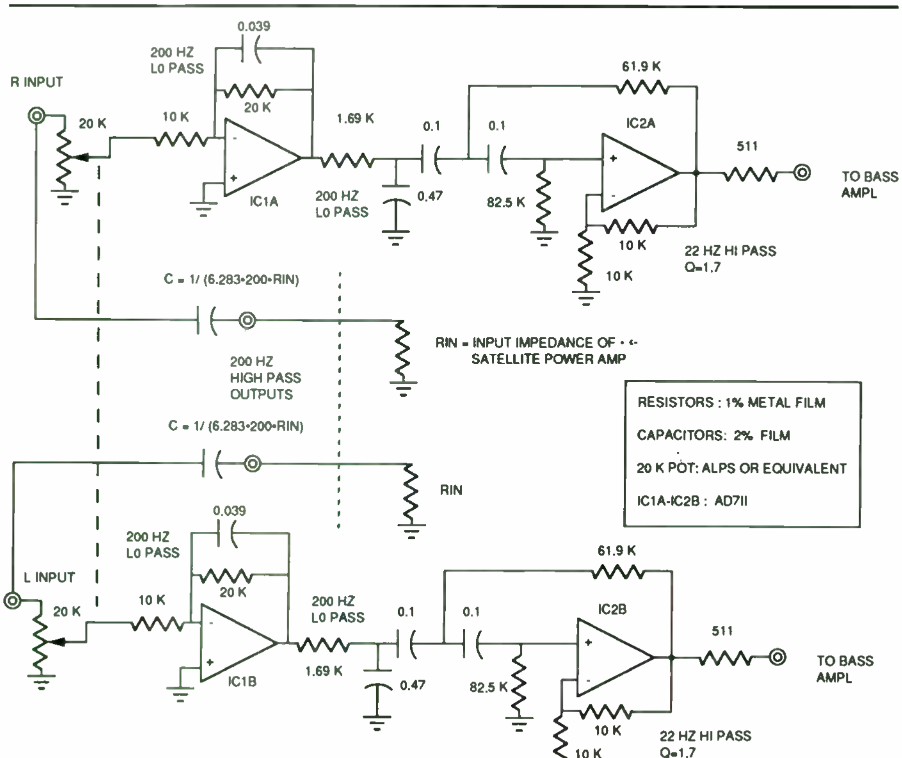


FIGURE 3: Modified crossover for the Swan IV.

REFERENCES

1. D'Appolito, J.A., "The Swan IV Speaker System," *SB* 4/88, p. 12

2. Chalupa, R., "A Subtractive Implementation of the Linkwitz-Riley Crossover Design," *Journal* of the Audio Engineering Society, 1986 July/August, p. 557.

MORE POWER FOR LESS

BY MARK GADZIKOWSKI

When my NAD 3140 integrated amplifier went kaput after seven years of nearly continuous use, I thought about buying a cheap receiver until it was repaired. After a discouraging trip to a local Best Buy, I read a Radio Shack flyer in which the 20W integrated circuit (IC) amplifier chips I had been eyeing were on sale. I figured I could build an interim amp for less than $100.

All the components in my amplifier are available at Radio Shack. The total cost of the components, from cabinet to ICs, is less than $70. Even with a power supply, it costs less than $35 per channel, and is even less for people who have spare parts kicking around. A stereo bi-amp (or even a tri-amp) for an active crossover system could be constructed from this circuit for less than $200.

ABOUT THE AUTHOR

Mark Gadzikowski has recently completed an MA in English at Iowa State University. He currently supports himself selling custom loudspeakers and audio electronics. This is his first article in Speaker Builder.

CONSTRUCTION SIGHT. The construction is so simple that even someone who has never built a kit can complete this project. Only a few components are required—one IC with seven capacitors and four resistors per channel. The circuit is tolerant of minor component variances and forgiving of my "solder first, test later" approach.

BOARD ALREADY? The schematic for the amplifier is shown in *Fig. 1.* Note both channels are identical, and both need a power supply (*Fig. 2*). They may share a single power supply and ground. My permanent prototype was built on two one-sided circuit boards, each about 3 inches square. If you know what you're doing, you might be able to build the amp to fit inside an existing nonamplified component (such as a mixer, preamp, or tuner) or to build a small portable amp.

For beginners, I recommend a circuit board with foil traces. Those with breadboarding experience may be comfortable

TABLE 1

20W POWER AMPLIFIER PARTS LIST

Part Rating	Resistors' Number
C1, 8	1.0μF Film Pc 272-1055 (2)
C2, 9	220pF Elec Pc 272-124
C3, 10	470pF Elec Pc 272-125
C4, 11	47μF Elec Pc 272-1027
C5, 12	100μF Elec Pc 272-1028
C6, 13	10μF Elec Ax 272-1013
C7, 14–17	2,200μF Elec Ax 272-1048
CR1	50 PIV 276-1161
IC1, 2	TDAI52O A 276-1305
R1, 5	270Ω 271-016 (2)
R2, 3, 6, 7	22kΩ 271-038 (2)
R4, 8	680Ω 271-O21 (2)
T1	120/25.2V AC 273-1366
Heatsinks	TO-220 case 276-1363 (2)
Grease for heatsinks	276-1372

Miscellaneous and Optional Parts

Case 2″ × 8″ × 6″ HWD 270-272
Fan 32 CFM 273-242
Fuse holder chassis mount 270-739 (2)
Fuses 1.5A 270-1274 (3)
IC socket 18-pin DIP 276-1992
Input jack RCA 274-346 (4)
PC board with foil 276-154, without foil 276-1396 Speaker jack spring clips 274-621 (1 per channel)

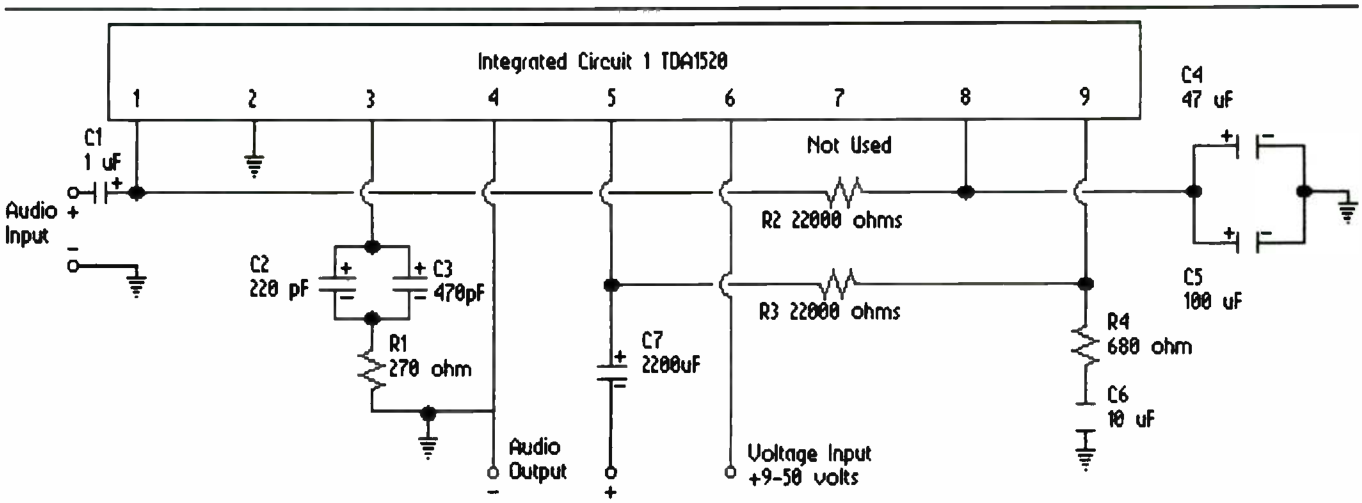


FIGURE 1: Schematic diagram of a 20W power amplifier. (Two required for stereo).

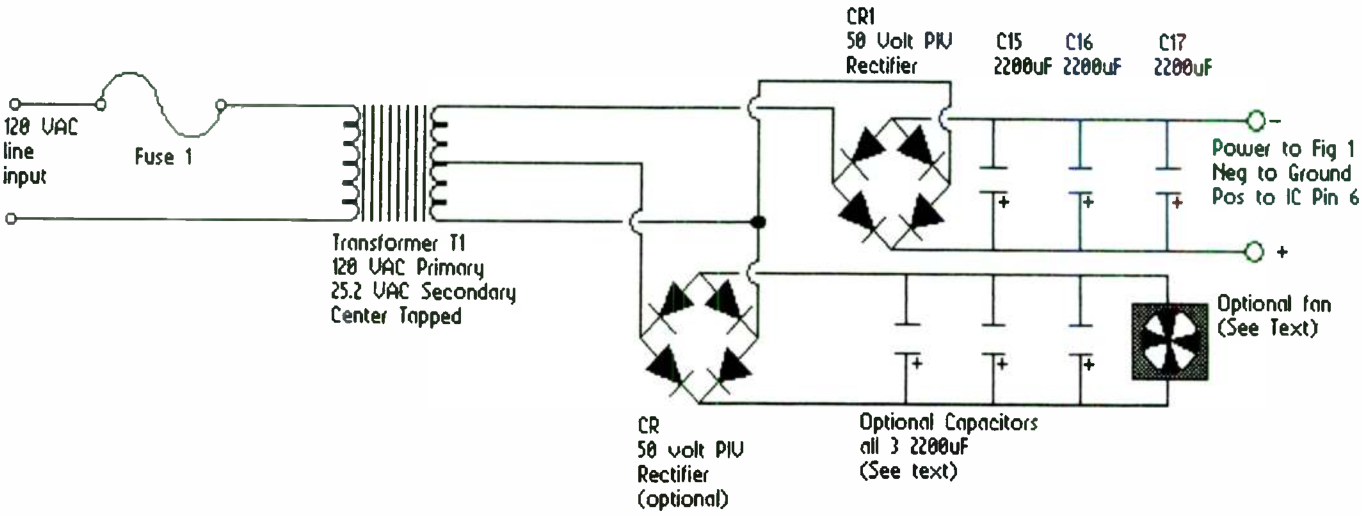


FIGURE 2: Schematic diagram of the power supply. The fan is optional.

building on perfboard, but an experimenter's PC board is easier to work with. On a PC board, I don't have to solder jumpers between components. I can solder the components quickly for optimum placement (see *Fig. 3* for an arbitrary parts layout) and then worry about connecting them later. Thus I avoid soldering any component twice and possibly damaging it. You may want to design your own layout, since mine wastes a lot of space. After I was halfway through the project, I realized everything would have to fit on one circuit board with room to spare. A custom board may be available by the time you read this, but if not, you can get everything you need to build your own amplifier at Radio Shack.

SOCKETS THAT MATCH. Rather than use the 18-pin sockets in the parts list, I used two 16-pin IC dual-in-line/package sockets for my prototype for three reasons. First, I never solder directly to an IC. Second, pin 7 is not used on the ICs, so one side of a 16-pin socket can be used for the remaining eight leads if they are bent carefully to fit (*Fig. 4*). Third, one socket could not hold both ICs and the heatsinks I used. *Figure 5* offers an alternative design. The sockets hold my rather weighty assembly (IC, heatsink, and mounting hardware) surprisingly firmly.

HOT TUNES. Dissipating the IC-generated heat is important because the total power produced (and the general circuit life) is limited by overheating. The standard case heatsinks for a TO220 worked well for me. I had to drill two holes in each (*Fig. 6*) to make heatsinks designed for a TO220 fit the TDA1520 single-in-line package. Before you tighten the nuts and bolts, use heat-conductive grease to ensure optimal heat transfer to the sinks. Wipe off all excess grease, as it conducts electricity as well.

With careful circuit planning and layout, you could make the ICs and heatsinks rest against the wall of the cabinet for additional dissipation (*Fig. 7*). The sinks could be omitted if the ICs were mounted in a metal enclosure. Don't forget that the enclosure will be electrically grounded if the ICs are put against a wall. It's probably a good idea to ground the cabinet as shielding against radio-frequency interference (RFI).

I have not had any problems with heat dissipation, even when the amplifier is driven hard for long periods, but I have a 4-inch muffin fan bolted to the top of my amp (*Fig. 8a*). I could have been more stylish, but I wanted to finish my project quickly and never have to worry about the heat buildup in the case.

Radio Shack sells a fan that is less than 2 by 2 inches, can run off 12V DC, and can be mounted discreetly on the side or

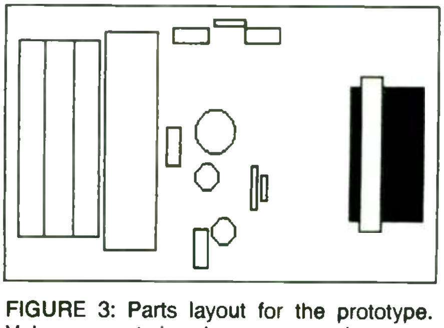


FIGURE 3: Parts layout for the prototype. Values are not given because part placement is not critical.

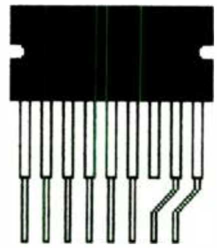


FIGURE 4: Pin bending form for the 16-pin IC socket.

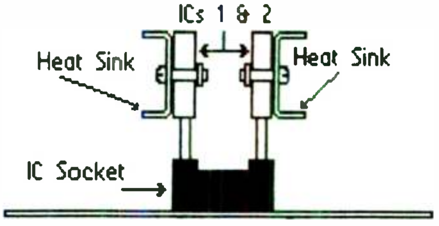


FIGURE 5: Possible IC socket mounting.

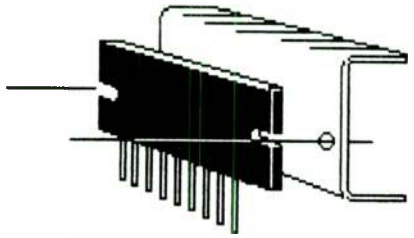


FIGURE 6: Holes to be drilled for using the TO220 heatsink with a 9-pin IC.

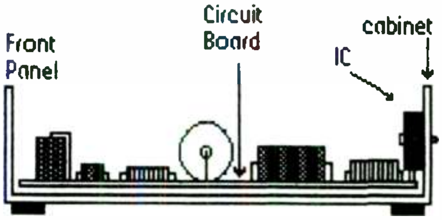


FIGURE 7: Side view of the enclosure when used as a heatsink for the IC.

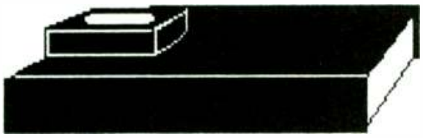


FIGURE 8a: Chimney fan.

FIGURE 8b: Rear fan.

rear of the enclosure (*Fig. 8b*). Another rectifier and more filter capacitors would be needed to power a 12V DC fan. Wire the second rectifier as in the schematic, but use the 12V center tap of the transformer. Do not run a 12V fan from the rectified 25.2V output.

A fan will increase the total cost to $80. Weigh the cost against your aesthetic requirements and normal room temperature. I'm not sure you need one, but I've never tried using the amp without it.

CHARGED UP. I made certain to use capacitors rated at 50V. Since the ICs are rated at 50V maximum and I had no specific power supply in mind when I started, this seemed like a good idea. If you're certain that you'll never power the circuit with voltage that high, you might be able to use lower voltage (and possibly less expensive) capacitors. If you use the recommended transformer, 35V capacitors are satisfactory.

All but two of the capacitors (C1 and C8) are axial electrolytics. Be sure the positive side of each is connected to the positive side of the circuit. Usually just one side of the capacitor is marked.

The schematic supplied with the ICs called for capacitance values that Radio Shack does not stock, so I used smaller capacitors in parallel for C2 and C3 and for C4 and C5, as well as their counterparts on the second channel. I had no problems with the small difference in capacitance.

VIVA LA RESISTANCE! The Radio Shack schematic shows 2.7Ω in series with 0.1μF capacitor between IC pins 4 and 5, the speaker outputs. I omitted these parts in my circuit because I couldn't get my amp to work with them. I think the resistors made the output load drop too low for the IC (5.5Ω with an 8Ω speaker connected) and caused strange clicking and squealing noises instead of the proper output.

Even without the resistors, I couldn't drive my 4Ω Klipsch KG2s properly or two pairs of 8Ω EPI100 speakers in parallel. The Klipsch speakers were fine at low volumes, but the amp clicked and squealed when I tried to turn up the sound. High volume levels are no problem with higher resistance speakers. (My housemates rejoiced when I put the Klipsch on our "public" stereo and took the 8Ω EPI100s into my room.) The prototype now runs two pairs of 8Ω speakers in series.

POWERFUL OPTIONS. During construction, I powered the amp with a 9V DC adapter from a Timex Sinclair Computer. I had no problem with the low voltage, even though the supplied data sheet for the ICs says they need 15V to 50V DC. I did notice a small difference in maximum volume when I changed to a 35V supply.

Also, the power kept going out on me. I checked the solder connections again and again and substituted several different rectifiers. As I completed the enclosure, I had so much trouble with what I subsequently discovered was a bad 120V to 25.2V transformer that I ended up placing a DC input jack on the enclosure so I wouldn't have to keep opening it.

I am now taking 35V from an old Pioneer reverb unit. Don't try this unless you're confident of your skills. You need to be able to recognize the layout of the external unit's power supply, most notably the points after the rectifier. A mistake with power in an external unit could damage both units (not to mention yourself) irreparably. Using another audio unit for power also restricts operation of the amp to the area where the other unit is located.

For safety's sake, use a fuse whenever you run 120V into an enclosure. I cannot emphasize this enough. Fuses have saved my projects and my flesh on more occasions than I care to admit.

I didn't have any problems with hum, even when the AC wires were very near the circuit boards. The power supply has what may be overcautious filter capacitors. My childhood 5V transistor-transistor logic (TTL) supply, from which I borrowed the circuit design, has only three 1,000μF capacitors, rather than the three 2,200μF capacitors in my schematic, but I'm paranoid about hum. You may want to use separate power supplies for each channel. I didn't have the inclination or the parts to do so.

THE HOLE THING. I don't own a drill, so mounting hardware on the cabinet was a chore. For holes ¼ inch or more in diameter, I used a hand reamer. I decided to omit all superfluous items, such as a power switch and volume controls. This still left me with two speaker terminal boards with screws, two input jacks, a power input jack, and three holes to mount each circuit board in the enclosure.

I have a scroll saw with a snake-like dentist's drill attachment that holds drill bits up to ⅛ inch diameter. Unfortunately, all the holes I needed (except the ones for the mounting screws) were bigger than that. Using a small drill to start those holes was better than having to start them with the hand reamer, but it was no fun.

Caveat: If you have a hand reamer, be sure to wear a glove when using it. A palm is a terrible thing to waste. Also, when drilling, place your work on a firmly seated piece of scrap wood so you can drill all the way through without wobbling. I also put a piece of heavy paper between the work and the wood to prevent scratching the finish.

MORE PIGS IN THE WINGS? After about a month of use, my amp suddenly began squealing like a pig and the heatsink for one channel became too hot to touch. I was experiencing what Ralph Gonzalez called supersonic oscillation (*SB* 2/89, p. 33). He said that oscillation is affected by capacitors.

I'm no theoretician, but the only construction difference between my two channels was in a capacitor, so I guessed it was the source of the problem. I had soldered a 10-year-old 1.0μF capacitor on one channel, instead of using a new one. When I replaced the old 1.0μF, the squealing stopped.

The squeal was audible on both channels, although only one channel was producing it. Apparently it was of suitable frequency to be induced on a nearby circuit board. Perhaps this is a good argument in favor of separate power supplies and total isolation, or at least shielding, of the two stereo channels.

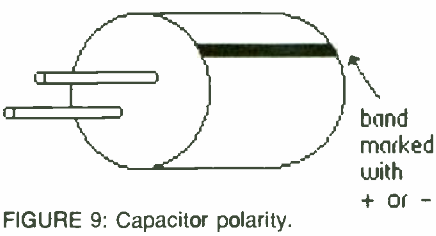


FIGURE 9: Capacitor polarity.

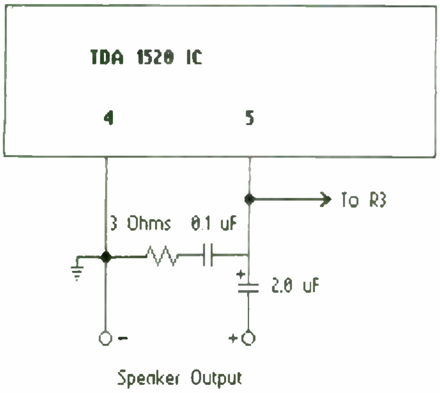


FIGURE 10: Alternative output circuit. Details have been omitted so that the differences between this circuit and *Fig. 1* are clear.

One last note about the output circuit: I found a completely contradictory schematic on page 47 of the Radio Shack *Semiconductor Reference Guide* (1989 ed.), a fragment of which is shown in *Fig. 10*. In contrast to the schematic that comes with the TDA1520, this one calls for a 0.1μF capacitor and 3Ω of resistance to be placed in series across the output from the IC (pins 4 and 5) and a 2.0μF capacitor (rather than 2,200μF) to be placed in series with the output. Perhaps omitting this circuit contributed to my oscillation problem. My amplifier works without these components, however, so you may not want to purchase them.

GIGO. The amp worked fine from the beginning, except for the problem with the output speaker loading I mentioned earlier. Even with only the 9V supply, one housemate informed me that the amp was plenty loud when it woke her up on the floor below me, on the other side of the building.

In regard to volume, don't use the input from a component without a level control. You'll blast your ears—and possibly your speakers—if you do. Experience taught me that the full output is quite loud.

I tried to add input level volume controls by placing a potentiometer in series with the input. This never reduced the volume to zero, even with a 1mΩ potentiometer, so I gave up and used an external disco-type line level mixer for volume adjustment. To use this amplifier with a magnetic cartridge (turntable), you will need to use this type of mixer or a preamp for proper equalization and the correct input level.

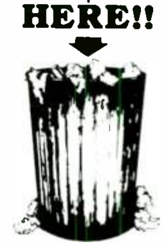
I also realized what the infrasonic (rumble) filter had been doing for my NAD amp. On my homebuilt amps I had to be careful about feedback between my loudspeakers and turntable at moderately high volume levels. Perhaps I'll build a rumble filter for this unit when I learn more about crossovers and high-pass filters.

OPERATIONAL AMPLIFIER. I finally stopped tinkering after a few days and began to listen seriously to music. Since my compact disc player (a JVC XL-V550) has a variable output level, I ran it directly into the amp. Noise was not a problem. The data sheet for the IC says its signal-to-noise ratio is 76dB to 80dB, its total harmonic distortion is 0.01%, and its slew rate is 9V/μS. I'm sure that some *SB* readers could hear the difference between an amp built with the off-the-shelf parts I used and precision components, but I'm not one of them.

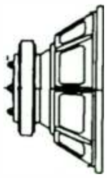
Oddly, when I first used the amplifier, I noticed a dramatically enhanced treble. I originally attributed it to the change from my Klipschs to a newly constructed pair of speakers using Radio Shack ribbon tweeters (which have since been discontinued). But I have come to realize this was due to a subtle degradation of the electronics (capacitors) in my NAD amp. My homebuilt amplifier's frequency response was more accurate than the NAD before it was repaired.

My NAD 3140 is back on my stereo shelf with my Klipsch speakers. I delegated my homebuilt amplifier to service my MIDI and computer workstation, where it drives modified Unilines (*SB* 4/88, p. 28). I can play my synthesizer in stereo and hear the low bass of the lasers' whine as the Thargoids shoot up my forward shields in deep-space warfare.

A LETTER Asking For Help Without A Self Addressed Stamped Envelope Goes HERE!!



VIFA KNOWS TWEETERS



Madisound Speaker Components
8608 University Green
Box 4283 Madison WI 53711
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Fax: 608-831-3771

Model	Size Description	Cost Per Unit
D19TD-05	¾" Dome	$11.00
D20TD-05	¾" Dome	$12.50
D19AD-05	¾" Dome	$16.00
D25AG-05	1" Dome	$18.00
D25AG-35	1" Dome	$22.00
D25TG-05	1" Dome	$13.00
D25TG-55	1" Dome	$13.50
D25TG-85	1" Dome	$16.50
D26TG-05	1" Dome	$13.50
D26TG-35	1" Dome	$18.00
H25TG-05	1" Horn	$15.00
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H25TG-35	1" Horn	$17.50
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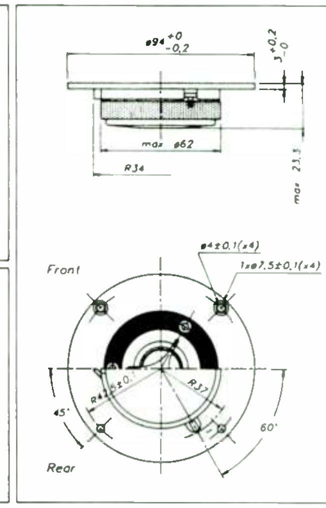
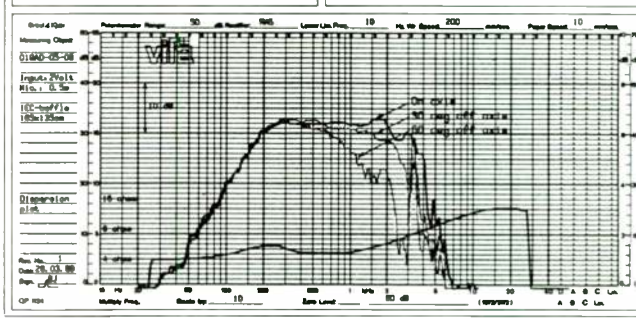
¾" DOME TWEETER D19AD-05-06

SPECIAL FEATURES

- ALUMINIUM ALLOY DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- HIGH LOSS SUSPENSION
- PISTON MOVEMENT TO ABOVE 20KHZ
- HIGH PRECISION PHASE SHIELD
- PREPARED FOR CUSTOMIZED FRONT

NOMINAL IMPEDANCE		6 Ω
NOMINAL POWER (IEC 268-5)		100 W
FREQUENCY RANGE		3-20 kHz
SENSITIVITY (1W, 1m)		87,5 dB
EFFECTIVE DIAPHRAGM AREA		4,4 cm²
VOICE COIL RESISTANCE		4,2 Ω
OPERATING POWER		7,1 W
VOICE COIL DIAMETER		19 mm
VOICE COIL HEIGHT		1,3 mm
AIR GAP HEIGHT		2 mm
FREE AIR RESONANCE		1700 Hz
MOVING MASS (Incl. air)		0,2 g
FORCE FACTOR, B x l		2,2 Txm
MAGNET WEIGHT	(3,7 oz)	105 g



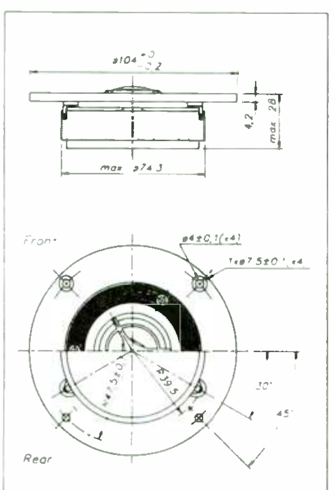


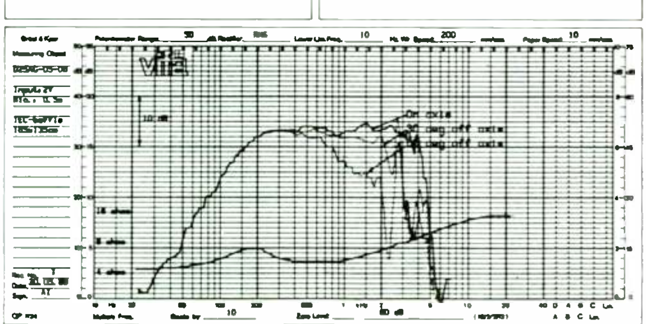
1" DOME TWEETER D25AG-05-06

SPECIAL FEATURES

- ALUMINIUM ALLOY DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- HIGH LOSS SUSPENSION
- PISTON MOVEMENT TO ABOVE 20KHZ
- HIGH PRECISION PHASE SHIELD
- PREPARED FOR CUSTOMIZED FRONT

NOMINAL IMPEDANCE		6 Ω
NOMINAL POWER (IEC 268-5)		100 W
FREQUENCY RANGE		2,5-35 kHz
SENSITIVITY (1W, 1m)		90 dB
EFFECTIVE DIAPHRAGM AREA		7,1 cm²
VOICE COIL RESISTANCE		4,6 Ω
OPERATING POWER		4 W
VOICE COIL DIAMETER		25 mm
VOICE COIL HEIGHT		1,6 mm
AIR GAP HEIGHT		2 mm
FREE AIR RESONANCE		1500 Hz
MOVING MASS (incl. air)		0,3 g
FORCE FACTOR, B x l		3,5 Txm
MAGNET WEIGHT	(8,5 oz)	240 g



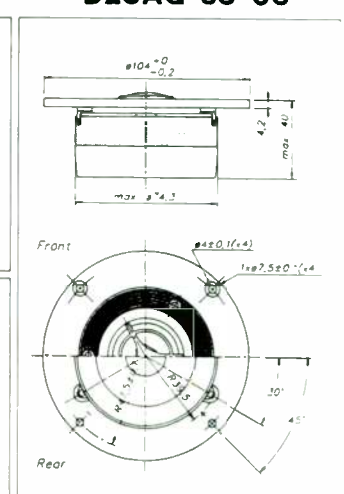


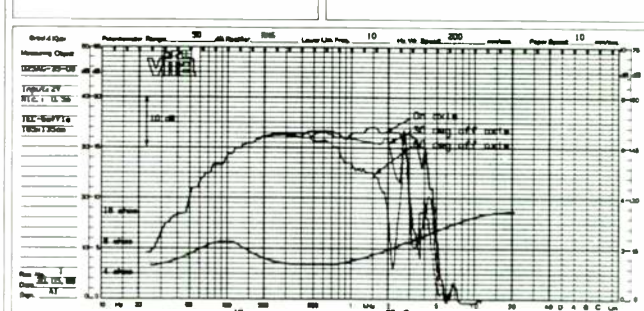
1" DOME TWEETER D25AG-35-06

SPECIAL FEATURES

- ALUMINIUM ALLOY DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- DOUBLE CHAMBER
- FLEXIBLE VOICE COIL BRAIDS
- PISTON MOVEMEMT TO ABOVE 20KHZ
- HIGH PRECISION PHASE SHIELD

NOMINAL IMPEDANCE		6 Ω
NOMINAL POWER (IEC 268-5)		100 W
FREQUENCY RANGE		1,5-35 kHz
SENSITIVITY (1W, 1m)		89 dB
EFFECTIVE DIAPHRAGM AREA		7,1 cm²
VOICE COIL RESISTANCE		4,6 Ω
OPERATING POWER		5 W
VOICE COIL DIAMETER		25 mm
VOICE COIL HEIGHT		1,6 mm
AIR GAP HEIGHT		2 mm
FREE AIR RESONANCE		850 Hz
MOVING MASS (incl. air)		0,3 g
FORCE FACTOR, B x l		3,3 Txm
MAGNET WEIGHT	(8,5 oz)	240 g



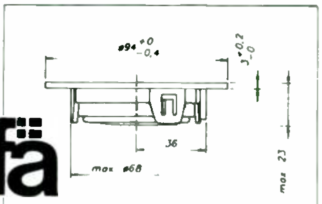


MADISOUND KNOWS VIFA

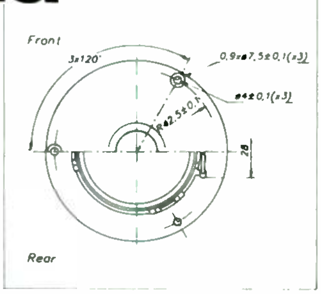
¾" DOME TWEETER — D19TD-05-08

SPECIAL FEATURES:

- HIGH LOSS DIAPHRAGM
- MAGNETIC FLUID
- HIGH POWER VERSION
- GOOD DISPERSION



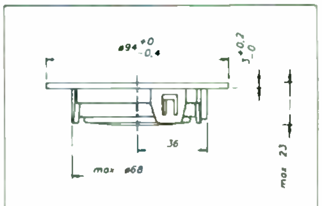
NOMINAL IMPEDANCE	8	Ω
NOMINAL POWER (IEC 268-5)	80	W
FREQUENCY RANGE	3-20	kHz
SENSITIVITY (1W, 1m)	89	dB
EFFECTIVE DIAPHRAGM AREA	4,2	cm²
VOICE COIL RESISTANCE	6,2	Ω
OPERATING POWER	5,0	W
VOICE COIL DIAMETER	19	mm
VOICE COIL HEIGHT	1,4	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	1700	Hz
MOVING MASS (incl air)	0,2	g
FORCE FACTOR, B x l	2,6	Txm
MAGNET WEIGHT (3,7 oz)	105	g



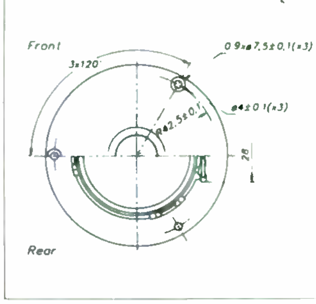
¾" DOME TWEETER — D19TD-00-08

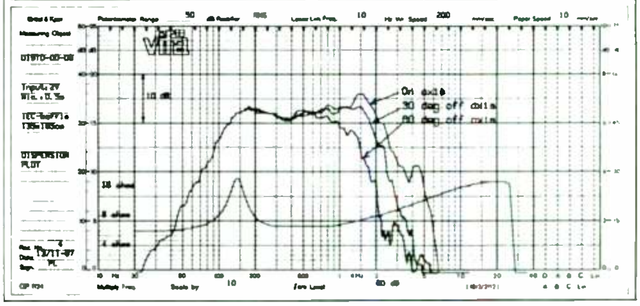
SPECIAL FEATURES:

- HIGH LOSS DIAPHRAGM
- GOOD DISPERSION



NOMINAL IMPEDANCE	8	Ω
NOMINAL POWER (IEC 268-5)	50	W
FREQUENCY RANGE	4-20	kHz
SENSITIVITY (1W, 1m)	90	dB
EFFECTIVE DIAPHRAGM AREA	4,2	cm²
VOICE COIL RESISTANCE	6,2	Ω
OPERATING POWER	4,0	W
VOICE COIL DIAMETER	19	mm
VOICE COIL HEIGHT	1,4	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	1400	Hz
MOVING MASS (incl air)	0,2	g
FORCE FACTOR, B x l	2,6	Txm
MAGNET WEIGHT (3,7 oz)	105	g

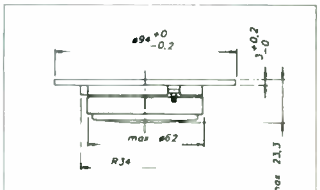




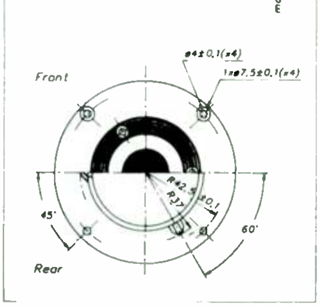
¾" DOME TWEETER — D20TD-05-06

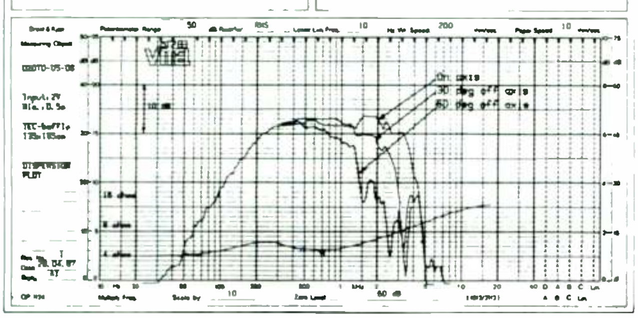
SPECIAL FEATURES:

- FABRIC DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- PREPARED FOR CUSTOMIZED FRONT
- UNCOLOURED SOUND



NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268-5)	100	W
FREQUENCY RANGE	3-40	kHz
SENSITIVITY (1W, 1m)	88	dB
EFFECTIVE DIAPHRAGM AREA	4,4	cm²
VOICE COIL RESISTANCE	4,2	Ω
OPERATING POWER	6,3	W
VOICE COIL DIAMETER	20	mm
VOICE COIL HEIGHT	1,3	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	1700	Hz
MOVING MASS (incl air)	0,2	g
FORCE FACTOR, B x l	2,2	Txm
MAGNET WEIGHT (3,7 oz)	105	g

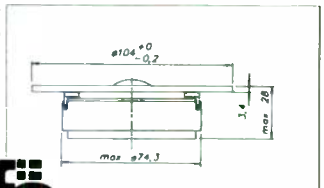




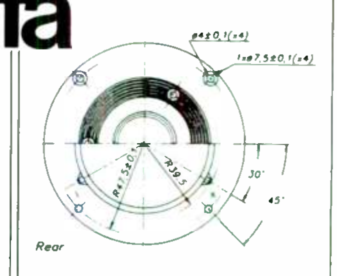
1" DOME TWEETER — D25TG-05-06

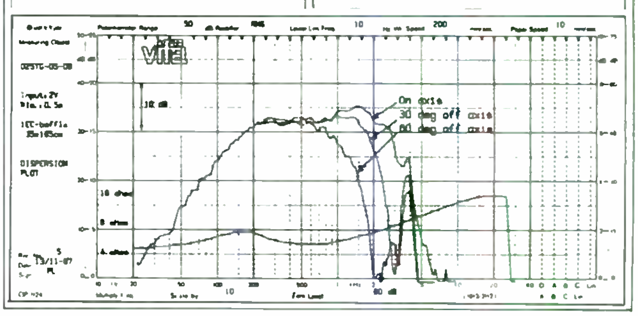
SPECIAL FEATURES:

- HIGH LOSS DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- PREPARED FOR CUSTOMIZED FRONT



NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268-5)	100	W
FREQUENCY RANGE	2,5-24	kHz
SENSITIVITY (1W, 1m)	90	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	4,0	W
VOICE COIL DIAMETER	25	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	1500	Hz
MOVING MASS (incl air)	0,3	g
FORCE FACTOR, B x l	3,5	Txm
MAGNET WEIGHT (8,5 oz)	240	g





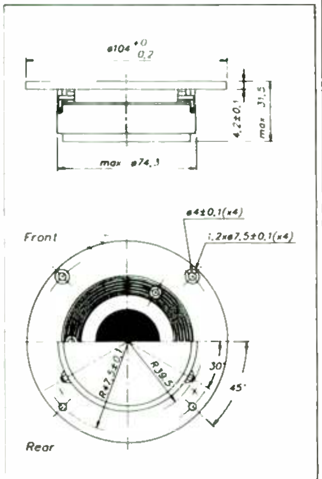
VIFA KNOWS TWEETERS

1" DOME TWEETER D26TG-05-06

SPECIAL FEATURES:

- FABRIC DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- PREPARED FOR CUSTOMIZED FRONT
- UNCOLOURED SOUND

NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268 5)	100	W
FREQUENCY RANGE	3 30	kHz
SENSITIVITY (1W, 1m)	91	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	3,2	W
VOICE COIL DIAMETER	26	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	1500	Hz
MOVING MASS (incl air)	0,27	g
FORCE FACTOR, B × l	3,3	Txm
MAGNET WEIGHT (8,5 oz)	240	g

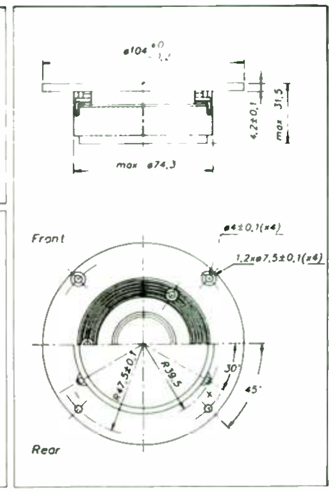


1" DOME TWEETER D25TG-55-06

SPECIAL FEATURES:

- HIGH LOSS DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- PREPARED FOR CUSTOMIZED FRONT

NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268–5)	100	W
FREQUENCY RANGE	2,5–24	kHz
SENSITIVITY (1W, 1m)	91	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	3,2	W
VOICE COIL DIAMETER	25	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	1500	Hz
MOVING MASS (incl air)	0,3	g
FORCE FACTOR, B × l	3,5	Txm
MAGNET WEIGHT (8,5 oz)	240	g

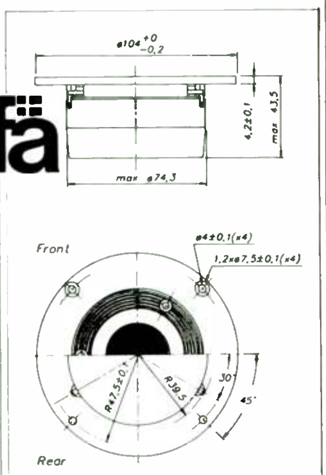


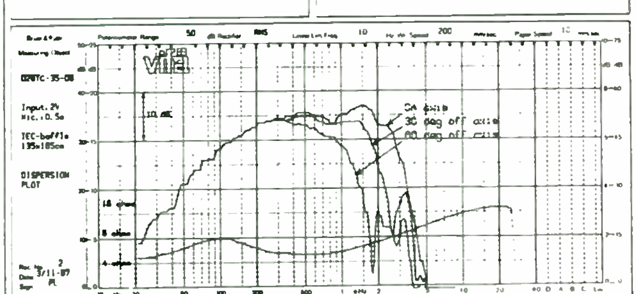
1" DOME TWEETER D26TG-35-06

SPECIAL FEATURES:

- FABRIC DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- DOUBLE CHAMBER
- FLEXIBLE VOICE COIL BRAIDS
- PREPARED FOR CUSTOMIZED FRONT

NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268–5)	100	W
FREQUENCY RANGE	1,5–30	kHz
SENSITIVITY (1W, 1m)	90	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	4	W
VOICE COIL DIAMETER	.6	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	940	Hz
MOVING MASS (incl air)	0,27	g
FORCE FACTOR, B × l	3,3	Txm
MAGNET WEIGHT (8,5 oz)	240	g



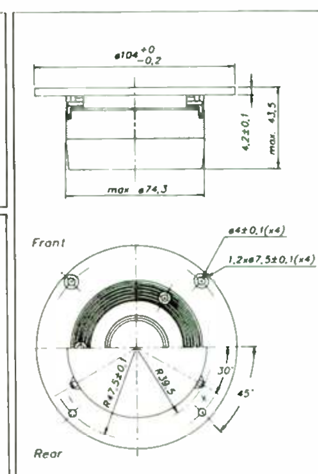


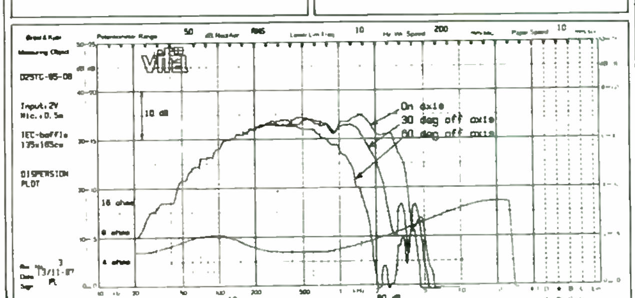
1" DOME TWEETER D25TG-85-06

SPECIAL FEATURES:

- HIGH LOSS DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- FLEXIBLE VOICE COIL BRAIDS
- DOUBLE CHAMBER
- PREPARED FOR CUSTOMIZED FRONT

NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268–5)	100	W
FREQUENCY RANGE	1,5 24	kHz
SENSITIVITY (1W, 1m)	90	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	4	W
VOICE COIL DIAMETER	25	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	750	Hz
MOVING MASS (incl air)	0,3	g
FORCE FACTOR, B × l	3,5	Txm
MAGNET WEIGHT (8,5 oz)	240	g





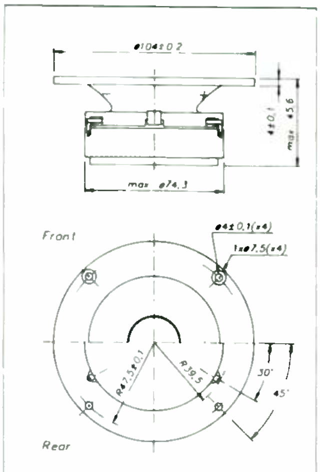
MADISOUND KNOWS VIFA

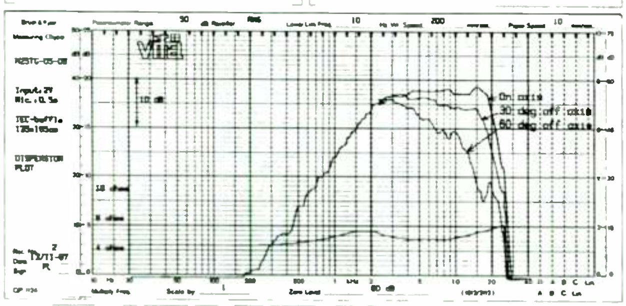
1" IMP. TRANSFORMER H25TG-05-06

SPECIAL FEATURES:

- HIGH LOSS DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- IMPROVED AIR LOAD ON DIAPHRAGM
- MAGNETIC FLUID

NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268 5)	100	W
FREQUENCY RANGE	2,5 20	kHz
SENSITIVITY (1W, 1m)	95	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	1,3	W
VOICE COIL DIAMETER	25	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	1500	Hz
MOVING MASS (incl. air)	0,33	g
FORCE FACTOR B x l	3,3	Txm
MAGNET WEIGHT (8,5 oz)	240	g



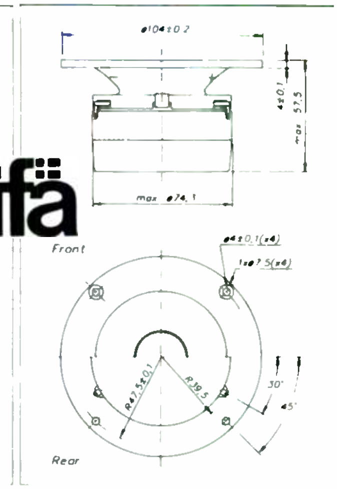


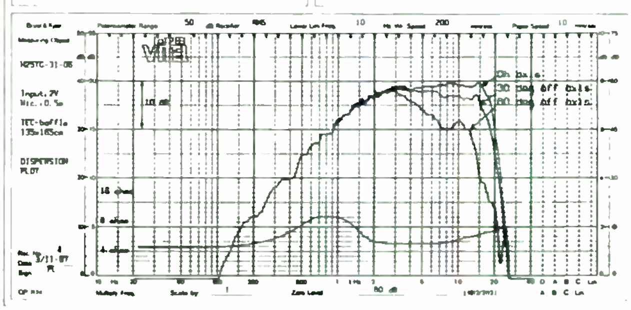
1" IMP. TRANSFORMER H25TG-31-06

SPECIAL FEATURES:

- HIGH LOSS DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- IMPROVED AIR LOAD ON DIAPHRAGM
- FLEXIBLE VOICE COIL BRAIDS
- DOUBLE CHAMBER

NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268 5)	70	W
FREQUENCY RANGE	2 20	kHz
SENSITIVITY (1W, 1m)	96	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	1	W
VOICE COIL DIAMETER	25	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	850	Hz
MOVING MASS (incl. air)	0,33	g
FORCE FACTOR, B x l	3,3	Txm
MAGNET WEIGHT (8,5 oz)	240	g



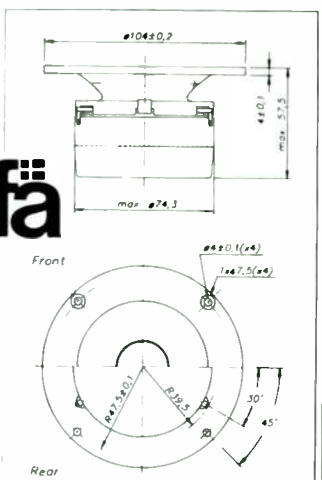


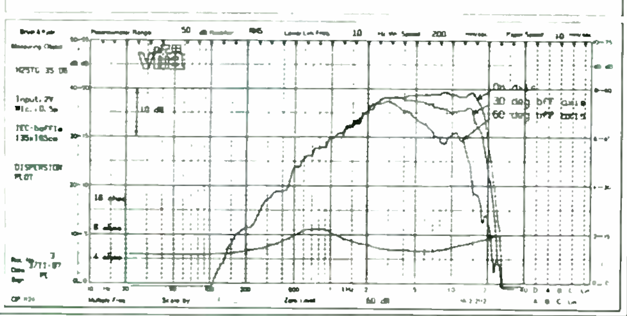
1" IMP. TRANSFORMER H25TG-35-06

SPECIAL FEATURES:

- HIGH LOSS DIAPHRAGM
- IMPROVED AIR LOAD ON DIAPHRAGM
- "BUTTERFLY" VC ASSEMBLY
- MAGNETIC FLUID
- FLEXIBLE VOICE COIL BRAIDS
- DOUBLE CHAMBER

NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268 5)	100	W
FREQUENCY RANGE	2–20	kHz
SENSITIVITY (1W, 1m)	95	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	1,3	W
VOICE COIL DIAMETER	25	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	750	Hz
MOVING MASS (incl. air)	0,33	g
FORCE FACTOR B x l	3,3	Txm
MAGNET WEIGHT (8,5 oz)	240	g



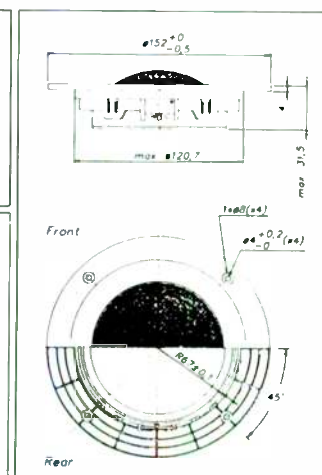


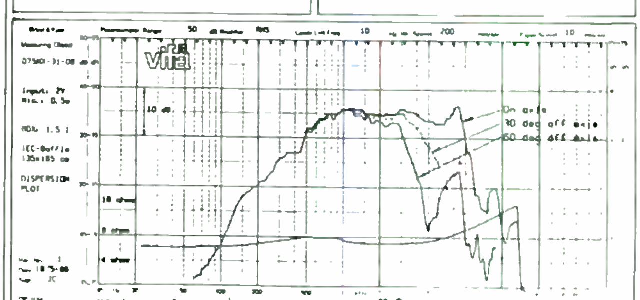
3" MIDRANGE DOME D75MX-31-08

SPECIAL FEATURES:

- PRECOATED FABRIC DIAPHRAGM
- LOW MOVING MASS
- INTERNAL FERRITE MAGNET
- VERY LINEAR IMPEDANCE
- INTERNAL DIAPHRAGM PROTECTION GRIL

NOMINAL IMPEDANCE	[illegible]	[illegible]
NOMINAL POWER (IEC 268 5)	[illegible]	W
FREQUENCY RANGE	0,5 [illegible]	kHz
SENSITIVITY (1W, 1m)	[illegible]	dB
EFFECTIVE DIAPHRAGM AREA	[illegible]	cm²
VOICE COIL RESISTANCE	[illegible]	Ω
OPERATING POWER	[illegible]	W
VOICE COIL DIAMETER	[illegible]	mm
VOICE COIL HEIGHT	[illegible]	mm
AIR GAP HEIGHT	[illegible]	mm
FREE AIR RESONANCE	[illegible]	Hz
MOVING MASS (incl. air)	[illegible]	g
FORCE FACTOR B x l	[illegible]	Txm
MAGNET WEIGHT (8,5 oz)	[illegible]	g





Fast Reply #EF593

Software Report

BOXMODEL and PXO
Nothing Is Perfect

By Richard Pierce

BOXMODEL and *PXO*, by **Robert M. Bullock, III.** Available from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371, each $50.

BOXMODEL, by Robert M. Bullock III, is a program for IBM PCs or compatibles incorporating a unified loudspeaker model. This model encompasses the three major types of direct radiator loudspeakers, sealed box, vented box, and passive radiator, although the latter two could be viewed as manifestations of vented boxes. High-pass equalization is also included in the model. BOXMODEL is an outgrowth of Bullock's and Bob White's earlier offering, BOXRESPONSE. It is a vast improvement, but does have some quirks.

Loudspeaker software falls into two categories: design and simulation. BOXMODEL clearly fits the latter category. If you know a lot about how your system is designed (driver, enclosure, and equalizer parameters), BOXMODEL will tell you how the system performs. A design program might take a proposed system specification (response shape, cutoff frequency, enclosure constraints) then recommend driver parameters and enclosure tuning that best fit the requirements. BOXMODEL makes no recommendations about how best to use a driver or enclosure. That's up to you. It just tells you how well (or how badly) you've done in putting the system together.

BOXMODEL is supplied on a 5¼" 360K IBM format floppy disk. I tested it on a 12MHz 286 AT compatible system, which has an 80287 math coprocessor. Installation to hard disk consists of copying the files from the floppy to the hard disk. No configuration is necessary.

Getting Started

The program begins with an introductory screen, which seems obligatory on all IBM software. I find it a bit annoying to be reminded who did what every time I run something. Everybody does it, so it's unfair to single out this program. Next, you see a screen divided into two windows. The top window, designated "Data Entry," has menus for driver parameters (Thiele/Small model), enclosure type, losses, box, the equalizer, vent or passive radiator, and so forth. The bottom window, labeled "Miscellaneous Parameters," lists items such as driver electro-mechanical parameters, alignment data, and other parameters. More on this later.

In BOXMODEL you enter parameters in each of the relevant sections, such as driver Thiele/Small parameters, enclosure type, box losses, and so forth. The order in which you enter data isn't important. More significantly, you are not forced into a set order of prompting and answering, so you can change one parameter without being forced into re-entering data. This is a vast improvement over programs such as BOXRESPONSE.

When BOXRESPONSE became available, I was using a slightly different approach to loudspeaker design and modeling software. In 1985, I implemented several enclosure design programs, using a spreadsheet calculator program written for RT-11 on DEC PDP-II computers, instead of the conventional programming languages such as C, Pascal, or FORTRAN. (I had abandoned BASIC a decade earlier.) I could enter data not in a time ordered sequence prescribed by the software design, but in the order *I* decided.

If a calculation depended on one or more values I had yet to enter, an error message popped up, but that message was useful, such as: "there's not enough data." I could, once I entered the data, go back and change any input parameter, hand-optimizing a design. I later ported the models to other spreadsheet calculators, such as Lotus 1-2-3, Microsoft's Multiplan, and several public-domain spreadsheet programs, on several architectures and operating systems.

BOXMODEL's input method is reminiscent of the spreadsheet paradigm: you're presented with all the data at once, or at least an important subset, and you can change any one parameter at will.

New and Improved

Another very significant improvement over BOXRESPONSE is the concept of a unified model for direct radiator enclosures. While BOXRESPONSE had this to some degree, Bullock has extended the model to include passive radiator systems as well, and in a fashion that makes passive radiator design's common underlying principles clear.

Once the data is entered, BOXMODEL performs its calculations quickly and, as far as I can determine, accurately—at least as accurately as the model and the input data allow. The program ran without any major disasters or crashes. I couldn't flummox it with ridiculous data. I tried to design a system based on a 3" woofer with a 20Hz resonance and an X_{MAX} of 3". It let me, albeit with appropriately weird results.

Once you've entered all parameters, you can plot the system's magnitude or phase response, its power output, impedance magnitude or phase, driver excursion, and, as appropriate, the vent air velocity or passive radiator excursion versus frequency. The plots are nice looking and well presented, especially on my high-contrast EGA system. I would have preferred grid lines or axis tick marks to get a better handle on the response curve.

Hair-Pulling Time

BOXRESPONSE was difficult to use. If one parameter was wrong or you wanted to change one to see how the response might be affected, you had to start from scratch. BOXMODEL has eliminated this problem. It has, however, done so by replacing it with completely different problems.

I have been involved in audio and loudspeakers for more than 20 years, some of that full-time, including directing a driver manufacturer's engineering department. I've consulted for many well-known loudspeaker companies. Over the last 15 years, I have been a professional software engineer. Much of the work I've done has involved writing software that must interact directly with people. I've become sensitized to making the user interface as easy and as natural as possible.

While BOXMODEL is quick and accurate in its calculations, the user interface is clumsy and tedious and contributes to inaccurate entry. Menu items are selected by moving the cursor (using the cursor arrow keys) to a particular menu's title, then hitting the Enter key. This opens the particular menu box and places the cursor "on" the first data field. From here, things get interesting—if not annoying and aggravating.

Moving through the items in a particular menu box is fairly straightforward. Once you've selected an item, however, you must move the cursor into the field by using the right-arrow cursor key. This places you in the left-most position in the field. The natural tendency is simply to type in the number. This is wrong. If, for example, you move into the f_S (resonant frequency field) and simply type 36 and Enter to set the resonance to 36Hz, in all likelihood, you will not set it to 36. If the field already had, say, 22.5 in it, you would have ended up with 3,622.5Hz. To enter 36, you would have to move the cursor to the left "2" in "22.5" and enter, explicitly, "36.0" or "360" (the decimal point is quite annoyingly skipped over, whether or not it is entered).

Woe Is Me

At that point, hit the Enter key to complete the field, which will lock whatever you've typed into that field, and move you on to the next field, or it will exit the menu, if you're not within a field. Another annoyance here, "Enter" does seemingly different things at seemingly similar items in a menu. Its action is not consistent. I found this method of editing and entering data very inconvenient and annoying, and in the two or three hours I spent using BOXMODEL, I never got used to it.

Another irksome problem involves using the cursor keys to get around the menus. When moving from menu to menu, you use the left-arrow key to move left, and the right-arrow key to move right. That seems logical enough, but say you're on the second row menu for vent diameter, and you want to move to the menu above it (enclosure type). Naturally, you would use the up-arrow key to get there. Unnaturally, it works exactly the same way as the left-arrow key, moving you to the high-pass equalizer menu. You need three more cursor moves to get to where you want to go.

As a software engineer, I have a good guess why the menus and cursor keys work the way they do. I suspect the range of menus are implemented as either a linked list or a one-dimensional array of some menu structure. Moving the cursor keys moves you to the next or previous member of the list or array. That's great, except solving the problem is trivial, once you've designed your menu structure. If your menu structures are in an array, make it two-dimensional. If they are linked lists, make two sets of pointers, one set for the menus to the left and right, another for the menus above and below.

Moving Pains

The time when all this gets *really* annoying is when you're trying to do some work. As a test (and as entertainment), I decided to see what I could do with a 6½" woofer in a vented box with a third-order equalizer. I ended up with a 20-liter enclosure, tuned to 40Hz, which led to a QB3-like response, drooping low-end and all. The equalizer was set up with one ncorner frequency at about 30Hz, the other at about 15, with the damping set to about .5, giving about a 6dB boost at 30Hz. The resulting system was about ±2dB above cutoff.

I decided to tune the system further by playing around with the damping. Here's what was involved: from the plot screen, any key to exit, then five left arrows to get to the HP EQ menu, then enter to open it up. This placed me on the EQ order field. I didn't want that, so I hit down arrow to get to the other entries. Well, that didn't work, because the HP EQ menu is just a little different than the others. So I hit "Enter." Ah! There I am, on "Cr1," the first corner frequency. Two more down arrows got me to the "Dmp."

I wanted to change the damping factor from 0.425 to 0.427. The character I wanted to change was five characters over, so I hit the right-arrow key five times, hit seven to replace the five and noted I set the damping factor not to 0.4270, but to 0.4257. Why? Because the decimal point is automatically skipped, so even though it looks like five moves are necessary, it's really only four. I just hit 0 to replace the 7, then I hit Backspace to delete the 5, then hit 7 to replace it, and found I replaced 0.4257 with 0.4257.

Pain and Choice

Why? Because backspace (the standard delete-character-to-the-left key) is ignored. Once again I replaced the 7 with a zero,

Continued on page 47

OLD COLONY'S USER-FRIENDLY SOFTWARE

Speaker Designer™
Release 1.2 by Stuart E. Bonney

A loudspeaker system design aid and modeling tool for use with both closed and vented systems over the frequency range from 10 to 300Hz. Computes and displays system frequency response, power handling capabilities, and relative sound pressure level (SPL) outputs for each of 26 discrete frequencies over this range. Includes one year support by the author when the user registers this Shareware product. Each **$19.50**

IBM 5¼" 360K DS/DD **SPD-1B5**

Loudspeaker Modeling Program
by Ralph Gonzalez (*SB* 1, 2, 3/87)

LMP produces a full-range frequency response prediction for multi-way loudspeakers, including the effect of the crossover, driver rolloffs, interdriver time delay, "diffraction loss," etc. (Includes author support.) Each **$17.50**

Specify:

Apple II 5¼" SS/DD (DOS 3.3)	**LMP-1A5**
Apple Macintosh 3½" SS/DD	**LMP-2M3G**
IBM PC/XT/AT 5¼" DS/DD	**LMP-2B5G**
Commodore 64 5¼" DS/DD	**LMP-1C5**

"Souped-Up" LMP provides professional-quality graphics and a fast, friendly, user interface. Remains compatible with LMP data files and is available for IBM PC (CGA, EGA, VGA, or Hercules graphics) and Macintosh computers. The Macintosh version also provides square wave analysis with audible output.
Each **$49.50** (upgrade price: $39.50*)

Specify:

IBM PC/XT/AT 5¼" DS/DD	**LMP-3B5G**
Apple Macintosh 3½" SS/DD	**LMP-3M3G**

*Original LMP disk or sales receipt must be included w/ order.

Driver Evaluation and Crossover Design
by G. R. Koonce (*SB* 5/88)

Disk 1 evaluates the suitability of drivers for closed, vented and passive radiator enclosures, and allows detailed designs of vented boxes.

Disk 2, in addition to driver evaluations, allows the design of first-, second-, and third-order crossovers. 5¼" IBM: 360K, DS/DD. Each **$12.50**

Specify:

Driver Evaluations	**DRI-1B5**
Crossover Design	**CRS-1B5**

Active Filter Design
by Fernando Garcia Viesca (*SB* 4/88)

Calculates component values for Butterworth filters in four configurations: High- and low-pass in second- and third-order. (Includes author support.) Each **$17.50**

IBM 5¼" 360K DS/DD **ACT-1B5**

Two-Way Active Crossover Design
by Gary Galo (*SB* 5/88)

Performs the calculations for the eight two-way active crossover designs described by Bob Bullock using formulas exactly as given in the articles; plus a program to calculate V_{TH}. (Includes one year user support.) Each **$20**

Specify:

IBM 5¼" 360K DS/DD	**TWO-1B5**
IBM 3½" 720K DS/DD	**TWO-1B3**

Stepped Volume Controls
by Joseph O'Connell (*TAA* 4/88)

These ready-to-run Mac programs come on a 3½-inch SS/DD disk initialized as a 400K disk for compatibility with all machines. Also included are the Pascal source codes, should you wish to customize them for your own use. **Program A.** Precisely matches the resistor values to the measured or estimated source and load impedances, yielding great accuracy. Your volume control can have 3 to 99 positions. The program will ask you how many dB each step should be attenuated and has provisions for a standard audio taper or any other taper you devise. **Program B.** Calculates the taper that will result with your actual resistor values, because you are limited to standard values or with series and parallel combinations. It can also show the effects of different source and load impedances on the taper. Both programs (contained on the same disk) allow you to save their output to a text file and include author support via mail. Each **$25**

Apple Macintosh 3½" SS/DD **SVC-1M3**

BULLOCK & WHITE

BOXRESPONSE

Model-based performance data for either closed-box or vented-box loudspeakers with or without a first- or second-order electrical high pass filter as an active equalizer [*SB* 1/84]. *The program disk also contains seven additional programs as follows:*

Air Core: This program was written as a quick way of evaluating the resistance effects of different gauge wire on a given value inductor. The basis for the program is an article in *Speaker Builder* (1/83, pp. 13-14) by Max Knittel. The program asks for the inductor value in millihenries (mH) and the gauge wire to be used. (NOTE: only gauges 16-38.)

Series Notch: Developed to study the effects of notch filters in the schematics of some manufacturers. Enter the components of the network in whole numbers (i.e., 10 for 10μF and 1.5 for 1.5mH) and indicate whether you want one or two octaves on either side of resonance. Output is frequency, phase angle and dB loss.

Stabilizer 1: Calculates the resistor-capacitor values needed to compensate for a known voice coil inductance and driver DC resistance.

Optimum Box: A quick program based on Thiele/Small to predict the proper vented box size, tuning and −3dB down point. It is based only on small signal parameters, therefore, it is only an estimate of the response at low power (i.e., limited excursion).

Response Function: Calculates the small signal response curve of a given box/driver combination after inputting the free-air resonance of the driver (f_S), the overall "Q" of the driver (Q_{TS}), the equivalent volume of air equal to the suspension (V_{AS}), the box tuning frequency (f_B), and the box volume (V_B). Output is the frequency and relative output at that frequency.

L-Pad Program by Glenn Phillips: Appeared in *Speaker Builder* (2/83, pp. 20-22). It is useful for padding down a tweeter or midrange while still retaining the same load as the driver itself.

Vent Computation by Glenn Phillips: Calculates the needed vent length for 1, 2 or 4 ports of the same diameter. Input box volume in cubic feet and required tuning frequency (f_B), output is vent length and vent area for each case.

Medium: 5¼" SS/DD Disk. Price, **$25.**

Specify:
BOXRESPONSE

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BOXRESPONSE BOX-2B5G **$50**

PASSIVE CROSSOVER

This disk is a result of Mr. Bullock's extensive research concerning first-, second-, third-, and fourth-order passive crossovers in *Speaker Builder* 1, 2 & 3/85; **$25**

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CALSOD

Computer-Aided Loudspeaker System Optimization and Design
by Witold Waldman

CALSOD is a new entry into the field of crossover network optimizing software available for the IBM PC desktop computer. It combines the transfer function of an LC network with the acoustic transfer function of the loudspeaker, by using some form of iterative analysis. CALSOD creates, through the process of trial-and-error curve fitting, a suitable transfer function model which it can then optimize. The program is the subject of CALSOD author Witold Waldman's research paper "Simulation and Optimization of Multiway Loudspeaker Systems Using a Personal Computer" which appeared in the Audio Engineering Society *Journal* for September 1988, pp. 651-663. CALSOD differs considerably from other software since it models the entire loudspeaker output of a multiway system, including the low-end response, and the summed responses of each system driver.

The program performs a lot of tricks. One of the more spectacular of these allows the designer to specify the location of the driver acoustic centers using an XYZ coordinate system. Thus, if the designer expects to mount a driver combination on a flat baffle, the summed response can be optimized to compensate for rearward displacement of a woofer's acoustic center with respect to a tweeter. CALSOD can model up to seven drivers at a time in a four-way system giving the summed response and acoustic phase response of the entire system.

The CALSOD program comes on a single 360K floppy, and requires one directory and two subdirectories in installation, plus access to the DOS GRAFTABL file, which it uses for a couple of special symbols. The 133-page User Manual, provided on a second disk, is well written, adequately describes the various program functions, and contains an excellent tutorial example, which demonstrates the use of the program.

Specify:

IBM 5¼" 360K DEMO	**CAL-2B6D**	**$ 5.00***
IBM 2 x 5¼" 360K DS/DD	**CAL-2B6**	**$65.00**
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*** Usable as credit toward later purchase of full CALSOD.**

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Continued from page 45

then used the left-arrow key to get to the 5, replaced it with a 7, then hit "Enter" to close the field, "Enter" to exit the menu, then five right-arrow keys to get back to the plot menu, "Enter" to open the plot menu, "Enter" to go ahead and plot (presuming the right plot mode is selected). Then I had a plot telling me I really should have used 0.423, not 0.427. Repeat until you are done.

Many other minor annoyances make BOXMODEL almost painful to use. Help is provided for each menu, yet Help is disabled once you've entered a menu. There is no help provided for individual entries, only for menus. You can keep up to four configurations (named "A," "B," "C," and "D") and recall and compare them. Yet, no screen indicates the one on which you are currently working.

Seemingly related parameters, such as the driver's Q_{MS}, Q_{ES}, and Q_{TS} are found in two places on the screen. Q_{MS} and Q_{ES} are entered into the Data Entry window, and Q_{TS} is displayed in the Miscellaneous Parameters window in the other box. Sure, Q_{TS} is derived from the other two, but is it "miscellaneous other"? What if you know Q_{TS}, but not Q_{MS} and Q_{ES}? Well, you guesstimate the other two, and fuss with them until Q_{TS} is right. That might be easy for someone like me, having a catalog of typical driver parameters floating around in my head, but it's a different matter for a less experienced user. Presumably, the design cannot proceed without all three, but I would like to have the option of not caring what the other two are.

When in Doubt

Some derivations, such as sensitivity, cannot be made, but I may not care. A similar problem exists with the enclosure losses section. Three entries describing enclosure losses, Q_{AS}, Q_{LS}, and Q_{PS} are presented in the Data Entry section. In the "miscellaneous other" box, we have a derived Q_{BS}. This seems an inappropriate place for this figure. I would like not to bother with the breakdown of losses, but to enter a specific Q_{BS}. The results won't be as accurate, but I am usually at the earlier stages of design, and I'm only looking for an approximation.

Further, the "miscellaneous" section displays the driver's electro-mechanical parameters, which is OK, except they contribute very little to my knowledge about how the system works.

There is a print screen function for the entry screen, but BOXMODEL provides no way of getting graph hardcopy, although third-party utilities such as **capture.com** and others can be used. It would be useful to have some way of exporting the data via a simple ASCII file to other programs, for graphing, analysis, or whatever.

Most annoying is the use of the Escape character. Escape is normally used in BOXMODEL to get out of a menu, leaving its contents unchanged. The inadvertent and errant Escape hit can leave you stranded in DOS, as the Escape character is also the exit from BOXMODEL. I prefer a specific menu item I select, and execute to quit. Several times I set up a model and accidentally hit Escape, losing all my changes.

BOXMODEL's problems are not unique. A whole raft of programs suffer similar ills. A classic example is Peter Schuck's XOPT crossover optimization program. The basic idea is wonderful, but Schuck's software implementation is lousy. BOXMODEL is far above XOPT in several ways: it is not fragile like XOPT, its user interface, in spite of my criticism, is better, and its display and graphs are nicer and more useful. (I am well aware XOPT and BOXMODEL do two very different things. I am comparing them on the basis of software quality, not purpose.)

Somewhere in BOXMODEL there is a tool that is fast, accurate, versatile, and useful, and could be a lot of fun. Unfortunately, it is buried under a nearly impenetrable user interface that takes away most of the fun, and a substantial amount of utility. Fix the user interface, and BOXMODEL could be a winner.

Designer Software

Bullock also recently released PXO for IBM PCs and compatibles, which assists in the design of passive crossover networks. It is a compilation of many Bullock design principles that have been published in *Speaker Builder* and in the AES *Journal* over the last decade. His work has been valuable, and I've incorporated much of it in my own software.

PXO is an example of design software. You supply it with a set of basic parameters, such as crossover frequency, order, and response type, and it provides a component list. It also shows resulting response curves.

PXO is a program I would use as the first step in a passive crossover network design. It provides the values and topologies I would use as the input to some optimization process, the final goal of which is an integrated system of drivers, enclosure, and crossover, which meets a target specification. It is not an optimization program, and it makes some simplifying presumptions. For example, it presumes purely resistive drive impedances, no response anomalies in the drivers, no consideration for placement and interference effects—perfect, lossless, linear components.

It cannot design conjugate networks for compensating the non-resistive behavior of driver impedance ("resistifying") either. This is not a criticism, rather it is loudspeaker design reality. You've got to start somewhere, and the more simplifying assumptions you make up front, the faster you converge on a plausible set of solutions. Once you have these, then start worrying about the higher order effects. PXO gets you to that point.

PXO presents an input system the same as BOXMODEL and, regrettably, suffers from the same problems. It also has two windows more logically laid out. The top window is used for data entry, and the bottom one values for the proposed network.

Icing on the Cake

The program has its good points. As you change the crossover order (first- to fourth-order are supported), the number of bands in the system (you can select two- or three-way), and other issues, it lists the values, and shows the components' actual schematic layout. It uses the IBM extended character set for this, so its effect is somewhat crude, but useful. It lists the values in tables with parts designators (L1, C4, and so forth), and the layout has these parts designators in place of the actual component symbols. It would be icing on the cake if the component values where to appear where they belong on the layout, but this is, at worst, a minor inconvenience.

You can plot amplitude and phase response of the whole system or of each individual output (low-pass, band-pass, or high-pass). You can also display input impedance to the entire system. The graphs are, as in BOXMODEL, of generally good quality, though the amplitude versus frequency curves suffer from some problems. The graphs are on a highly expanded scale, being only a few decibel high. While this is useful for looking at the fine detail surrounding the crossover (the graph is displayed from −6 to +3dB), it is useless for examining the response well outside the band.

The detail around the crossover is useful for determining how drivers will interact, but it's just as important to see what tweeter feed is two or three octaves below its crossover point to determine what power-handling problems it may encounter. Having both expanded (say 10dB full-scale range) and compressed (like 50dB) would greatly enhance PXO's graphing. PXO cannot deal with anything other than straightforward designs. There's no provision for unsymmetrical networks, different cutoff frequencies for high-pass and low-pass outputs, and so forth. Granted, adding these would undoubtedly complicate the algorithms. As a first pass, get-into-the-ballpark tool, PXO is well-suited to the task.

If Bullock were to solve the user interface problems on this, as well as on BOXMODEL, PXO would be a simple-to-use tool for designing passive crossover networks.

Robert Bullock replies:

After reading Mr. Pierce's review, I made changes to BOXMODEL and PXO. The main menus of both programs now have an explicit Quit command. Pressing the ESC key will still terminate either program, however. But now you won't lose data in BOXMODEL because it is written to a file called BOXMODEL.SES when you end your session. The program automatically loads BOXMODEL.SES if it is in the same directory. Otherwise, it is created when you start. This new feature lets you interrupt a BOXMODEL session and return to it later with the same data intact.

Mr. Pierce's complaint about the needless entry of the equalizer order in BOXMODEL before you can change equalizer shape parameters is well taken. These two activities have been duly uncoupled.

Limiting the movement in menus in both programs to one dimension can require an excessive number of keystrokes. Rather than adopting Mr. Pierce's suggestion of two-dimensional movement, I opted to allow direct access to any menu item by pressing a key associated with that item. The capitalized letter in the otherwise lowercase item label determines the relevant key. This method has been incorporated into both BOXMODEL and PXO.

For the most part, the HELP key in BOXMODEL was active only on main menu items. In the revised version, virtually all menu and data items have their own HELP message. The HELP messages now give suggestions as to how to simulate entering Q_{TS} or Q_{BS} in BOXMODEL.

BOXMODEL and PXO now allow the use of grid lines, but this option severely slows the proceeding on some machines. That is why it is optional. Also, from the main menu of PXO graphs, you can choose one of two decibel ranges.

Except for α, h, and μ, others have requested the data in the Miscellaneous Parameters window of BOXMODEL. I have eliminated these three items and tried to improve the look of what remains.

Now I will comment on other complaints Mr. Pierce had. His objection to the user interface appears to be due to the method I have chosen for data entry. In the data entry windows, you scroll through the items and highlight the one you wish to change. To access it, you press the right-arrow key. This brings you to the leftmost position of that field. You can *edit* the current value, but cannot enter a new value.

Mr. Pierce wishes to enter a new value at this point. When you initially configure your system, you *would* enter new values, but thereafter you would simply make changes to those values, not change them completely. For this reason, I prefer my data-editing approach to his new-entry approach.

Given our differing expectations about data entry, it is not surprising he finds my technique inconvenient. He did cause me to look a little closer at my approach, though, and I made a change in both programs that will help him achieve more accurate data entry. Instead of moving into the first position in the field, the cursor moves to the first digit of the data. This should eliminate the error he describes in trying to change f_c.

He suggests it would be useful to export the data from BOXMODEL via an ASCII file to other programs. I am not sure whether he is talking about the system parameter data or the plot data calculated by the program. In the former case, it is doubtful another program would take exactly the same data set, so such a file's value is questionable. In the latter case, graphs from another program probably would not produce any new information.

I see nothing inconsistent about my use of the ENTER key. I use it in three situations: to *enter* a menu choice into the program, to *enter* a completed data item, and to *enter* a completed window of data items into the program.

I obviously did not make clear the meaning of the system designations A, B, C, and D in BOXMODEL. B, C, and D are storage designations only. A is always the system currently displayed. When you recall B, for example, its contents are displayed; in other words, it becomes A. You can plot any combination of stored systems along with the displayed system; the names A, B, C, and D distinguish which ones they are.

Finally, for the price of BOXMODEL, I think it is too much to expect it to have a graphics dump capability. I rely on the graphics dump facility provided in MS-DOS 4.

Product Review

The Precise Monitor 10

By Vance Dickason
Contributing Editor

Monitor 10, Precise Acoustics Laboratories, a division of Onkyo USA Corp., Suite, 200 Williams Dr., Ramsey, NJ 07446, (201) 934-1335, $1,600.

This is this first product review I have done for *Speaker Builder* and also the first *SB* review of a non-kit commercial loudspeaker. Before discussing the product, I would like to define my criteria and the purpose of this report.

Most loudspeaker reviews inform prospective consumers of its salient characteristics, presumably to aid their buying decision. The typical loudspeaker review performs an objective evaluation and then applies a subjective judgment to the loudspeaker's sonic quality.

Generally, most reviewers attempt to correlate the measured data with their observations, although they disagree as to the absolute degree they can accomplish this. The best reviews, such as those in *Stereophile* magazine, also offer a comparison to other manufacturers' products. This can include a hierarchy of acceptable choices in a given price range.

What follows is similar, but my intention is quite different. This review is not aimed directly at informing loudspeaker buyers because most *SB* readers would rather build their own. This critique's primary goal is to reveal the manufacturer's engineering and design procedure.

The result is a review that looks deep into the "guts" of the speaker and tries to ascertain the product tradeoffs and design decisions made from creative inception to packaging and sale. This perspective will provide *SB* readers with technical insight into the creative methodology of commercial applications and will suggest useful techniques applicable in amateur and limited production situations. Conversely, it may also suggest what to avoid, which is just as valuable.

To put manufacturers at ease, I do not intend to supply information that will allow readers to clone successful designs. Network schematics are provided, but not component values. Driver Q data will be given, but not details of motor geometry. Even if a manufacturer uses an off-the-shelf driver, he frequently has altered the motor or suspension geometry, added shorting rings, or performed some other modification not necessarily available to

the amateur (or to other manufacturers for that matter).

The Overview

The Onkyo Precise line of loudspeakers has received considerable press. In fact, the designer, Keith Johnson, discussed some of the design technology employed in the Precise line of speakers in a previous issue of *SB* (6/88, p. 9). The effort underscores an attempt by a large Japanese manufacturer to gain a foothold in the US market—not an easy task. It is no secret Japanese loudspeakers have never enjoyed the nearly unchallenged success of their audio electronics. The advantage Onkyo was trying to gain was to marry stateside engineering with offshore production to create a successful product. It was a fairly simple formula: to get an American-sounding loudspeaker, hire an American design engineer. If it were only that easy.

The Monitor 10 is the flagship. It is a three-way vented design using a 10″ woofer. The 1″ cloth dome tweeter and 6″ paper cone midrange are mounted in a modular satellite enclosure placed above the woofer box in a fashion similar to KEF and B&W designs. Tweeter and mid driver are staggered for time alignment™, giving the small enclosure a kangaroo "look." The mid/hi-frequency box is back from the edge of the woofer enclosure's front baffle, appearing to be placed in alignment, although phase relationships at the 120Hz woofer-to-mid crosspoint are not nearly so critical. The model used for this review was one of the earlier ones and did not include the modification allowing control of bass output.

The Woofer

The woofer in the Monitor 10 is a 10″ paper cone device using a rubber half roll surround, a paper dust cap, and a 20oz. ferrite magnet. I measured the Thiele/Small parameters of the driver using the Audio Precision System 1 analyzer, a computer interfaced test instrument that combines the functions of a sweep generator, level meters, and a distortion analyzer in one high-performance package that is capable of a wide range of audio measurements.

Using the automated Q measurement program,[1] I discovered data for both woofers was practically identical with F_S = 22Hz and Q_{TS} measuring close to 0.26. My first impression was that a Q_{TS} of 0.26 is a low Q for a 10″ woofer with this size magnet structure. Putting all the parameters into LEAP[2,3] offered a quick explanation. The cone assembly weight plus air mass load (M_{MS}) was only 28 grams, at the low end of the range for a 10″ driver. Eight-inch woofers have M_{MS} values of 15–28 grams, and many typical 10″ drivers have M_{MS} values of 35–50 grams. Closer examination of the cone revealed it to be a thin curvilinear type, which I nearly creased when I tried to flex it slightly between my fingers to check for stiffness.

Keith Johnson, according to the *SB* review and company literature, used his

TABLE 1

MONITOR 10 SPECIFICATIONS

Type	three-way, bass reflex
Woofer	10″ (25 cm)
Midrange	6½″ (16 cm), polymer laminated
Tweeter	1″ (2.5 cm)
Frequency Response	20Hz–3.5kHz
Rated Input Power (DIN)	100W
Max. Input Power (EIAJ)	250W
Impedance	4Ω
Recommended Amplifier Output Power	50–250W
Sound Pressure Level	90dB/W/m
Dimensions	15⁹⁄₁₆″ x 44¼″ x 13⅜* (396x1,124x340*mm)
Weight	70.5 lbs. (32 kg)
*including grille	

interesting DMSA (differential mode stress analysis) technique to design the cones in the Precise line of speakers. According to the literature, DMSA "enabled them to eliminate colorations which standard measurement methods often miss." While I do not doubt Mr. Johnson did so, I find no evidence this work showed up in production. I encountered a lightweight, off-the-shelf paper cone no different than any other; it did not appear to be the result of any special or sophisticated development process.

Using the DRA Labs MLSSA analyzer and an ACO Pacific 7012 microphone, I measured the response of the driver (*Fig. 1*). This measurement (and all other measurements except the near-field and groundplane woofer measurements) was made outdoors with the speaker placed on a 5.5′ tower. The microphone is more than 90″ above the ground; therefore, the distance for the first ground reflection back to the mike is about 16′. At 1,130 feet per second, it would take 0.014 seconds (14mS) for the ground reflection to arrive at the microphone. This defines the window for the impulse response and determines the low-end accuracy of the measurement. Low-end accuracy of FFTs is generally given as 1/(time period) of the impulse window, in this case 1/.014 = 70Hz. Data below 70Hz with this time edit is unreliable.

The graph depicts the woofer frequency response with (dotted line) and without (solid line) the crossover. Notice the severe breakup modes between 2kHz–4kHz, typical of underweight cones. This type of response problem, sometimes called "cone cry," is noticeable considering the network used, but more on that later. If

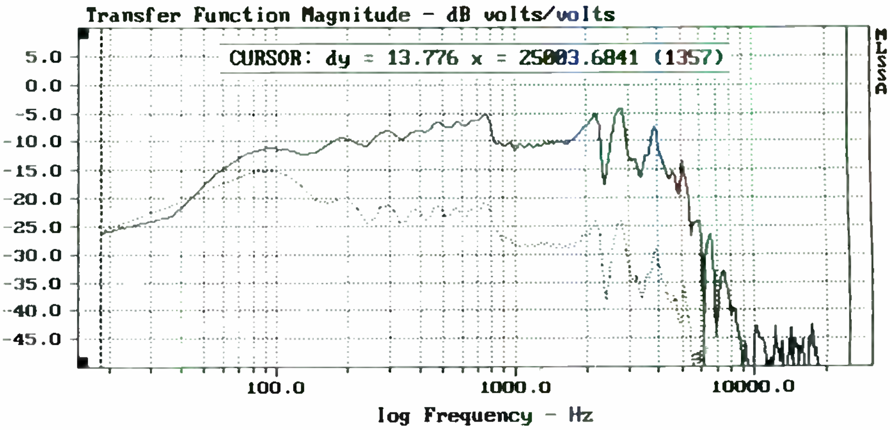


FIGURE 1: Monitor 10 woofer response.

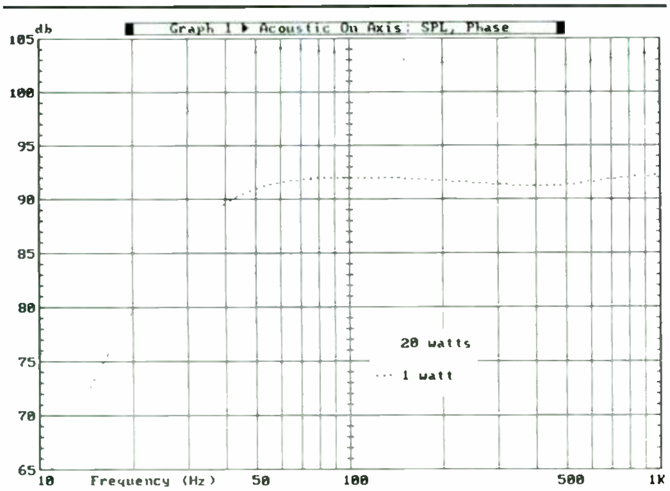


FIGURE 2: Woofer simulation. Monitor 10 vented at 20W and 1W.

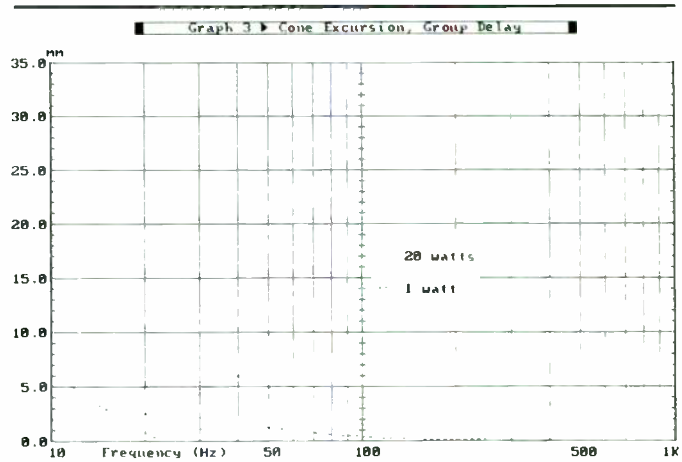


FIGURE 3: Woofer cone excursion simulation. Monitor 10 vented box at 20W and 1W.

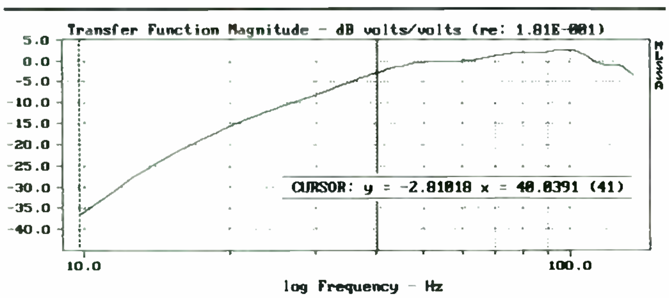


FIGURE 4: Woofer low frequency response (measured ground plane f_3 = 40Hz).

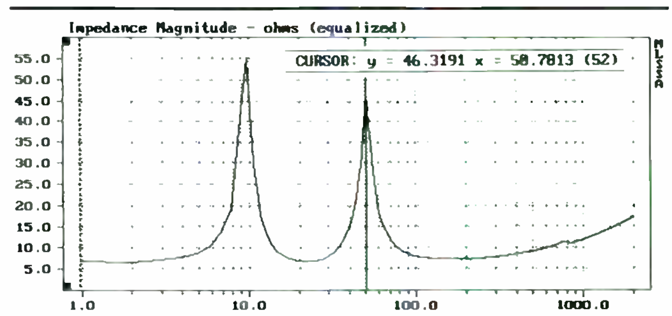


FIGURE 5: Woofer impedance magnitude without crossover.

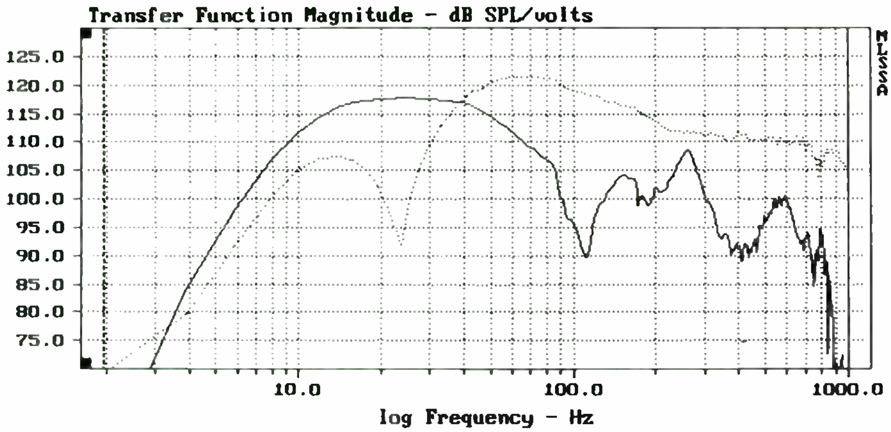


FIGURE 6: Woofer and port response.

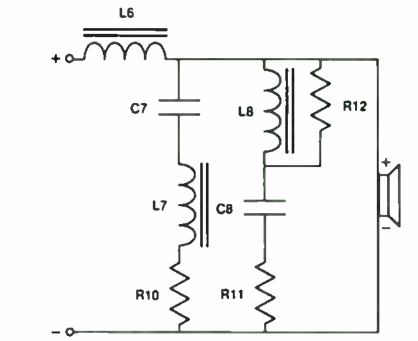


FIGURE 7: Diagram of the woofer network.

DMSA were used to develop this cone, I am at a loss to understand what the performance criteria were. Note the network lowered the overall sensitivity of the woofer by at least 4dB. LEAP calculated the woofer reference efficiency on a baffle at 92dB, while the speakers' overall broadband efficiency measured about 86dB (referenced to 1W/1m).

The LEAP simulations of this woofer, using the cabinet's 2.3 cubic foot net volume with 80% of the volume occupied with damping material, are given in *Fig. 2*. The graph shows two curves, the low-end response at 1W with a nominal 20°C voice coil temperature and at 20W with the voice coil temperature set at 100°C. *Figure 3* gives the cone excursion curves at the same power and temperature levels. At the 20W level, LEAP predicted an f_3 of 40Hz and a required excursion of 6mm with an SPL of 104dB. Since the X_{MAX} of this driver is nearly 6mm, it could be expected to stay fairly linear in operations up to SPLs sufficient for home-listening situations.

The MLSSA outdoor groundplane measurement, depicted in *Fig. 4*, confirmed the f_3 point to be about 40Hz. The measurement impulse was windowed at 70mS, which yields accuracy to about 14Hz. Groundplane measurements are made by positioning the cabinet outdoors on a hard reflective surface, clear of any boundaries for at least 30 feet and with the microphone on the ground directly in front of the speaker. The results of groundplane measurements are close to free-field measurements.[4] LEAP simulation also placed the upper and lower impedance peaks at 10Hz and 50Hz, the values measured using MLSSA's impedance mode (*Fig. 5*, without crossover).

The cabinet was constructed with 0.75" MDF with a 1.25" MDF front baffle and has curved front edges. The literature suggests the baffle is somehow isolated from the rest of the cabinet because "slits where the baffleboard joins the sides of the cabinet dramatically reduce the vibrations transmitted to the enclosure sides."

There is a 1/16" wide groove 1/8" deep around the top and sides of the baffle where it attaches to the rest of the box. Glue blocks attach the baffle to the enclosure, which would certainly be the dominant effect directly coupling baffle to box in terms of vibration. Although I did not use accelerometers and measure the effect of a similar enclosure with and without the cosmetic "slit" around the perimeter of the front baffle, I would be

Continued on page 54

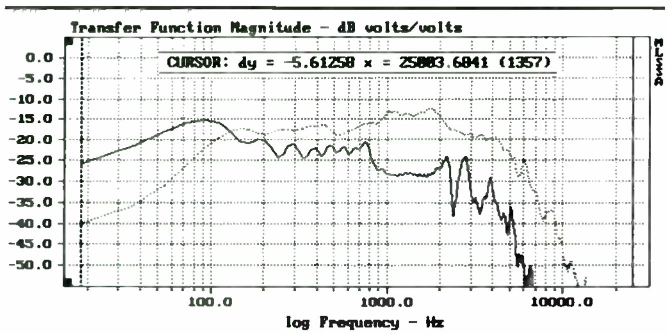


FIGURE 8: Woofer and midrange response.

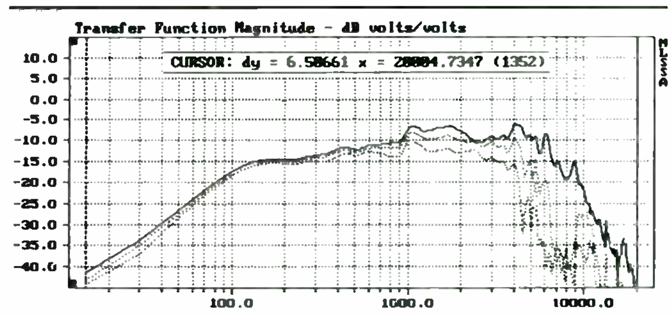


FIGURE 9: Midrange response 0°, 30°, and 45° off-axis.

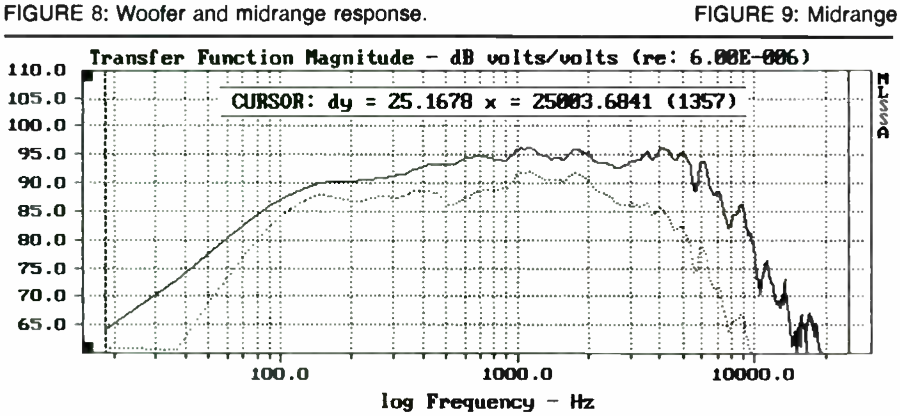


FIGURE 10: Midrange response with the network.

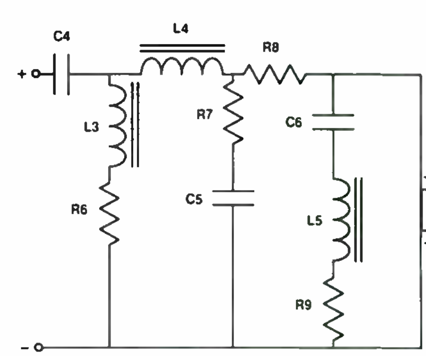
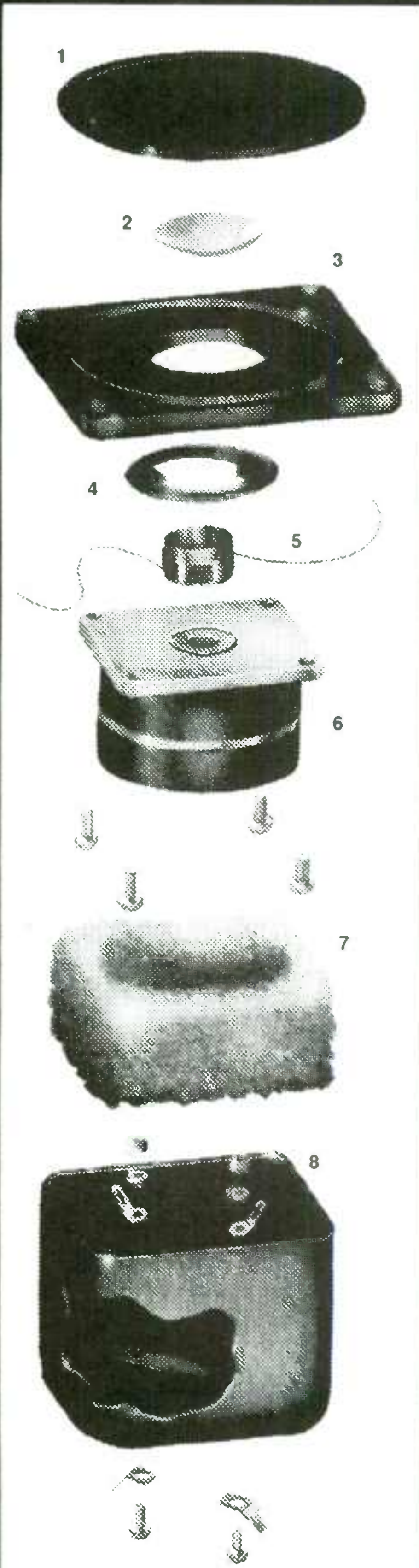


FIGURE 11: Midrange network diagram.

The Accuton ceramic concave dome units are made using the most recent materials and processes, such as laser-cutting.
They truly offer an impressive sum of techno logical prowesses.
Pure transducers , the Accuton ceramic concave dome drivers will yield precedence to music.

ACCUTON



1.Acoustically transparent screen.

2.Pure alpha crystal aluminum oxyde ceramic diaphragm. Computer calculated shape takes maximum advantage of the superior drive (energy transfer from the VC to the concave dome) attained by concave domes over traditional convex domes.

3.High mass high damping diecast frame with profiled channels.

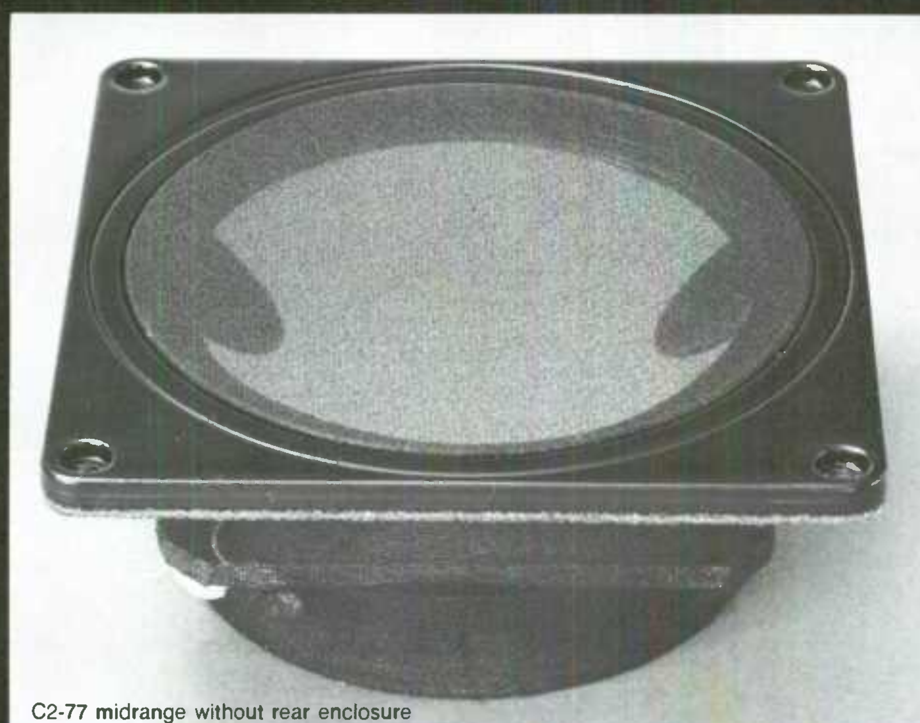
4.Linear "transparent" suspension.

5.Fluid guided VC. VC is cemented to diaphragm for full energy transfer.

6.Dual ferrite high precision magnet assembly.

7.Anechoic envelope.

C2-11 tweeter



C2-77 midrange without rear enclosure

8.Rear enclosure provides reaction-free damping of the sound waves emerging from behind the diaphragm and its suspension.

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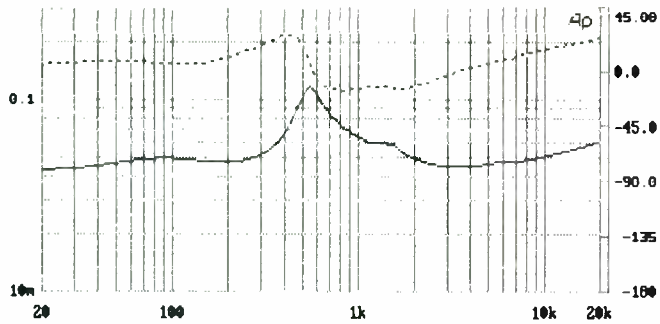

FIGURE 12: Tweeter impedance magnitude and phase.

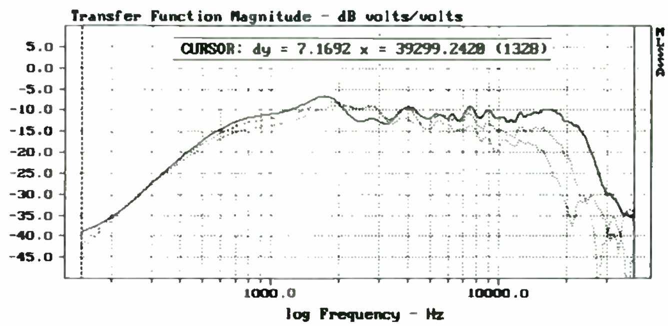

FIGURE 13: Tweeter response 0°, 30°, and 45° (no crossover).

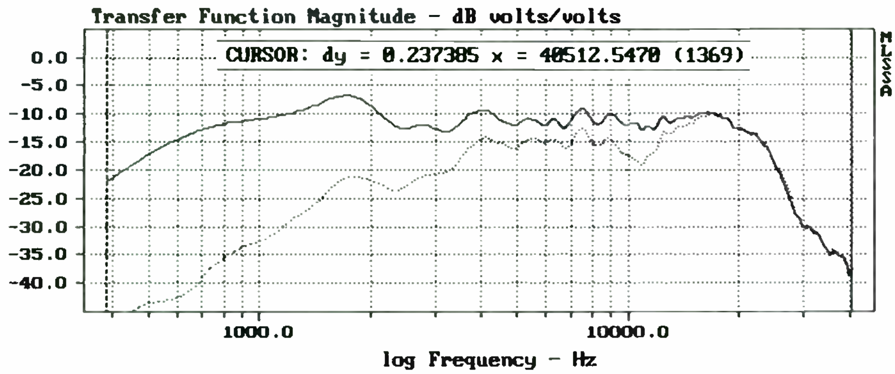

FIGURE 14: Tweeter response with network.

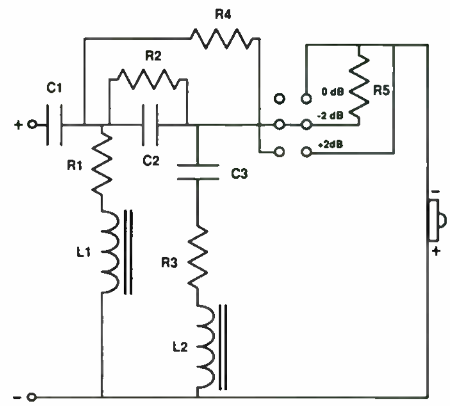

FIGURE 15: Tweeter network.

Continued from page 52

surprised if such a minor change in box geometry had a significant effect on box vibration modes.

The enclosure contains a considerable quantity of gray felt batting (approximately 80% of the volume), like that used in automobiles. This added damping alters the woofer low-end response and results in a more shallow rolloff than generally encountered with vented designs. In fact, using LEAP, this woofer's response in a sealed box of the same volume is close to the damped vented-box response measured in *Fig. 4*. The main difference is the cone excursion at high SPLs was 1 or 2mm greater than that of the damped vented design. The front baffle of the woofer enclosure, finished in oak veneer, also had an ⅛″ felt pad mounted around the woofer. This thickness altered the woofer's response only slightly in its basic operating range.

Bracing is minimal and consists of one MDF 14″ x 0.75″ x 1″ side-to-side crossbrace and another 16″ x 6″ x 0.75″ diagonal side-to-side crossbrace, both in the lower half of the enclosure. A small 7″ x 1″ x .75″ shelf brace is used for the port tube. The port is fabricated from a 23″ length of plastic 4″ diameter tube with a single 90° bend to allow it to fit within the enclosure. This is a fairly large port and it should be sufficient to prevent serious compression at high SPLs.

Figure 6 shows the near-field woofer and port measurements, windowed to 100mS

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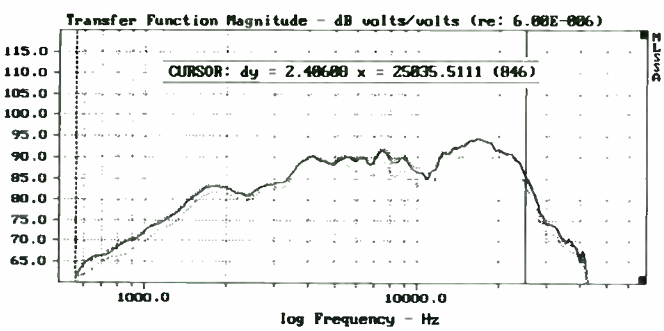

FIGURE 16: Tweeter attenuation control response.

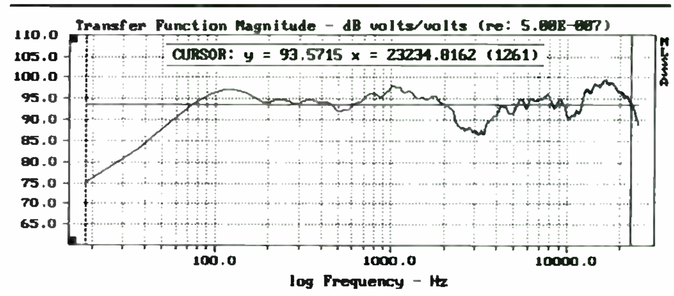

FIGURE 17: Monitor 10 on-axis frequency response.

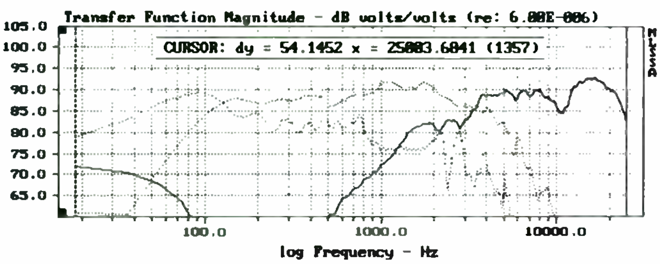

FIGURE 18: Monitor 10 individual driver responses.

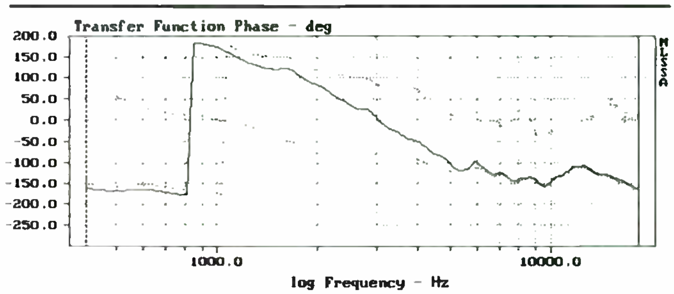

FIGURE 19: Tweeter (dotted line), mid (solid line), and composite phase curves.

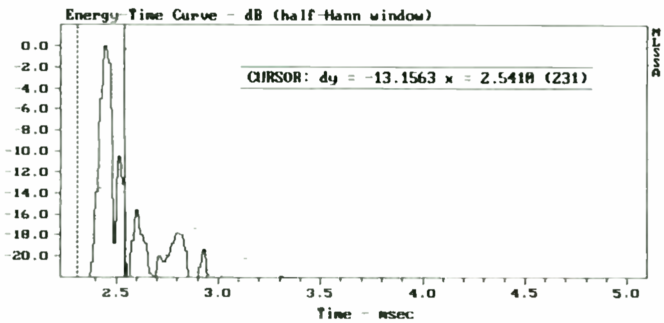

FIGURE 20: Tweeter (solid line) and mid (dotted line) ETC curves (no crossover .0935mS).

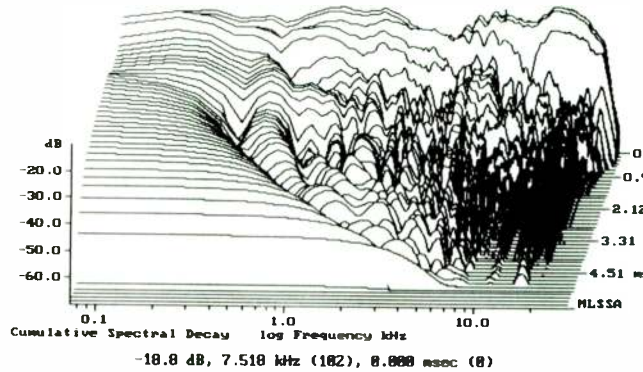

FIGURE 21: Monitor 10 cumulative spectral decay curve.

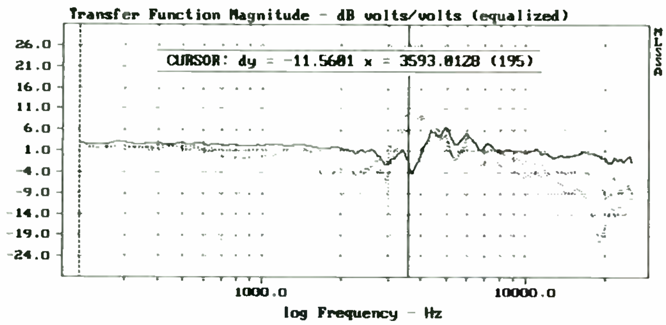

FIGURE 22: Normalized horizontal off-axis response 15°, 30°, and 45°.

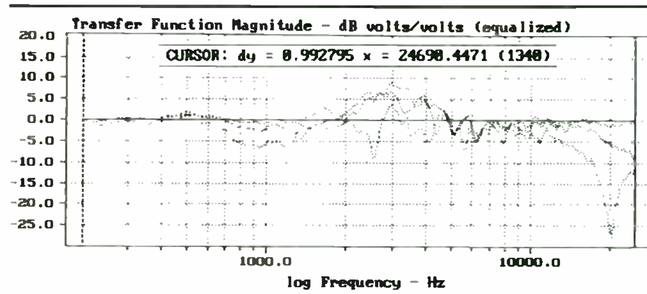

FIGURE 23: Normalized vertical polar off-axis response 15°, 30°, and 45°.

Continued from page 54
(accurate to 10Hz). Near-field measurement, with the mike placed within 0.25" of the driver, simulates anechoic far-field response and is mostly independent of the surrounding environment.[5] Since the network allows the woofer to operate without much attenuation into the midrange, the port also outputs significant upper frequency information. This is one of the arguments for having a vent on the rear of an enclosure, as it is on the Monitor 10. The long port tube tunes this enclosure to about 23Hz.

The network used with the woofer is shown in *Fig. 7*. L6 provides a first-order low-pass filter at about 120Hz, yielding the response given by the dotted curve in *Fig. 1*. The two LCR circuits (C7, L7, R10 and R11, C8, L8, R12) are conjugate filters that virtually damp the upper impedance peak and control the reactive rise of the woofer. C7, L7, R10 is an LCR filter that operates at 50Hz to eliminate the upper resonance peak of the vented enclosure. R11, C8, L8, and R12 form a complex network that compensates for reactive rise of the woofer impedance and operates with the first-order filter. It would be difficult, though not impossible, to come up with the values of this circuit empirically without computer optimization, as was used in the design of this speaker.

The Mid-Bass Driver

The mid-driver is a 6" (S_D = 123cm) type constructed with a 15oz. magnet, a paper cone, a soft polyamide dust cap, and rubber surround. Using the System 1 with the automatic Q measurement program gave a driver-free air resonance of f_S = 62.5 and a Q_{TS} = 0.54. The upper mid-bass/tweeter enclosure is constructed of 0.75" MDF with all edges and corners finished in a glossy black lacquer. The volume of the module is about 0.4 cubic feet, and is sealed and stuffed with the same material used in the woofer enclosure. The Q_{TC} for this small closed box, measured without the network, was 0.926 and had a system resonance of 99Hz.

The dotted curve in *Fig. 8* shows the response of the mid-bass driver and the woofer (including the network). F_3 of the mid-bass driver in its enclosure measures about 110Hz. In terms of sound quality, a Q of 1 for a mid-bass driver operating in this range will tend to impart a "warm" quality to the speaker. However, the measurements I made were at small signal levels, so I expect this driver would develop a fairly peaked response at higher SPLs around 90–100dB. I probably would have opted for a small signal Q_{TC} closer to 0.7–0.8.

Figure 9 gives the off-axis response of the driver without the network; this indicates a good choice for a crossover frequency would be around 2.5–3kHz. *Figure 10* shows the response of the device with and without the network; *Fig. 11* gives the diagram of the bandpass network topography. C4, L3, and R6 form a second-order high-pass section; R6 shapes the response. C4 is made of multiple NP electrolytics bypassed by a small value film capacitor. L4, R7, and C5 look like a first-order low-pass filter and a conjugate circuit ("Zobel"); R8 attenuates the driver by 3-4dB. The LCR trap (C6, L5, R9) is resonating more than a half octave above the Q_{TC} of the driver/enclosure, rather than being exactly at the box resonance frequency.

Using computer circuit optimization techniques will yield results that look at the driver response, impedance, and network interaction simultaneously, so the apparent outcome of the circuit is not necessarily intuitive. C6 is a large value nonpolarized (NP) electrolytic bypassed with a smaller value Mylar film cap. Notice the midrange driver is connected out of phase in respect to the woofer.

The Tweeter

The Monitor 10 tweeter has some interesting characteristics. The device has a large 20oz. magnet structure, but otherwise looks like a basic 1" cloth dome tweeter. The impedance curve (without the network) is shown in *Fig. 12*. The resonance occurs at 550Hz, low for a tweeter that does not use a vented pole piece or rear cavity.

The off-axis response curves in *Fig. 13* revealed the driver to have a respectable flat response out to slightly above 20kHz, as well as smooth off-axis response out to 45°. *Figure 14* compares the tweeter response with (dotted line) and without (solid line) the high-pass network, shown in *Fig. 15*.

This network has some puzzling features. A straightforward third-order high-pass is formed by C1, C2, R1, L1, and R2. R1 and R2 shape the response, and a computer optimization program such as XOPT by Peter Schuck came up with the values. C1 and C2 in this circuit are film capacitors. The LCR circuit (C3, R3, L2) seems unusual. It operates at 10.2kHz and appears to exaggerate the response anomaly between 10.2-20kHz. Emphasis in this region generally is a good way to bring out electronic defects like hiss and noise and most designers avoid it. The tweeter is connected in phase with the mid, and out of phase with the woofer.

The tweeter attenuation circuits are also not my favorite type. *Fig. 16* shows the response changes with the three different switch positions: 0, −2dB, and +2dB. R5 has the most effect, selected by putting the switch in the −2dB switch position. Switching in a series resistance like this tends to shift the crosspoint slightly; however, it is not significant in this application. R4 shorts across the high-pass filter, but only enhances the response by about +0.5dB, not the indicated +2dB.

The Whole Thing

Now for the bottom line. *Figure 17* shows the overall on-axis response of this speaker. The horizontal crosshair cursor is on in the MLSSA software so you can readily see the response is ±6dB—nothing to get excited about (all measurements were made without grilles, unless noted). *Figure 18* gives you a better sense of how all three drivers combine (ignore the tweeter response below 100Hz; FFTs tend to put out bad low-frequency data).

The dip on-axis in the vicinity of the mid-bass/tweeter crosspoint is due to their being out of phase through the crossover region as shown by the phase curves in *Fig. 19*. The drivers are mounted on a staggered baffle for time alignment™, but the tweeter is still forward of the mid-bass by 0.75″. Looking at the ETC curves for both drivers in *Fig. 20*, you see an arrival time difference of 0.0935mS (the ETC plots were processed using a half-Hann window).[6] This time difference includes the physical separation plus the difference in acceleration times for the two drivers.

The post processed MLS impulse yielded the cumulative spectral decay curve in *Fig. 21*, which indicates a lot of upper range activity beyond 3mS. This is not necessarily bad, but many of the tweeters I have reported on in *Voice Coil* will have CSDs that show activity fairly well damped by 3mS. Note this plot goes to −60dB compared to the −30dB range for CSD curves published in *Stereophile*. I think the additional 30dB of range gives you a better feeling for decay anomalies.

Instructions in the Monitor 10 manual suggest you turn the speakers inward for "improved stereo image." This means listening nearly on-axis. Unfortunately, the speaker probably sounds better off-axis. This is readily apparent looking at the horizontal normalized off-axis curves in *Fig. 22* and the vertical normalized off-axis curves in *Fig. 23*. Both graphs normalize the off-axis response to the on-axis response. This means the on-axis response appears as a flat line; the off-axis data is displayed as a deviation from that flat line.

It is easy to see that the large dip on-axis at 3kHz (the mid/tweeter crosspoint region) becomes somewhat filled in off-axis. Getting off-axis of the tweeter also reduces the 10-20kHz peak problem. This is not the preferred methodology, although suggesting the listener be 30° off-axis would probably be helpful. Ideally, you would like a flat on-axis response with as flat a power response as possible, what you hear from Dr. Floyd Toole and the folks at the NRC in Canada.

Tight quality control in a given loudspeaker pair is important for producing a good stereo image. If each speaker in a pair is carefully matched and nearly identical, the imaging quality will be maximized. *Figure 24* shows both speakers in the review sample on the same plot, indicating a deviation of less than 1dB except in the 2-5kHz region where deviation is around 2.5dB. This is not that bad, but also not as spectacular as the QC response matching done by companies like KEF.

The Monitor 10 has a large grille assembly, or hat, like the B&W 801, which is made from wire and produces minimal response problems. The Monitor 10 grille assembly, however, is made of MDF and produces a fair amount of response deviation when in place. *Figure 25* shows the response with the grille hat in place normalized to the response without it. The response region between 2kHz and 6kHz exhibits a deviation of 4-8dB.

Last, the impedance magnitude and phase plot for this speaker are given in *Fig. 26*. The lowest impedance magnitude ocurs at about 3.7Ω in the 50Hz to 90Hz reion. Looking at the Nyquist plot in *Fig. 27*, the worst case capacitive load comes at 12kHz and would not be a problem for an amplifier. The large loop in the 200-

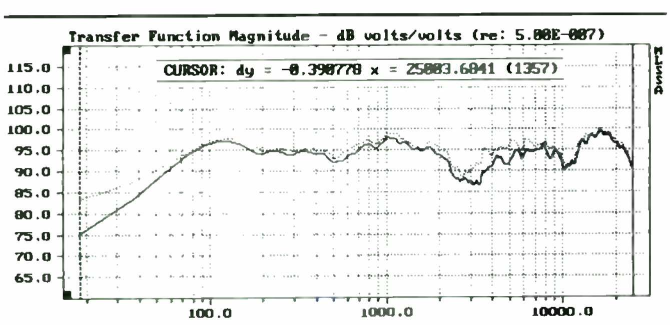

FIGURE 24: Comparison of the left and right speaker responses.

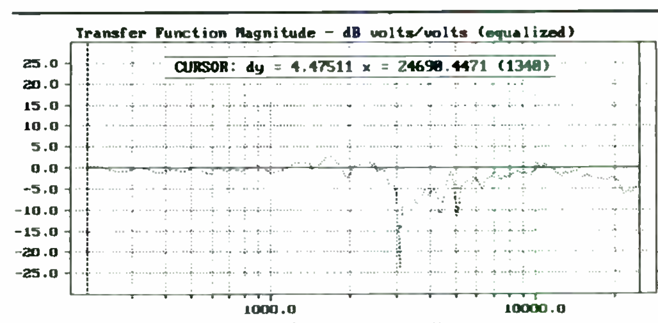

FIGURE 25: Normalized response of the speaker with the grille "hat."

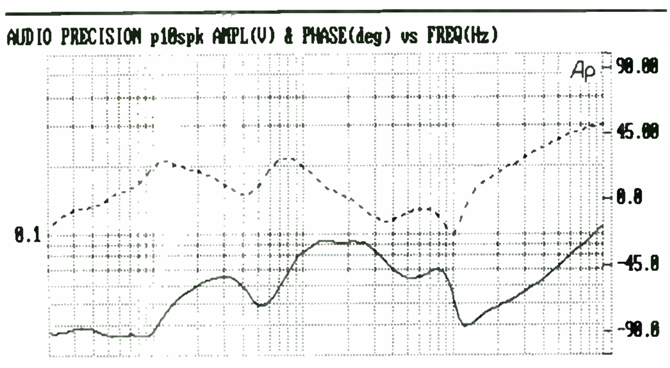

FIGURE 26: Monitor 10 impedance magnitude and phase.

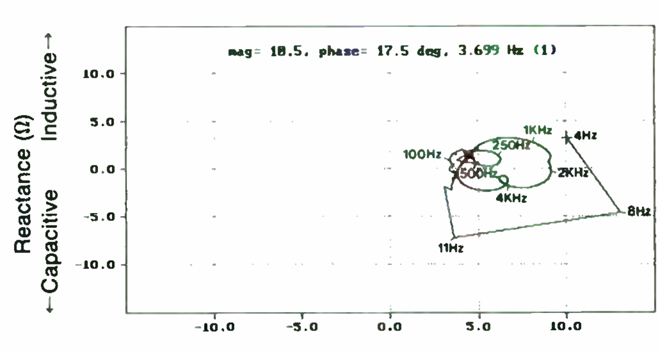

FIGURE 27: Complex impedance Nyquist plot (ohms equalized).

500Hz region is indicative of the broad overlap between woofer and mid.[7]

Subjective Evaluation

I took the Monitor 10 speakers to the Chelsea Audio/Video showrooms for subjective evaluation and comparison. Chelsea is Portland, Oregon's leading high-end salon retailer, and the listening session was arranged courtesy of Chelsea's top professional salesperson, David Bunton. The panel of "experts" was composed of me, Larry Arntson, and David. Larry has been in the industry for more than 20 years and does professional loudspeaker design consulting. David has been selling high-end hi-fi for close to 20 years and at one time or another has heard it all.

Two other loudspeakers (the Thiel CS 2 and the Infinity RS 7) were used as references for this evaluation. The CS 2 is a three-way 8″ design retailing for about $1,650, and the RS 7s are a three-way 12″ design retailing for about $1,400. Since the Precise Monitor 10 retails for $1,600, the other speakers were ideal comparisons.

The listening setup included a Mark Levinson No. 23 amp and No. 26 preamp and an Accuphase DP80 CD player. The speakers were connected to the electronics using Monster Cable, which, considering the associated equipment, was somewhat pedestrian but adequate. (The same setup was used for demonstrating Chelsea's new pair of Thiel CS 5s, which sounded better on the Monster Cable than some of the more exotic types, and we didn't bother changing it.)

A variety of CDs from rock to classical were selected from David's collection of store demonstrations. The Monitor 10s were clearly outclassed by the others, especially the CS 2s. Bottom end detail on the Monitor 10 is mediocre at best and overall spectral balance was judged poor, including a tendency toward excessive sibilance. All of us believe the Monitor 10 would be difficult to listen to over a prolonged period of time, an impression I think directly traceable to the light paper cone 10″ driver. This loudspeaker is nicely finished, but its overall execution is inadequate for a product in this price range.

Several modifications have been made on the Monitor 10 since I received the samples for review. However, this review is fairly consistent with Ed Long's evaluation of the same speaker (and a current version) that appeared in the September 1990 issue of *Audio* (pp. 98–110). I understand Precise has a new version in a different enclosure with new cosmetics and using similar drivers, which may replace this speaker at a lower price.

[*We did not receive the manufacturer's comments to this review at the time of publication—Ed.*]

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1. Cabot, Richard C., "Automated Measurement of Loudspeaker Small Signal Parameters," 81st AES Convention, Preprint 2402.
2. Strahm, Chris N., "Complete Analysis of Single and Multiple Loudspeakers," 81st AES Convention, Preprint 2419.
3. Just so there is not any confusion, I am employed by Audio Teknology Inc. which produces LEAP (the loudspeaker enclosure analysis program), although our primary effort is in the manufacture of a high-performance $90,000 live sound reinforcement mix console. In fact, if you buy a copy of the program, you will generally talk with me about it. Although it would be hard to say I am not prejudiced toward its usage, I also do not know of any other low-end analysis software that can provide the same dynamic analysis accuracy. If you can't afford LEAP, however, Bob Bullock's BoxModel is a good alternative.
4. Gander, Mark R., "Ground-Plane Acoustic Measurement of Loudspeaker Systems," *Journal*, AES Vol. 30, No. 10, Oct. 1982.
5. Keele, Jr., D.B., "Low-Frequency Loudspeaker Assessment of Nearfield Sound-Pressure Measurement," AES *Loudspeaker Anthology*, Vol. 1, p. 344.
6. Vanderkooy, John and Stanley P. Lipshitz, "Uses and Abuses of the Energy-Time Curve," 87th AES Convention, Preprint 2857.
7. Dickason, Vance, "How to Plot and Understand Complex Impedance," *Speaker Builder* 2/88, p. 15.

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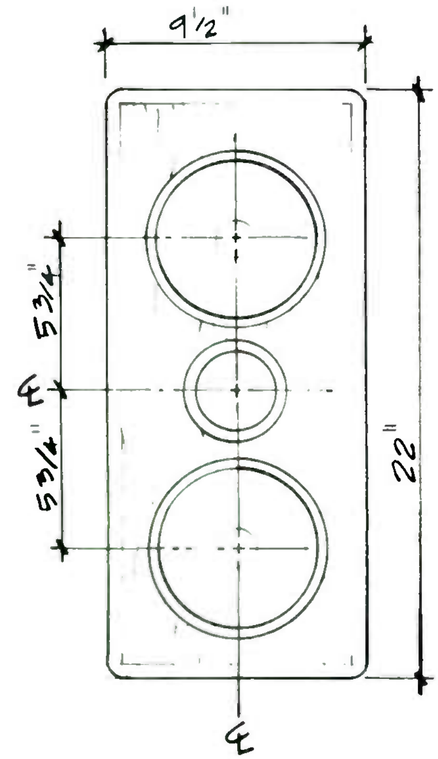

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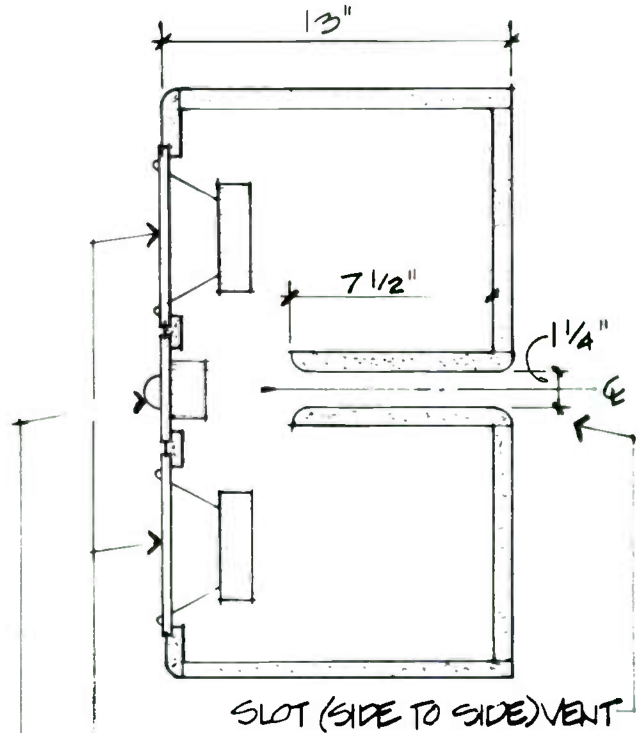

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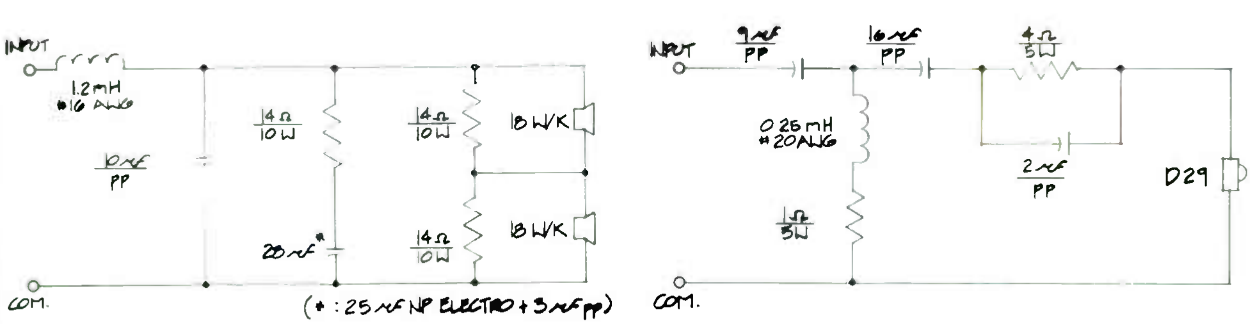

CROSSOVER

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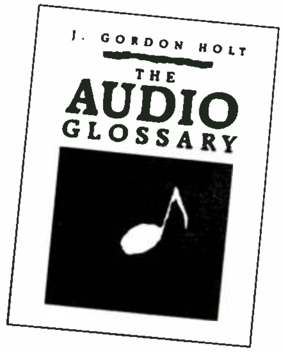
Book Report

The Audio Glossary

By Peter E. Sutheim



The Audio Glossary, **by J. Gordon Holt**, 1990. Available from Old Colony Sound Lab, PO Box 243, Dept. B91, Peterborough, NH 03458-0243, (603) 924-6371, $9.95 (softcover), $17.95 (hardcover), $30 (limited edition).

A glossary, says a dictionary (which is not the same as a glossary), is a collection of specialized terms with definitions. The specialization here is entertainment audio—or what the lay reader would call "stereo" (as in "Wanna hear my new stereo?"). It is the reference that, as an audio writer and broadcaster, I have wished for hundreds of times to respond to perplexed queries from readers and listeners. "Do you know," goes the typical question, "of a book that would teach me the basic stuff or at least explain the terms so I could understand them?"

Again and again, I would sigh and say I knew of none, at least no one book, to satisfy that need to know. Textbooks, professionals' handbooks, engineering manuals, directories, and consumer guides proliferate (and have, typically, a three-year half-life, or so it seems). But books directed at the ordinary Josephina, who wishes only to inform herself well enough to stare down an audio salesperson, have been few.

None of the more or less current books[1] has a glossary close to the scope of Holt's—understandable but hardly acceptable. Granted a glossary is a book in itself. Readers of these technotexts feel a need (and, one could argue, have a right) to have things spelled out. What's more, few things aid clear thinking like wrestling to the ground a term such as *absolute phase (or polarity)* in 29 words, as Mr. Holt has done. Aside from the space requirements of a glossary, the labor demanded is of a heroic order (Holt's glossary boasts 1,900 definitions); moreover, many skilled writers lack the nearly lunatic methodicalness called for.

A good glossary includes alternative terms, draws distinctions between similar notions, and, perhaps most importantly, tries to be reader-friendly. It is, therefore, full of cross-references, partly to implement two conflicting goals: comprehensiveness and conciseness. Cross-referencing allows the definitions to be complete without needless repetition. Cross-references also adds depth to a glossary. They stimulate curiosity. Extensive cross-referencing is one of the most attractive features of Mr. Holt's book, which received extensive editorial assistance from fellow audio journalist Peter Mitchell.

Almost every time I found a technical term that, from the standpoint of a nontechnical reader, needed an explanation, one was available in its proper alphabetical place. However, I would have preferred not to guess at whether or not a term was defined elsewhere: I wish such terms embedded in a definition had been distinguished typographically (boldface, italics, small capitals) to pop out.

The Choice of Words

What of the terms chosen and their definitions? Obviously, one of the most fundamental decisions is what to put in a glossary and what to leave out. Holt seems to have leaned toward inclusion, resulting in some eccentrically playful entries: *blivet* and *thingamabob* are in, but *gizmo*, one of my favorites, is not; in the perhaps slightly more earnest department of subjective sonic description, *chalky* and *plummy* are in, but *chocolaty* is not.

For those wishing to make sense of subjective reviews using metaphors, Holt's glossary will be of some help, but words whose semantic fields overlap are not compared or distinguished. For example, are *sweet* ("having a smooth, softly delicate high end) and *silky* ("treble...that is velvety-smooth, delicate, and open") synonyms? Or could more be said about each and where they overlap?

Editor Mitchell suggested a separate section of the glossary for subjective terms, but it was evidently not taken up. I find this ironic because in the early years of *Stereophile* (founded in 1962), Holt promulgated a potentially helpful array of descriptive terms, two for each of the ten octaves of the audible spectrum. The first of each pair was used to name a deficiency in that band, the second an excess. The system never caught on, although the words were common and well chosen—perhaps because audio writers, all madly individualistic, would have had to grant Holt too much authority. Absorbed into the mainstream of audio jargon, those words would have helped hundreds of equipment reviewers and millions of readers pin down more precisely the sonic attributes described, rendering subjective reviewing a little less arbitrary. Perhaps it's time to republish that list.

In its coverage of engineering, scientific, and mathematical terms related to home audio, Holt's book conveys the greatest authority. Almost all the terms for unfamiliar words and phrases you might find in a review are included. The book ranges widely, sometimes into surprising regions. *Boolean logic* is given a quick definition and so is *three-phase supply*, neither term in the first rank of audio-related terminology. My guess is they appeared in *Stereophile* and were tagged for inclusion.

I was surprised to find *acceptor* and *donor* (two arcane terms from solid-state physics) and surprised, though pleased, to find those terms meticulously cross-referenced through *doping*, a technique used in producing semiconductors. A few oddball entries raise the eyebrows, such as *vaudio* ("the audio portion of a video program") and *bipole* as distinct from *dipole*, even though the distinction is not made clear. (I have not been able to find bipole in any other dictionary I have, so I'm not sure what it means, either.)

When Holt moves beyond the areas most familiar to him (he is not an engineer), the authority is wobblier. Tackling acceptor, Holt adds to an opaque but correct one-sentence definition ("An electron-deficient dopant used to produce a p-type semiconductor material") the following unhelpful and almost irrelevant second sentence: "Indium is frequently used as an acceptor material in german-

ABOUT THE AUTHOR

Peter Sutheim has written for numerous audio and electronics magazines for the past 25 years. He heads the Audio-Visual Department at Occidental College in Los Angeles and produces a weekly audiophile radio show.

ium." True, but when did you last see a germanium transistor in a new product? And is it useful to know the names of these elements without knowing anything about those properties making them useful in semiconductors?

Music terms also contained boo-boos. Holt defines *bass clef* as "the lower staff of two, which includes the key of F." It's not a staff at all but a symbol that designates the range covered by the staff it's written on. A staff so marked may not be the lower of two, even in the two-staff piano scores implied; it may be one among several in an orchestral score. It no more and no less "includes the key of F" than any other clef does—it's just that the F below middle C lies, by convention, on the fourth line up. *Treble clef* is clobbered with a similar solecism. Is it important to have included these at all? And was it necessary to include *piano*, the instrument, along with the dynamic indication for soft? (*Violin* is not included, nor is *cello*, although *violining* is.)

Some words ought to have been included that weren't. *Uncorrelated* is used in the definition of *ambience extraction* but is not defined. *Correlated* is, but only in the following sense: "high degree of similarity between the L and R stereo signals." This does apply in ambience extraction, but leaves out the sense in which it was used. For example, Carver's *autocorrelator* is described as an ingenious analog noise-reducing device that gated-out noise by sensing its "uncorrelatedness," as contrasted with music, which had a much greater degree of correlation (harmonic relationship).

Dimension, a much abused buzzword in commercial writing, has its serious meanings, which ought to have been recaptured and fixed within an audio context. A core concept of Richard Heyser's work was that music-listening is a multidimensional experience. Suitably chosen dimensions (pitch, intensity, width and depth, and time) are mutually exclusive and do not correlate well to the reduced number of dimensions in the audio signal (intensity and time). This, according to Heyser, has profound consequences, some seen in the seemingly irreconcilable arguments between the measurers (who deal with a reduced number of dimensions) and the listeners (who experience music holistically and yet try to make sense of what they hear in the engineers' universe of voltage and spectrum).

It is ironic this potentially major contribution to a resolution of the endless dispute between measurers and listeners has received so little attention since Heyser's death in 1987 and has been branded as too obscure to be helpful. A brief entry in Holt's glossary might have helped to stimulate more interest.

In a few cases, adding a word or two would have made a definition more complete, as with *cathode-ray tube*: "Deflection coils control the position of the luminous dot." To write "Deflection coils or plates" would have included the family of electrostatic-deflection CRTs as well. *Alnico*, a magnetic alloy (not a compound as stated), contains significant amounts of iron and copper as well as the elements listed. These may be mere quibbles to many, yet it with this kind of thoroughness that real authority rests.

Then there are the off-the-wall, nutty inclusions that will make some giggle and others wince. *Cassette*: "(1) A removable case containing a length of magnetic tape and its supply and takeup spools. (2) A small cass." Or the second definition offered for *backing*: "investment capital."

Concluding Remarks

The personal preferences and prejudices manifested throughout Holt's book are endearing, if sometimes exasperating. (See Holt's definition of *hangover*, *hi-fi*, *exotic*, and *aleatoric*.) They reveal it as the work of one man. Praiseworthy though the effort be, it is compromised by some of the qualities that commend it. One can't help wishing Holt had consulted a musician, a metallurgist, or even various dictionaries to check the accuracy of his definitions.

Errors in fact or judgment could have been fixed by peer review, a standard practice exhaustively employed in the preparation of every kind of authoritative dictionary. Holt's Glossary is more like a Poor Richard's Almanac—pithy, witty, interesting, helpful, but hardly to be relied on as the final authority in the field.

I find this balance between the personal and the anonymously authoritative most exasperating about Holt's book. No reader would mistake Ambrose Bierce's *Devil's Dictionary* for a Merriam-Webster. *The Audio Glossary*, however, presents itself as a serious work larded with personal observations definitely not to be confused with *Bluff Your Way in Hi-Fi* (Ravette, England, 1987). Yet reading it is like spending an evening with an affable companion who seems to know what he's talking about, but who now and then pulls a fast one on you, straight-faced—mostly intentionally, sometimes not. Fun, but not quite, well, reliable.

Aside from random minor editorial or typesetting errors, which are almost impossible to weed out completely in a first printing, a few more systematic typographical mistakes stand out. Hyphens appear where minus signs ought to be, sometimes confusing the sense of an expression. I also found it annoying, as mentioned earlier, that technical terms within a definition are not typographically highlighted.

The outcome of Holt's immense and well-intentioned labors is a useful but incomplete book. Still, even with its faults, Holt's glossary deserves a place on every audiophile and audio writer's shelf. It's a courageous work. Anyone who thinks he can do better is welcome to try.

JGH responds:

Some of Mr. Sutheim's criticisms are well taken. If ever a revised *Audio Glossary* is published, they will be taken seriously. The others are not, and will not be.

- A bipole is a dipolar device whose front and back radiation is in-phase. That is clearly stated in the definition.
- Although silicon is the semiconductor material in most modern bipolar transistors, germanium is the material with which most people who know a little bit about transistors are most familiar.
- Vaudio is a useful, if ugly, term to use when discussing a device that accepts inputs from audio/video and audio-only sources.
- The bass clef is defined as that staff on which F below Middle C is on the second highest line.

Perhaps I overestimate Mr. Sutheim's opinion of my encyclopedic knowledge base, but his suggestion that I wrote this whole thing off the top of my head, without consulting experts in their fields and dictionaries in those fields, is at once flattering and insulting. Peter Mitchell, certainly a worthy "peer," spent many hours annotating and commenting on the original version, and, practically all of his suggestions were followed. It was further checked by editor/publisher Ed Dell and his technical staff. Is Mr. Sutheim really suggesting that this should have been a committee effort?

The reviewer seems to feel I committed a grievous sin by not legitimizing his apparent worship of the late Richard Heyser's writings. I will grant him that Heyser was a master measurer, but I am not the only intelligent person in high-end audio who feels Heyser's forays into theoretical psychoacoustics were more New-Age than scientific. Physiologically, our ears are strictly Old-Age; even after several billion years of evolution or a few days of creation (whichever way you look at it), they still respond to nothing more than frequency, intensity, and time, no matter how exhaustively and tediously we try to analyze the sounds fed to them.

As much as I would love to be deliciously irked by Mr. Sutheim's nitty-picky review of my book, I really can't be. After all, the magazine I founded in 1962 created an environment where equipment reviewers who had never designed so much as a plug, felt free to explain to preamp designers how they *should* have done things and to demonstrate their hypersensitivity to sonic pollution by pretending a barely audible imperfection was a FATAL FLAW. Mr. Sutheim's review is in complete harmony with this "madly individualistic" approach to reviewing.

But "of some help" to those trying to make sense of subjective reviews? Sheesh! Peter, you're a nice young lad, but your judgment needs more time to mature. This Glossary was originally conceived *specifically* as a lookup reference for readers of *Stereophile*, *The Absolute Sound* and the other magazines whose product reviewers hear differences between products and try to describe what they hear.

Originally, it covered *only* subjective-testing terminology, and this is the book's strongest area. No other has even attempted to explain subjective audio's "arcane" terminology, let alone do it this well. (And no other will ever be written by the person who devised and pioneered the use of most of these terms.) The rest of the glossary was added after I came to the rather appalling realization that most readers of audio magazines don't know diddly-squat about *anything* electronic, and were not likely to go out and buy a $75 book to find out about it. Yes, there may still be inaccuracies in *The Audio Glossary*, but there are very few, and none of them is outright wrong—only somewhat misleading.

My only remaining question about Mr. Sutheim's review is: Why the ill-concealed hostility? Perhaps, someday, he will explain.

REFERENCE

1. The most recent glossary is Laura Dearborn's *Good Sound* (Quill Div. of William Morrow, 1987), which I have not seen, although a telephone query to the publisher revealed it contains a three-part glossary of musical terms, audio terms, and listening-evaluating vocabulary, with about 100 entries.

Ivan Berger and Hans Fantel's *The New Sound of Stereo* (New American Library, 1986) is good as far as it goes but rather uncritical in its acceptance of the questionable position that more advanced technology means better results, and is already somewhat dated (no fault of theirs). It has an index but no glossary.

Judy Davidson produced a breezy, illuminating, tradition-breaking home-audio guide, *The Quickest, Easiest Guide to Stereo* (Lancaster & Harrow, 1985; out of print), that tilted in the opposite direction (after all, no one said a flat earth has to be level).

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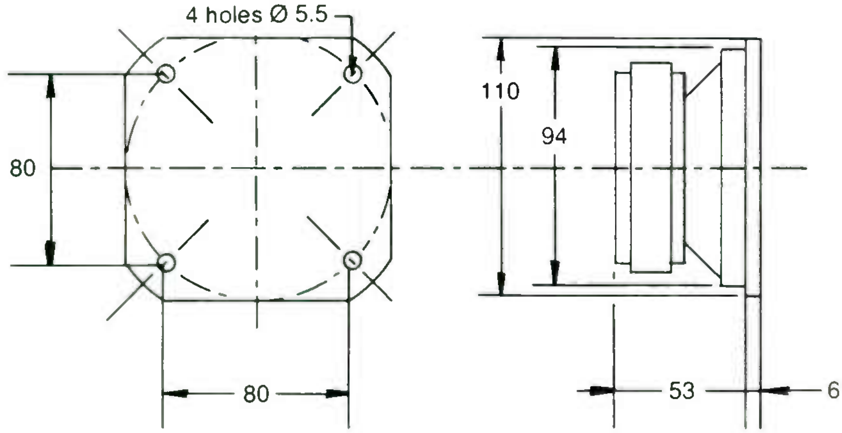

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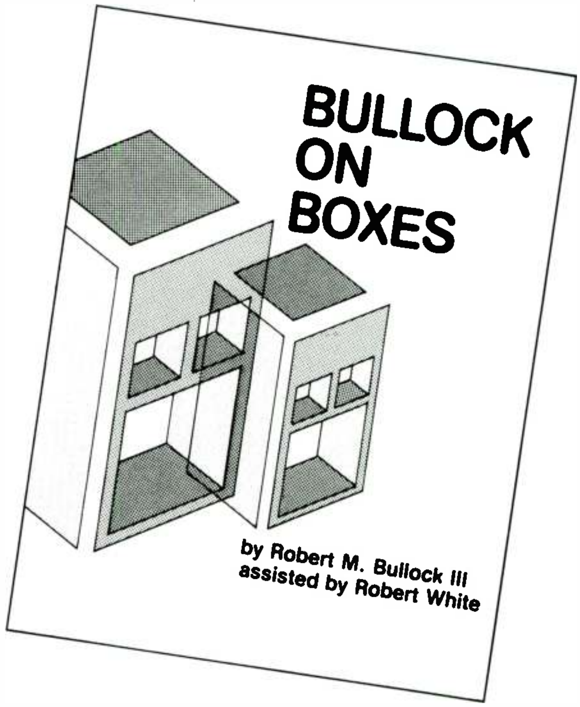

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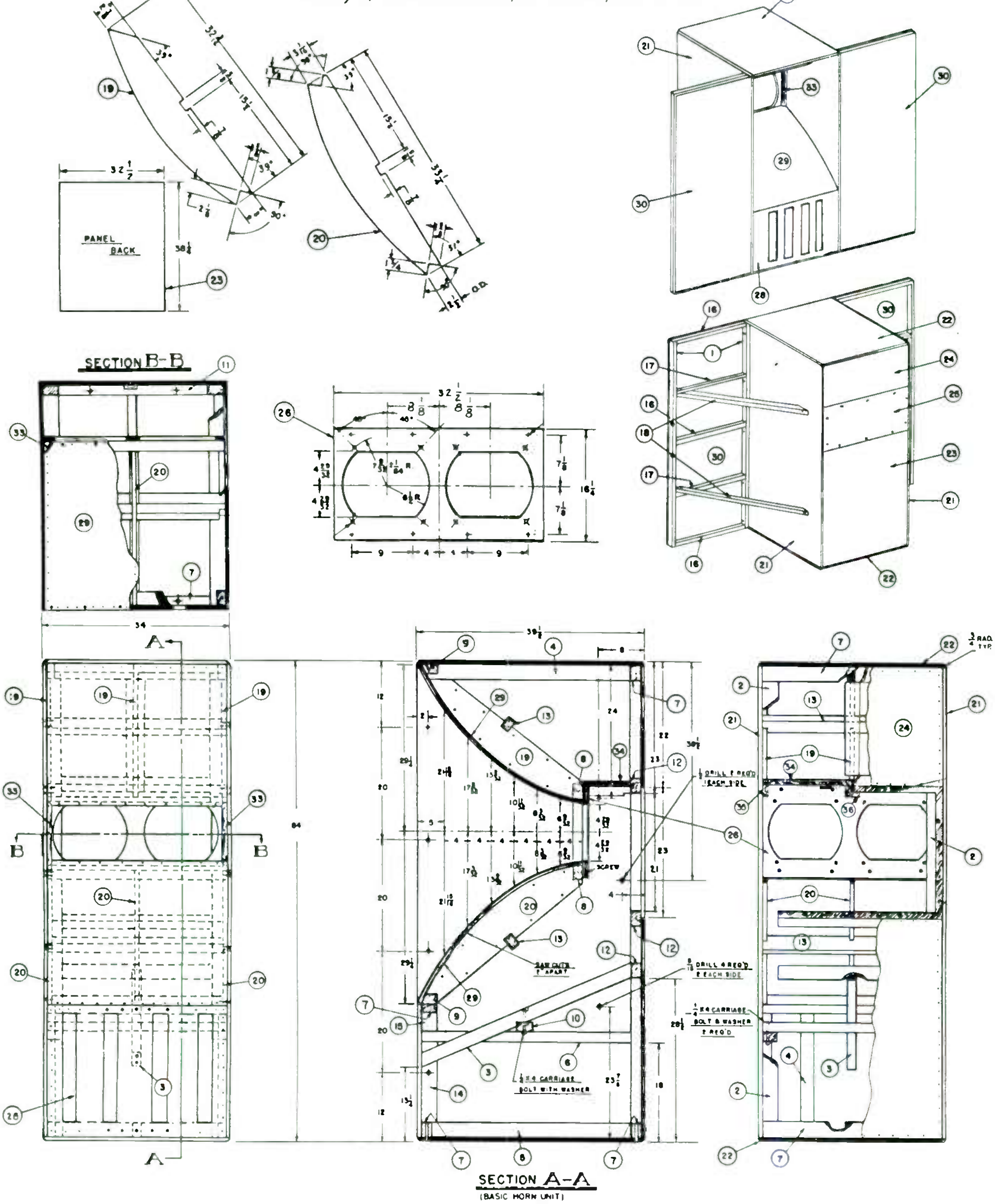


Book Report

Musical Acoustics

By Thomas W. Parsons

Fundamentals of Musical Acoustics, by **Arthur H. Benade**, New York: Dover Books, 1990, paperback, 608pp. Available from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371, $15.95.

In 1977 we published a review of this book, then just issued by Oxford University Press, but now out of print. Again Dover Books have worked their everyday miracle by retrieving the work from oblivion, updating it from the author's notes three years after his death, and reprinting the whole in a fine paperbound edition. The book is everything our reviewer Tom Parsons said it was, and reads like a novel. I strongly commend it to you for a delightful and highly illuminating reading.—ETD

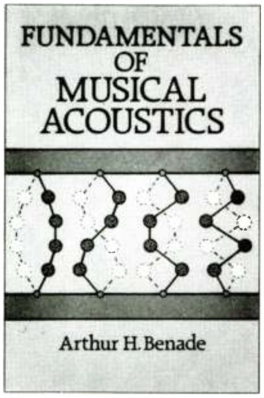
Arthur H. Benade's *Fundamentals of Musical Acoustics* represents a remarkable achievement. Most audiophiles have probably looked into an acoustics book at least once. The result is usually discouraging—long on mathematics and short on realistic applications. (In my experience, their woodwinds never have finger holes.)

Picture, then, a book on musical acoustics by an acknowledged master of the subject. The book guides the reader through the most subtle intricacies of every important family of Western musical instruments with virtually no mathematics, a bare minimum of equations, and no functions more complicated than an occasional square root. Benade, a professor at Case Western Reserve, has set forth a vast knowledge of musical acoustics so clearly you can read his book as you would a novel.

The Theory

The book is divided into two parts. The first, an introduction to basic theory, accounts for half the book; the second presents applications. The theory is developed by taking you through a series of experiments you can read as "thought experiments" or actually carry out. These experiments use commonplace (occasionally even ludicrous) equipment and are designed to help you develop an intuitive feel for the way vibrating systems work.

They familiarize you with such concepts as resonance, damping, modes of



vibration, frequency components, transient and forced oscillations, addition of components, and reciprocity. The presentation is always in terms of simple physical models and mathematical rules. Indeed, your first impression is one of being talked down to, but it quickly becomes clear this is not the case. The simple-sounding introduction lays the groundwork for detailed analysis of musical instruments.

Later, you will learn what finger holes do to the acoustics of a woodwind, why a harpsichord string sounds sweeter when plucked slowly than when plucked quickly, why a kettledrum has a vent in the bottom (it's not to equalize air pressure), and why recorder players have more trouble with intonation than string players do.

The author avoids higher mathematics because the book is not written for fellow physicists or graduate students. It is for the general reader, particularly for musical-instrument makers, some of whom he has been a consultant to for year. For these readers, equations only becloud the issue and are no substitute for the insight behind equation writing. This insight Professor Benade labors patiently to develop.

I have already listed some of the concepts introduced in the first part of the book. Damped vibrations are introduced by the clang of an ordinary skillet when struck by some hard object. This is a bizzare note (in more ways than one) on which to begin, but the complex vibrations of a skillet clang are a more relevant starting point for musical acoustics than (say) a tuning fork would be.

Modes of vibration, frequency components, and the superposition principle are illustrated by a pair of large steel nuts supported by rubber bands. (Several vibrational modes can be excited by hand in an ordinary retractile telephone cord; Professor Benade seems to have missed this one.) Transient and forced oscillations are induced in a large plank, hung as a pendulum and excited through a string of rubber bands by a motor drive crank.

The transfer functions of a system with many resonances are illustrated in a tin tray. In every instance, Benade strives to use ordinary objects—common enough not to unsettle the technically diffident reader and (with the possible exception of the motordrive crank) easily accessible.

After every experiment, Benade summarizes what has been learned in a series of numbered assertions. These serve to organize the test into alternating sections of exposition and summary, and after awhile you begin to watch for them. Another attractive feature is his use of "digressions," paragraph-long parenthetical explanations of points you may have been wondering about. By labelling them as digressions and putting them in smaller print, he keeps them from disrupting the flow of his discourse.

From these basic concepts, the book proceeds to room acoustics, to our perception of sounds (touching on the Fletcher-Munson curves, the loudness of complex tones, the localization of sounds, and the precedence effect), and to the phenomenon of pitch and the problem of equal temperament. With these topics, the theory ends.

The Instruments

In succeeding chapters, he analyzes different families of instruments in a style that's clear and easy to understand, but is far from elementary. His discussions of detail are difficult to find elsewhere, unless you have the time and ambition for an exhaustive literature search.

The discussion of equal temperament leads naturally to the piano, harpsichord, and clavichord. You learn how "inharmonicity" in the partials of piano and harpsichord strings makes equal temperament more palatable on them than on the organ (but you aren't told why harpsichord builders apparently scaled their instruments to minimize inharmonicity).

Next comes the voice, the first of the continuously driven instruments. The oscillation of the vocal cord subsequently serves to introduce that of the lips of the brass player, and the chapter on brasses similarly prepares you for two important chapters on the woodwinds. The brasses and woodwinds are Professor Benade's own research interest, and these are possibly the three best chapters in the book.

Anyone who owns a prized wind instrument will probably be alarmed at the descriptions of scraping, sandpapering, and lacquering to turn a mediocre woodwind into a first-class one. I wouldn't recommend trying it—this material is obviously directed at professionals, who have a fair number of duds available for tinkering.

The book ends with a discussion of the bowed strings and a chapter on musical curiosities (special effects, the "wolf not," and the like). Our present knowledge of violin acoustic is largely due to the pioneering work of Carleen Hutchins and her associates. It seemed Professor Benade's exposition of the placement of resonances in a food violin was less clear than Miss Hutchins' own, but his description of the "stick-slip" mechanism of bowing is good.

Concluding Thoughts

I was sorry to see no mention of the violin-varnish mystique, and John Schelleng's paper (which reveals the "secret" is to use as little varnish as possible) is mentioned only among the references. Maybe most lutists know this already.

Reading the chapters on brasses, woodwinds, and strings, I was struck by our primitive knowledge of keyboard instruments. Little is known about the distribution of soundboard resonances, for example, either in frequency or physically about the soundboard. To my knowledge, no one can fix up a failed piano or harpsichord as Professor Benade can a bassoon. I suspect research has been inhibited, not by complexity, but by sheer physical size of these instruments.

Otherwise, the past 20 years have seen progress in our understanding of musical instruments. Most has lain hidden in the journals until now. We would have cause enough to be grateful to Professor Benade if he had only unearthed the material and put it in his book. He has not been content merely to give us the results, however, but has also taught us the methods by which they were found.

With the exception of room acoustics, not much of the information is relevant to audiophiles. But for anyone who has wondered about the way the sources of his program material work, this book is a must. Even at $15.95, it's a bargain.

Ask Speaker Builder

THE DUST CAP SOLUTION

By Richard Pierce

When most people buy loudspeakers (if they buy them separate from some "rack" system or a boom box), they do not realize they are buying a "system," consisting of several drivers, a crossover, the cabinet, and so forth. To them "designing" a loudspeaker might consist of nothing more than selecting components from a catalog and sticking them in a box, without any thought of making sure things worked together properly. Regrettably, most loudspeaker makers do exactly that.

Even some of the more conscientious builders must resort to buying their drivers from a catalog. Setting up a production line to build drivers is expensive and complex, beyond the reach of most builders. They must trust their vendors to supply them with products that most closely match their requirements.

Of Manufacturers and Components

You can choose from dozens of driver manufacturers in this country, Europe, and the Far East. All vendors have their own set of criteria and prejudices to either the advantage or detriment of their customers. Some are capable of designing and manufacturing many if not all of their products' components, and they have a great deal of expertise, capable of producing the highest quality driver for a given price and application. This capability does not come cheap. It requires a tremendous investment in equipment, research facilities, and most importantly, expertise, of which there is not a lot to go around.

Other driver manufacturers simply choose their components from catalogs and assemble them. They often have no idea what constitutes good performance and have no means of measuring it anyway. They might not even have any design expertise on staff, simply mixing and matching components, sending the resulting samples off to customers until they are either happy or go somewhere else.

I've even encountered one Far Eastern manufacturer who was unable to perform the simplest assembly chores. After going round and round a half dozen times, trying to get the right compliance on a 6½″ woofer, I finally sent a FAX to the plant with a parts list, specifying each component by manufacturer, model, revision number, and color. I received a return FAX saying they understood what I was looking for and to expect an evaluation sample in a month or two.

The appointed day arrived, and on my doorstep was a box, air-freighted directly from the plant with the samples inside. I quickly opened the box to be faced with a pair of 10″ woofers and a note saying I should be happy with the samples, built especially to my order.

The problem with building a driver is the same as with building a system. Each component works in a not-so-subtle and not-so-obvious way to contribute to the performance of the driver. Minor details such as whether to use 0.1 or 0.2 gram of some adhesive on a magnet could lead to a 3dB difference in efficiency.

A System's Approach

As a design engineer for a driver company and a consultant to many speaker builders here and abroad, I understand that designing a driver has to be approached from a systems viewpoint. All too often a "system" is designed starting with components on hand, then trying to integrate them into the final combination (often referred to in software engineering as bottom-up design, or what I call "solutions looking for problems").

My approach has been to try to understand what the system is supposed to do, then to divide the problem into successively simpler problems until the solution becomes obvious (known as top-down design). This often means the specifications for which drivers to use is one of the last decisions in the design cycle. This way, the problem defines the solution, not the other way around.

This does not mean you can ignore the interaction between components. This design method forces you to face this problem every step of the way. It disciplines you to deal with the interface between components and subsystems and guides your design accordingly. It's not the easiest system to adopt and maintain, and few driver manufacturers seem to consider it important. Which brings me to the subject of this column. Two people have sent in questions about a seemingly innocuous component of drivers, the lowly dust cap.

For example, Tom Miller writes:

"I have two pairs of old small Tandberg systems. They have a 6½″ woofer. I have been unable to measure their free air resonance.... Either it is extremely narrow or essentially flat. I have also noticed the voice coil cover is far from air tight. It is something like gauze. My main reason for writing is to get some comments on what impact this may have on performance. Would coating or replacing this cover improve bass performance? Is this tied to the resonance measurement?"

Mithat Konar asks (somewhat more philosophically):

"What is a dust cap supposed to do? What does it *really* do?"

Let's deal with the more basic question first: "What is a dust cap supposed to do?"

The Voice Coil Gap

The electrodynamic drivers, woofers, midranges, and tweeters that inhabit the majority of loudspeaker systems generate sound by moving a cone or dome that, in turn, moves the air. The cone itself is moved by a coil of wire immersed in a magnetic field. When current is passed through the wire, a force is generated at right angles to the direction of current flow and the direction of the magnetic field. This force moves the cone.

In practical detail, the coil of wire, called the voice coil, is wound around a cylindrical former, ideally made of some rigid material such as aluminum, plastic, or in some cases, paper. The cylindrical shape of the former gives it much of its rigidity along its axis. The voice coil sits in the cylindrical voice coil gap. The pole piece of the magnet defines the inside diameter of the gap, while the hole in the magnet's top plate forms the outside diameter.

To maximize the strength of the magnet field in this gap, it needs to be as narrow as possible. At the same time, there must be enough clearance to allow the voice coil to move freely in this gap without touching the pole piece or the front plate. This clearance can be as little as a few thousandths of an inch for a tweeter, where the range of motion is quite small, to several tenths of an inch for large, high-power woofers, where the range of motion and chances for misalignment are greater and the actual size of the voice coil can vary immensely with temperature.

When assembling a loudspeaker driver, the maker must have access to the voice

Continued on page 70

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AUDNET62.EXE	102k	5/29/90	More Rec.audio talk
AUDNET63.EXE	79k	9/17/90	More rec.audio talk
BODE10.ZIP	86k	10/06/87	do BODE plots on Herc,CGA,EGA
BOUNDRY.ZIP	41k	3/04/90	Speaker placement program
BOXDZINE.BAS	11k	12/11/85	[No Description Available]
BOXES	2k	2/06/86	[No Description Available]
BOXLOSS.BAS	3k	9/08/87	BASIC program to calc. ported box losses
CABINETS.ZIP	92k	12/08/86	put together your own speaker cabinets.
CBS CD.ZIP	81k	10/25/87	CBS CD catalog
CCALC.EXE	53k	10/05/87	AUDIO- RPN calculator for msdos
CC LINE.ZIP	2k	4/15/90	Measurements of 831709 220 mm woofer (Peerless)
CE-AMP.ZIP	28k	3/07/87	DESIGN COMMON EMITR XSTR AMPS - No Docs ! sorry
CE-AMP2.ZIP	43k	5/08/87	Improved Xsistr Amp Design Prog with docs, -bugs
CEILING3.ZIP	56k	10/24/89	Distributed Spkr Layout with graphics.
CLASS-A.TXT	3k	2/16/88	text on class A and class B amps
CLOCK.BAS	1k	1/09/86	[No Description Available]
COIL.ZIP	2k	10/04/88	Radio-Electronics Nov.88 inductor program
CONTENTS	2k	6/05/86	a discription of the files inside JBL.ARC
DACUPGRD.TXT	6k	6/07/89	Availability note on better 16bit DAC + filtr
DC2DRAW.ZIP	17k	7/31/89	Designcad (Prodesign II) speaker drawings
DIFF.ZIP	62k	5/04/89	PR/QR design aid for IBM w/ or w/o 8087 DIFFUSION
DRIVERS.MAD	14k	8/02/86	Listing of Madisound Drivers unsqueezed
DRIVERS.ZIP	4k	9/16/88	A listing of Madisound drivers w/ specs & price$
ECD.ZIP	27k	9/02/88	ELECTRONIC CIRCUIT DESIGNER SHAREWARE (BASIC)
ELECT2.ZIP	46k	9/11/87	electrical calculations toolbox
ELECTRC5.ZIP	65k	9/21/89	Electrical wiring calculations
EVFR.ZIP	8k	6/30/87	Electro-Voice FR System Data Files for CADP
EVHP2.ZIP	33k	4/28/87	Electro-Voice HP Horn Data Files for CADP
EVHR.ZIP	21k	12/10/84	Electro-Voice HR Horn Data Files for CADP
EVTL.ARC	0k	10/25/89	ms-dos
EVTL.ZIP	15k	7/08/87	Electro-Voice TL Bass Box Data Files for CADP
FFT26.ZIP	81k	9/18/90	FFT source code in C (Discrete Fourier Transform)
FILTBOX.ZIP	4k	9/21/89	Simple vented box and xover design tools
FILTER.EXE	226k	9/16/89	active filter v2.01 major update
FILTER.LZH	224k	6/16/90	Build electronic Hi/Low-pass filters
FILTER.ZIP	26k	12/30/86	MS/DOS Active filter design w/shematics
HELM201.ZIP	22k	7/23/86	calculate helmholtz resonator dimensions
HELMHZ.ZIP	9k	9/22/89	spreadsheet helmholtz reasonator calculator w/doc
HIAUD01.ZIP	14k	2/07/90	Hi End Audio discussions on UseNet
HIAUD02.COM	58k	4/26/90	Rec.hi-end audio discussion,self extract
HIAUD03.COM	14k	5/29/90	Hi-end audio discussion, self extracting Lharc
HIAUD04.EXE	13k	9/17/90	more Hi-End audio talk
HIFIBUZZ.BAS	3k	8/05/85	AUDIO- generates random hi fi buzzword combination
HORN.ZIP	11k	4/25/90	Horn enclosure design
ICBOOK.ZIP	38k	9/01/88	ELECTRONIC SYMBOLS
JBL.ZIP	102k	6/06/86	compressed file containing all files in CONTENTS
JBLSYSTM.BAS	2k	2/06/86	[No Description Available]
KNEWBOX.LBR	51k	12/28/86	NEWBOX re-written for KAYPRO with epson printer
LEAPDEM2.ZIP	168k	4/03/88	NEW VER L'SPK DESIGN PRGM. MATH COPROC. NOT RE'QD.
LFDEMO.ZIP	112k	11/20/90	LowFrequencyDesigner 1.0 demo EGA/VGA
LISTBOX.BAS	4k	8/19/85	[No Description Available]
LNAP.ZIP	161k	9/22/90	Linear Circuit Analysis Program Menu Driven
LOGSIM.ZIP	38k	2/01/87	digital ttl circuit simulation MS DOS
LOTUSBOX.WKS	16k	8/13/86	Boxresponse in Lotus with graphing functions.
LOUDSP21.ZIP	52k	11/26/88	Ver 2.10 of loudsp enclosure program- lots new!
LOUDSPKR.ZIP	32k	10/24/88	design aid for ported or closed box system
LPAD.ZIP	16k	6/01/87	X-over attenuator design - color
LSDP.ZIP	8k	11/21/87	LOUDSPEAKER DESIGN SOURCE CODE IN C. UNIVERSAL
LTC PS10.ZIP	350k	12/15/89	LTC's PSPICE uploaded by Walt Jung
LTC_READ.COM	4k	12/21/89	Info on LTC_PS10.zip
MF10.ZIP	27k	3/23/87	[No Description Available]
MLSSA.AD	1k	1/24/88	advert for nice audio test program and card
MLSSA.ZIP	247k	12/07/89	Working demo of MLSSA audio analysis package
MOTDSCV2.EXE	289k	4/20/90	Database for Motorola Discretes. Ver.2
MOT_ICV1.EXE	192k	4/20/90	Database for Motorola I.C.'s. Ver. 1
MPAPERS.ZIP	81k	9/10/86	Mike Piede's mods to Peter Schuck xover opt.
NCLOSURE	6k	2/06/86	[No Description Available]
NEWBOX.BAS	20k	2/06/86	graphic program from JBL people for box responce
NEWLTSBX.ZIP	7k	10/12/89	CORRECTIUON TO LOTUSBOX
NEWSPEAK.ZIP	46k	11/02/89	UPDATED SPEAKER.ZIP FROM AUTHOR - MS/DOS.
NEWXOVER.ZIP	65k	5/29/89	Flexible shareware passive xover program
NOTCH1.ZIP	2k	10/02/89	Simple notch filter design program
NOTE	1k	2/06/86	[No Description Available]
NUTMEGAT.ZIP	120k	12/08/87	updated Graphic postprocessor for SPICE AT w/87,CG
NUT_XT.ZIP	122k	1/03/88	updated XT version of NUTMEG graphic postprocessor
OHMS.ZIP	27k	12/29/86	MS DOS - OHMS LAW CALCS & OTHEELECT FORMULAE
PADS	3k	11/07/85	[No Description Available]
PAPERS.BAS	21k	8/13/86	Peter Shuck's network opt. prgm. needs input rtn.
PAPERS.ZIP	30k	4/30/86	Peter Schuck's network optimizer w/8087 exe
PAPERS1A.ZIP	30k	9/09/86	corrections made to Schuck's net.opt. program
PARSE.ZIP	37k	11/27/87	MS DOS SPICE3a7 rawfile post-processor
PARSE11.ZIP	40k	11/30/87	MS-DOS update of PARSE.ARC
PCAPS	2k	10/08/85	[No Description Available]
PCB.ZIP	93k	4/10/89	Printed Circuit Board design prgm
PCROUTE2.ZIP	103k	4/17/90	Circuit board layout program
PERF450.ZIP	215k	12/26/89	Updated 12/89 version of PerfectBox Warren Merkel
PERFGRAF.ZIP	1k	11/02/88	EGA prtsc Util, Warren Merkel
PHOMAST2.ZIP	0k	5/29/90	[No Description Available]
PLL.ZIP	139k	10/24/89	Phase Locked Loop circuit design program
PLOTFUNC.BAS	2k	10/05/87	[No Description Available]
PMISPECS.ZIP	98k	4/22/89	Precision Monolthics databook for your PC
POLYBOX.ZIP	40k	3/11/86	Polydax|Audax basic program to choose drivers
PORTNOTE	3k	12/02/85	[No Description Available]
PORTS	4k	10/24/85	[No Description Available]
PRO-RESP.ZIP	36k	2/05/88	Speaker response program with source code
PSPICE2.ZIP	148k	12/30/86	part 2 of circuit program DEMO VERSION
PSPICE3.ZIP	62k	9/04/86	part 3 of 3 circuit program DEMO VERSION
PSPICE4.ZIP	300k	7/18/88	Update to school version of SPICE
PSPICTUT.ZIP	6k	8/18/87	Simple Turorial for passive networks w/ PSPICE
PYLESPEC.TXT	8k	12/24/86	param. for specific pyle speakers
Q	5k	4/09/86	[No Description Available]
Q&A	28k	6/03/86	[No Description Available]
RCFILTER.BAS	2k	8/05/85	[No Description Available]
REDROSE.ZIP	38k	1/29/88	strange digitized music/vocal
SCHEM.ZIP	23k	12/23/87	Schematic Master util
SCHEMA2.ZIP	166k	9/08/87	Schematic Capture demo from Omation IBM PC
SCHEMA2A.ZIP	148k	5/13/90	89' Update of Schematic CAD Demo EGA/VGA
SONGBS20.ZIP	49k	4/29/90	SongBase database for music collectors
SOUNSYS.ZIP	53k	8/07/85	Basic Sound System (PA) aids (ala Davis SSE text
SPEAKER.ZIP	45k	4/16/89	AN EGA DEMO CREATED WITH A NEW ANIMATION TOOL.
SPEAKERC.ZIP	42k	11/20/89	IBM CLONE ALIGNMENT PROGRAM CGA
SPEAKERH.ZIP	42k	11/20/89	IBM CLONE ALIGNMENT PROGRAM herc.
SPECSECS.ZIP	179k	11/10/87	motorola power transistor database
SPICEAT.ZIP	158k	12/08/87	updated SPICE full version for AT w/nn87
SPICEDOC.ZIP	29k	3/01/87	Berkeley SPICE3A.7 User's Guide
SPICEXT.ZIP	86k	1/03/88	updated XT version of SPICE
SPREV.ZIP	23k	8/21/90	stereophile review index
STNDRDS	15k	10/08/85	[No Description Available]
SUBWOOFE.ZIP	2k	10/27/89	Subwoofer design based on Aug. 1978, Audio.
SYMSPKR.ZIP	2k	11/01/89	Lotus Symmetrical load S=.7 spreadsheet
TAPEMKR.ZIP	54k	6/11/90	Cassette labels and database
TCXOVER.ZIP	33k	6/02/90	Terry Cejka's xover program
TLINE.ZIP	156k	11/19/86	MSDOS Transmission Line Driver program
TRANS.ZIP	33k	7/02/87	class a transistor amplifier modeling MS DOS
TRANSMAN.ZIP	179k	10/31/87	MOTOROLA TRANSISTOR XFER MANUAL
TRANSMOD.ZIP	36k	12/30/88	Transistor program
TS.BAS	1k	1/02/84	[No Description Available]
TVSAT.ZIP	13k	3/15/87	TVRO pointer
VARI.BAS	5k	7/21/88	calculate Variovents
VARIO.BAS	4k	10/12/88	compute variovent boxes
VBOXRES.BAS	4k	8/19/85	[No Description Available]
VTH.BAS	2k	7/12/87	Op amp parameter calc. (audio) ms-dos
WIRE.BAS	1k	10/05/87	[No Description Available]
WSTDGTV2.EXE	84k	4/20/90	Database for Western Digital I.C.'s. Ver.2
XOVER	7k	4/08/86	[No Description Available]
XOVRTOLS.ZIP	6k	10/02/89	Various Pgms. to design passive x-overs
XYFIT.ZIP	124k	5/28/90	Program to fit curves to data

Fast Reply #EF20

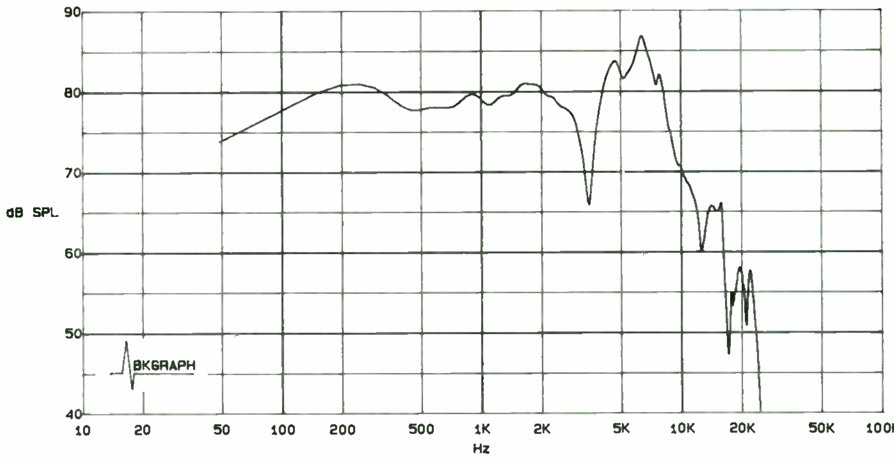

FIGURE 1: 5½" Taiwanese woofer/midrange 1m on-axis response (unsealed dust cap).

Continued from page 68

coil and gap so a jig can be inserted to align these components concentrically. Once the various adhesives have set sufficiently, he can remove the jig, leaving the voice coil and gap exposed to the elements. This poses a few significant problems, though. Dust and dirt can get into the gap, causing rubbing and distortion and perhaps premature failure of the driver. The clearances found in a typical 6½" driver are about the same size as a piece of sand. It is obvious, then, that some way of keeping foreign (or domestic) material out of the gap is necessary.

An acoustical issue exists as well. The exposed voice coil and gap provide, in a somewhat indirect fashion, a "leak" for the enclosure. This leak can often act as a bass reflex port, mistuning an otherwise carefully designed system. You need to seal this area somehow. I'll deal with other, more subtle acoustical issues a little later on.

The Essence of a Dust Cap

To solve these two basic problems, a small cover, usually dome-shaped, is glued over the voice coil. This is the dust cap. Its prime purpose, then, is to seal the driver and provide protection for the voice coil gap. Its effect, however, is far reaching.

Let's look at one problem the dust cap creates. When in place, the dust cap creates a small roughly cylindrical "enclosure," formed by the voice coil former, with the pole piece at one end and the dust cap at the other. When the driver operates, the air in this tiny enclosure is compressed as the cone moves inward, and rarified as it moves outward. In a woofer, it's not unusual for the cone to move ¼" peak-to-peak at low frequencies. The distance between the pole piece and the dust cap might only be ¾" or so. Under these conditions, the volume of this enclosure is changing by 33%!

When a volume of air is compressed by this amount, it responds by pushing back with significant and quite nonlinear restoring forces. But, since this chamber is not air-tight, there is a leakage path right through the gap. Unfortunately, this leak is quite small compared to the amount of air forced through it. The air often literally "whistles" through this gap, generating a lot of distortion. At best, you will hear a

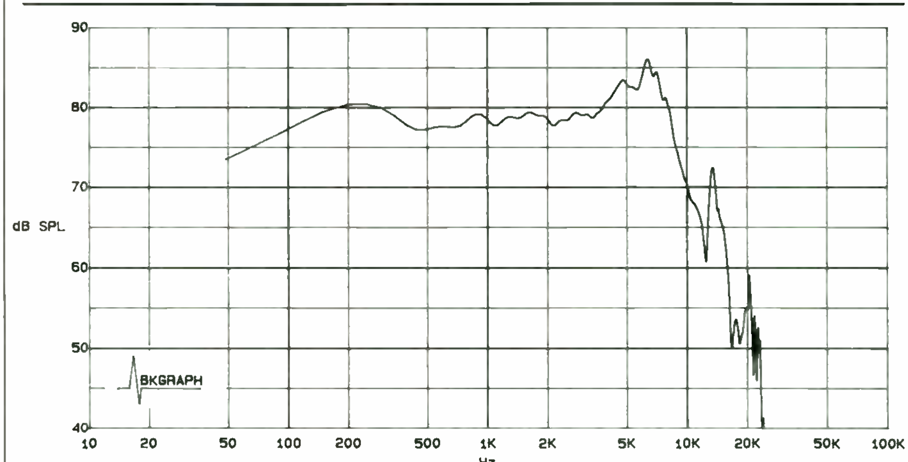

FIGURE 2: 5½" Taiwanese woofer/midrange 1m on-axis response (sealed dust cap).

loud "chugging" as the air is pumped through it.

You can relieve the pressure between the pole piece and the dust gap in several ways. The most popular method in medium and higher quality drivers is to punch holes in the voice coil former. Usually, a half dozen or so holes of the right size are enough to alleviate the problem. However, this method can weaken the former.

The best and often most expensive way is to have a hole of sufficient size bored right through the pole piece and out the rear of the magnet structure. If the magnet is designed properly (no easy task itself), the hole will not compromise the magnet, and it could even improve its performance. The air pressure will be relieved and the rigidity of the voice coil won't be compromised in any fashion. Using this method or the one described above, the pressure is relieved into the enclosure, in step with the cone itself.

The cheapest and worst possible way is to make the dust cap porous or leaky. This can be done by using an unsealed cloth dust cap, as Mr. Miller describes in his letter, by an open, porous paper cap, or by punching a hole in a solid cap. Indeed, the pressure is relieved but think about how it is. As the cone (and dust cap) move inward, the air is compressed in the enclosure in front of the pole piece and then forced outwards, contrary to the motion of the cone. Further exacerbating this, the air in the enclosure itself is being compressed by the cone as a whole, and is forced out through the voice coil gap.

In the Real World

So what are the practical consequences of this? Let's look at some real-world examples. I recently was asked by a client to try to figure out why one of his models sounded strange. I initially measured the on-axis response and found a 15dB hole at about 3.5kHz. This looked like a classic problem of cancellation between the woofer and the tweeter at the crossover (which was supposed to be at about 4kHz, close enough). I reversed the phase of the tweeter and the problem remained. Mystified, I measured the response of each driver. The tweeter measured as I suspected, but the woofer showed the same deep hole in the response (*Fig. 1*). When I tried to measure the Thiele/Small parameters of the driver by measuring its resonance in a sealed box, I came up with only very approximate results.

There, clearly displayed, was the hole I was measuring in the complete system. Closer inspection of the driver revealed it had a fairly open weave linen (or gauze) dust cap, like the one Mr. Miller describes in his Tandbergs. It was obvious from the above measurements (and others) the type of dust cap was contributing to the problem. I took a small bead of silicon rubber caulking, weighing no more than a tenth of a gram, and evenly coated the dust cap, sealing it completely (the secret here is that I used only enough to seal it, any more would have added enough mass to alter the driver significantly in other ways).

Remeasuring the driver showed a change for the better (*Fig. 2*). To confirm my suspicions further, I carefully removed the dust cap completely and the results were identical to the untreated sample.

What's Going on Here?

The air in front of the pole piece is being forced out by the inward motion of the cone itself (and forced inward by the outward motion of the cone). Also, air from the chamber behind the voice coil is being affected in a similar fashion. The result is that at about 3.5kHz, where these effects happen to be exactly out of phase with the air being moved by the cone itself, a cancellation occurs, leading to the deep hole in the response.

I've heard it suggested by others that these two chambers and the voice coil gap itself can act as a doubly tuned mini-reflex system, although I suspect the frictional losses of the air moving through the gap probably minimize this effect. However, it still could be a significant factor.

The point here is simple. The manufacturer of the driver, in an attempt to minimize the cost of the product, had skipped two reasonably well-proven means of solving one problem and ended up introducing another serious performance problem. The amount saved in using an unsealed dust cap in lieu of venting either the voice coil former or pole piece may have been as much as a nickel for a driver that costs the customer seven or eight dollars. The added cost to that customer to repair the problem, eventually having to find another, better source of drivers, is undoubtedly in the thousands.

Other Problems

A leaky dust cap has other adverse effects. I once had a fierce battle with an otherwise reasonable European driver manufacturer. I had specified their drivers to be used in several clients' systems. In one case, the system I had designed was a 6½" two-way, 16-liter vented-box system. The low end of this system performed admirably, being a QB3 alignment down 3dB at 53Hz. The drivers I had specified had sealed dust caps and vented voice coil formers.

The manufacturer, however, shipped them with sealed formers and porous dust caps. The resulting systems performed poorly at the low end, and no amount of redesign could make them approach the original specifications. The leaky dust cap provided a path for what was essentially a second port, mistuning the enclosure. No amount of physical evidence or theoretical discussion would move them to change their minds. They simply stated they felt it was not, in their minds, a problem. I suppose I could have convinced myself and my clients that it was not a problem, either. But I just couldn't convince the driver to change its mind.

The point to all this is that something as seemingly innocuous as a simple dust cap can have an enormous effect on the performance of a driver. I've given just one example of what can happen. I have not even touched on effects caused by differences in dust cap material, shape, size, and so forth.

More, then, to Mr. Miller's questions about his system. Indeed, a porous dust cap could lead to ambiguous resonance measurements, especially in an enclosure. Presuming the test setup is correct, however, you ought to be able to measure some, albeit vague, resonant peak in the impedance curve. Leaky dust caps would make the alignment of a system, at its best, chancy and difficult. You might try sealing the dust cap as I described above, but then how do you deal with relieving the chamber formed?

My recommendation is that if the enclosures are solidly constructed and are of suitable size, replace the woofers with modern components that do not have the same problems. Much has happened over the intervening years to improve the overall performance of drivers, and more knowledge exists about integrating drivers and enclosure.

Tools, Tips & Techniques

a la ART DECO

Here's a nifty construction technique that turns a plywood tube into a low-diffraction, Art Deco style cabinet.

I purchased a 5-foot length of a "half-round" from Long Island Laminates. This tube (6″ diameter, ⅝″ thick, and made of maple veneer) resembles plywood tubing cut in half lengthwise. By cutting two 10.5″ sections from it for the ends and adding mid-panels and top and bottom pieces (*Fig. 1*), I created a unique and handsome enclosure.

The mid-panels are ¾″ birch plywood, with edges recessed to attach to the half-rounds (*Fig. 2*). I glued the ends and mid-panels together, with a temporary spacer board in the middle and straps applying pressure on the outside (*Fig. 3*). I then glued on rectangular top and bottom pieces and made them flush with a router with a ½″ corner round bit by following the outside of the mid/end assembly.

I obtained the components (an AP-52 kit, which includes a Peerless 1592 woofer and an Audax DTW100T25 tweeter) from Just Speakers. I covered the inner surfaces of the cabinet with Acoustical Magic borosilicate ceramic coating and the exterior with a combination of equal parts of varnish and tung oil.

John Ott
Seattle, WA 98103

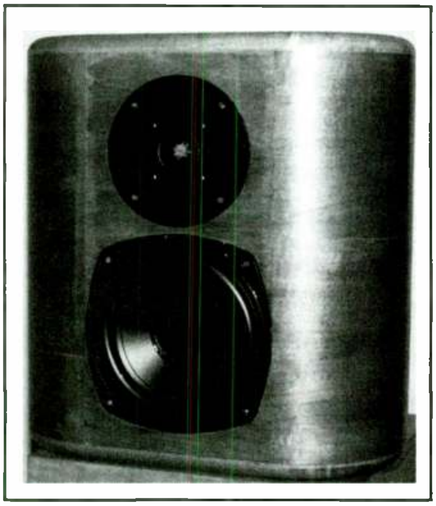

PHOTO 1: Cabinet created from a half-round.

SUPPLIERS

Acoustical Magic
1201 Jaynes Dr.
Grants Pass, OR 97527
(800) 654-4761

Just Speakers
3170 23rd St.
San Francisco, CA 94110
(415) 641-9228

Long Island Laminates
35 Engineers Rd.
Hauppauge, NY 11788
(800) 221-5454

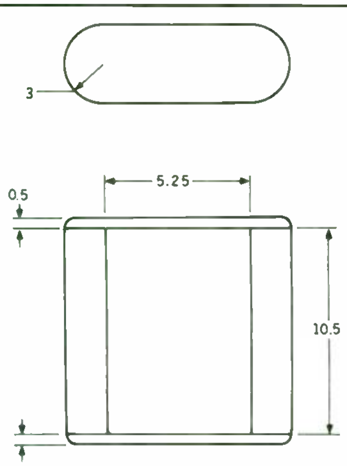

FIGURE 1: Dimensions of the enclosure.

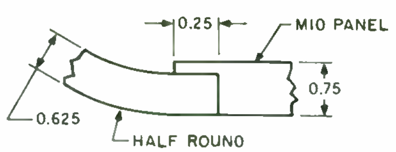

FIGURE 2: The mid-panels.

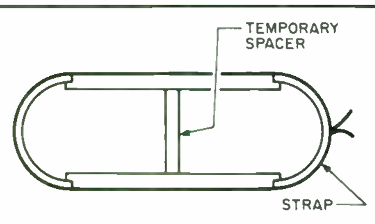

FIGURE 3: The ends and mid-panels glued together.

HANDY EQUATIONS

Here are some equations useful in designing vented box loudspeakers (*Table 1*). They are easy to memorize and allow for spur-of-the-moment designing and comparison of speaker drivers.

TABLE 1

VENTED-BOX LOUDSPEAKER EQUATIONS

QB_3	$\alpha = 50 \text{ Exp } (-9.5Q_{TS})$
	$h = 5.6 \text{ Exp } (-7.6Q_{TS}) + 0.75$
	$f_3/f_S = 6.1 \text{ Exp } (-4.5Q_{TS})$
SBB_4	$\alpha = 46 \text{ Exp } (-11Q_{TS}) + 0.85$
	$h = 1$
	$f_3/f_S = 14 \text{ Exp } (-8Q_{TS}) + 0.53$

Both sets of equations are based on a vented system with $Q_L = 7$ and are accurate to within 3% of the actual theoretical values (you can verify this with design tables) for Q_{TS} values of 0.21 to 0.39. The SBB_4 equations are also for lower Q_{TS} values. Note that Exp is the inverse natural log function. The equations are most accurate for Q_{TS} values around 0.30 to 0.36.

John Lipp
Houghton, MI 49931

SYSTEM TO GO

I combined several *Speaker Builder* projects and techniques with my own to create a gift for a friend. John Cockroft's small Isobarik (*SB* 3/85, p. 7) sits atop Richard Marsh's double-chamber woofer (*SB* 3/80, p. 7) which uses a 10″ Peerless driver (*Photo 2*). Four sides of the Isobarik's cabinet are a section of terra-cotta flue pipe cut with a ceramic circular saw blade *a la* Weems (*SB* 6/88, p. 37).

The front and back baffles are ¾″ particle board cut to fit and glued with Liquid Nails. The rough-surfaced and slightly irregular pipe is covered with two layers of black felt material glued with rubber cement. (I had previously used a section of round flue pipe, particle board, and black contact paper to build midrange enclo-

PHOTO 2: Isobarik atop double-chamber woofer.

sures for Peerless TP-165Rs. With Heil tweeters, these at first were paired with the woofers. This tubular enclosure also served to test several other 6″ brass-mids.)

The woofer cabinet is covered on all sides with black contact paper. This gives it an elegant appearance and covers dings in the particle board better than paint. On such a large surface, apply it carefully. First, clean the surfaces with an automotive tac-rag. It works best if you attach the paper at one end of a surface and pull the backing out from under it without stretching it excessively.

When the paper is adhered, you can smooth bubbles and cut out trapped particles, and you can cut around ducts and speaker holes with an X-acto knife. One disadvantage of contact paper, though, is it is not durable. (It looks best on front and back baffles with the remaining sides wood-veneered.)

The finished system sat on sturdy black 12″ speaker stands (not shown) constructed from particle board and sand-filled plastic pipes. I gave the three-way system to friends in two stages—the woofers and speaker stands first (to help out a pair of early B&K satellites), then about two years later the Isobarik's bumped the German pair to the daughter's system and the "CockMarsh" combo was born.

Fed by mid-fi components, the speaker system still sounds quite good even though its owner uses an equalizer to roll off the woofer instead of the low-pass filter I provided. Well, for old friends, you must forgive minor sins.

Rodney M. Baird
San Francisco, CA 94122

CUT-AND-DRY OPERATIONS

I have never liked gluing crossover components to a piece of masonite and interconnecting them with bits of wire even though the results are not visible when the project is complete. Etching circuits boards, however, is too slow, messy, and expensive for a passive crossover. With creative rearranging of components, the etching process can be replaced with a few cutting operations. *Figure 4*, for example, shows a basic two-way second-order crossover.

First, lay out the crossover so it minimizes the crossings of electrical paths. Generally, this will be with the common (ground) path in the center. Next, identify the nodes in your network and draw lines separating them from each other. (Nodes are sections of a circuit electrically

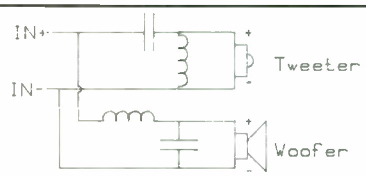

FIGURE 4: A basic two-way second-order crossover.

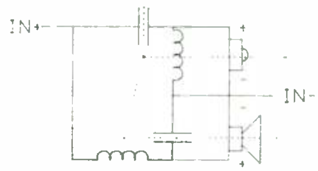

FIGURE 5: Layout that minimizes crossings of electrical paths.

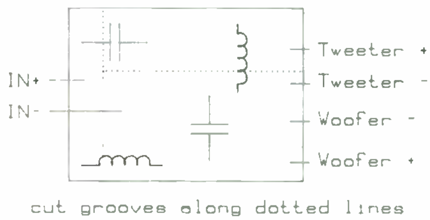

FIGURE 6: A crossover with three straight cuts.

equivalent—always at the same potential.) These lines should "cut" through every component in the crossover as shown in *Fig. 5*.

Transfer this layout to a piece of blank circuit board with copper clad on one side only (available at Radio Shack and other electronic supply stores). Make adjustments to account for board proportions and component sizes. All outputs do not have to be on the opposite side of the board from the inputs, and all inputs and outputs do not have to be at the edge of the board. Mark the component lead locations and draw the node separation lines on the copper side of the board. *Figure 6* shows how a crossover can be implemented with three straight cuts.

Drill the board for the component leads and cut the copper clad along the lines. I use a cylindrical burr in a Dremel tool held at a 45° angle to cut a v-groove through the copper. You should cut only as deep as necessary, but great skill or practice is not required. None of my boards have cracked, even when fitted with heavy ferrite-core inductors. Use an ohmmeter or continuity tester to ensure each node is isolated from the others.

Mount the components on the non-conductive side, passing the leads through the holes and soldering on the copper side. If you keep the leads short, resistors and most capacitors will need no other means of holding them to the board. For inductors and large capacitors, I recommend drilling holes in the board on either side of the component and using a nylon cable tie for mechanical stability.

Robert S. Green
Palatine, IL 60067

VENTING FRUSTRATIONS

Jim White has constructed a simple and effective spreadsheet version of the BOX-RESPONSE second- and fourth-order routines. LOTUSBOX.WKS, which you can download from the Madisound BBS at (608) 836-9473, is designed for Lotus 1-2-3, but you can use it with Microsoft Excel [*or Quattro Pro—Ed.*].

While the strengths of this spreadsheet include fast data entry and changes, easy storage of speaker parameters, and readable 1W and maximum SPL graphs, I have found it generates incorrect vent lengths. The speaker/box volume ratio (alpha) has been multiplied into the standard port calculation (as found in *The Loudspeaker Design Cookbook*, for instance), giving errors up to a factor of three. Removing the alpha factor (*I10) is easy, and the equation for cell I16 should be:

```
2124 * $D$19^2 / $D$13 / $D$11^2
    -0.732 * $D$19
```

I am confident Mr. White has a reason for using the alpha: perhaps he could enlighten us.

Stephen H. Shenefield
Cambridge, MA 02139

Technology Watch

SPEAKER CABINET

By Peter Muxlow

The latest thinking on the speaker box comes from B&W. Since wall vibration from the box walls can cause noticeable coloration of the original sound,[1] B&W has patented a unique way of bracing them.[2] To investigate its effectiveness, the company carried out the following tests.

First, they assembled a series of cabinets to test: a standard one of 12mm particle board walls with damping pads and bracing, an aerolam type (a space age material consisting of thin skins of aluminum separated by a 10mm aluminum honeycomb—Celestion makes a cabinet using this material), a concrete version, and the B&W patented product called the Matrix. They set up these inside an anechoic chamber (*Fig. 1*).

For the listening tests, they fed the input test signal (music and speech) to the cabinet. They electronically mixed microphone output of the anechoic chamber (the output of the test cabinet) with the input signal. Switching in and out and varying the amount of test-cabinet signal allowed the listening panel to study the effects of the enclosure coloration on the original signal. The use of headphones for comparison eliminated the loudspeaker and the listening environment.

The physical measurements came next. These included the amplitude of the cabinet wall vibration at chosen positions with sine wave and pink noise inputs, a modal analysis on the walls, and a Heyser energy-time test for the decay time of the wall vibration level when the loudspeaker is driven with an impulse input. According to B&W, these experiments confirmed the desirability of low-level vibration from the cabinet walls.[3] Equally important, they showed a rapid decay of vibration. B&W believes the Matrix met the above criteria better than the other enclosures tested.

In the B&W design (*Fig. 2*), the inside of the cabinet walls is grooved to accept a cellular matrix frame formed by perforated 3mm hardboard panels. These interlock and form cells of square cross-section as viewed from the front. The holes in the panels provide communication between adjacent cells, and all cells contain acoustically absorbent foam to reduce the resonance amplitude.

For construction, B&W preferred a polyvinyl acetate adhesive glue because it contributes to the overall damping by becoming rubbery rather than brittle when it sets. They say the cellular matrix frame increases cabinet wall rigidity and damping and has considerable cost advantages over aerolam and concrete.

REFERENCES

1. Tools, F.E. and S.E. Olive, "The Modification of Timbre by Resonances: Perception and Measurement, *JAES*, March 88, pp. 122–141.
2. AM patent 4690244.
3. Stevens, W.R., "Sound Radiated from LS Cabinets," Proc. AES 50th Conv. 1975.

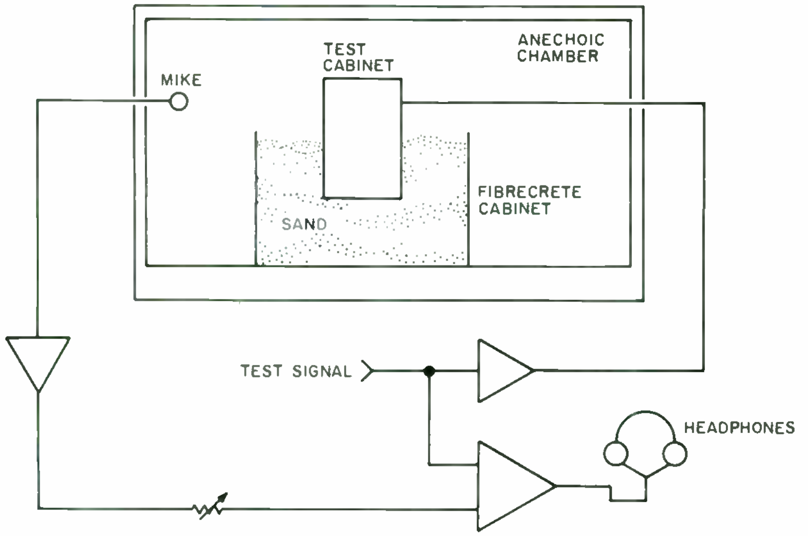

FIGURE 1: Diagram of the chamber used for testing the cabinets.

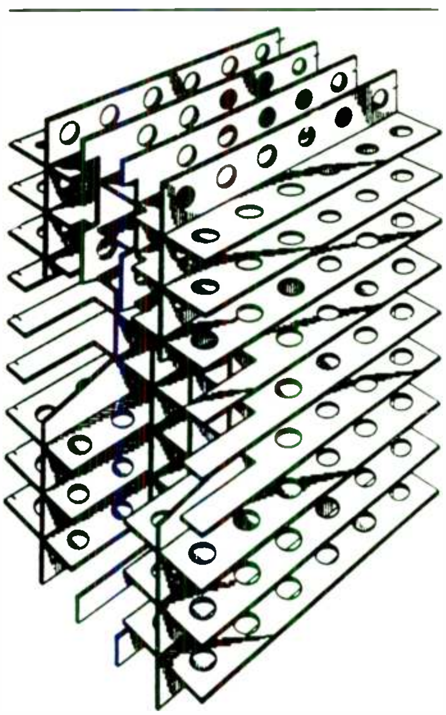

FIGURE 2: The B&W matrix frame.

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SPEAKER SAVER, FILTER

KH-2: SPEAKER SAVER AND OUTPUT FAULT DETECTOR. [3:77] This basic two-channel kit includes board and all board-mounted components for control circuitry and power supply. It features turn-on and -off protection and fast opto-coupler circuitry that prevents transients from damaging your system. The output fault detector has additional board-mounted components for speaker protection in case of amplifier failure. **$65**

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

AIDS & TEST EQUIPMENT

KK-3: THE WARBLER OSCILLATOR. [1:79] This unit will produce a swept signal covering any ⅓-octave between 16Hz and 20kHz. The total harmonic distortion at the output is less than 1.5%. The output voltage is adjustable from 0 to 1V. When used with a microphone it is as effective as a pink noise source in evaluating speaker system performance. It also reveals the listening environment's effect on sound through reflection and absorption. The sweep rate is set at about 5Hz. The kit includes 3¼" x 3⅜" circuit board, transformer, all parts and article reprint. **$70**

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. **$65**

KC-5: GLOECKLER 23-POSITION LEVEL CONTROL. [2:72] All metal film resistors, shorting rotary switch and two boards for a two-channel, 2dB per step attenuator. Choose 10k or 250kΩ. **$48**

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

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

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

SBK-E4: MULLER PINK NOISE GENERATOR. [SB 4:84] All parts, board, 1% MF resistors, capacitors, ICs, toggle switches included. No battery or enclosure. **$35**

CROSSOVERS

KC-4A: ELECTRONIC CROSSOVER, KIT A. [2:72] Single channel, two-way. All parts including C-4 board and LF351 ICs. Choose frequency of 60, 120, 240, 480, 960, 1920, 5k or 10k. KE-5 or KF-3 supplies are suitable. **$14**

KC-4B: ELECTRONIC CROSSOVER, KIT B. [2:72] Single channel, three-way. All parts including C-4 board & LF351 ICs. Choose two frequencies of 60, 120, 240, 480, 960, 1920, 5k or 10k. **$18**

KK-6L: WALDRON TUBE CROSSOVER LOW PASS. Single channel, 18dB/octave, Butterworth [3:79], includes three-gang pot. Choose 1: 19-210; 43-465; 88-960; 190-2100; 430-4650; 880-9600; 1900-21,000 hertz. **$60**

KK-6H: WALDRON TUBE CROSSOVER HIGH PASS: Single channel, 18dB/octave, Butterworth [3:79], includes three-gang pot. Please specify 1 of the frequencies in KK-6L. No other can be supplied. **$62**

KK-7: WALDRON TUBE CROSSOVER POWER SUPPLY. [3:79] Includes board, transformer, fuse, semiconductors, line cord, capacitors to power four tube crossover boards (8 tubes), 1 stereo bi-amped circuit. **$110**

SBK-A1: LINKWITZ CROSSOVER/FILTER. [SB 4:80] Th...-way crossover/filter/delay. 24dB/octave at 100Hz and 1.5kHz and 12dB/... 20Hz, with delayed woofer turn-on. Use the Sulzer supply KL-4A with KL-... **SOLD OUT**

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SBK-C1A: ELECTRONIC TWO-WAY CROSSOVER. [SB 3:82] 30Hz filter with WJ-3 board & 4136 IC adapted as one channel crossover. Can be 6, 12 or 18dB/octave. Choose frequency of 60, 120, 250, 500, 1k, 2k, 5k or 10k. The KL-4A/KL-4B or KW-3 are suitable supplies. **$32**

SBK-C1B: THREE-WAY, SINGLE CHANNEL CROSSOVER. [SB 3:82] Contains 2 each SBK-C1A. Choose high & low frequency. **$60**

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KW-3: BORBELY IMPROVED POWER SUPPLY. [1:87] This single channel, low impedance supply was designed for the exacting requirements of Erno Borbely's moving-coil preamp [2:86, 1:87]. The design utilizes polypropylene caps and 1% metal film resistors. LM317/337s are used in the preregulator and Signetics NE5534 in the op amp regulator. The kit includes a low profile 24V toroidal transformer, 4¼" × 5½" circuit board and all board-mounted components. Chassis and heatsink are not included. **$135**   Two or more, **$128**

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

KF-3: GATELY REGULATED SUPPLY. ± 18V or ± 15V @ 100mA. **$52**

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

KL-4B: SULZER DC RAW SUPPLY. ± 20V @ 300mA. **$60**

KL-4C: SULZER DC SUPPLY w/ toroidal transformer. **$85**

KH-8: MORREY SUPER BUFFER. [4:77] All parts, 1% metal film resistors, NE531 ICs, and PC board for two-channel output buffer. **$22**

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

KL-2: WHITE DYNAMIC RANGE & CLIPPING INDICATOR. [1:80] One channel, including board, with 12 indicators for preamp or crossover output indicators. Requires ± 15V power supply @ 63 mils.

Single channel **$58**   Two channels **$110**   Four channels **$198**

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

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

KX-1A: DISC STABILIZER. Set of 3 Sorbothane feet, 3 Tiptoes and Mod Squad's Disc Damper with 15 centering rings. **$70**

KY-1: BEERS' BUDGET CD MOD. [1:89] Kit provides POOGE-4 improvements without additional wiring or circuit boards. Complete parts for assembling amplifier modules and replacing DAC components. Article reprint included. Soldering skills required: not recommended for beginners. **$95**

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

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MOTIONAL FEEDBACK CORRECTIONS

On re-reading my article "An Acceleration Feedback System" (*SB* 1/90, p. 10), I discovered two errors. First, the formula for calculation Q° should read:

$$Q^{\circ} = \frac{R1}{2 \times R1 + R2} \times \sqrt{C1 / C2}$$

The formula on the disk is correct.

Second, the phase selector in *Fig.* 10 has been deprived of one important connection. *Figure 1* shows this corrected.

Since writing the article, I have become aware of "Motional Feedback with Loudspeakers" by J.A. Klaassen and S.H. de Konig (*Philips Technical Review*, Vol. 29, 1968, No. 5). This paper is a thorough treatment of the subject and includes practical details about the Philips drivers mentioned in my article. Among the more interesting points is the suggestion that the piezo-accelerometer should be mounted in an airtight enclosure to prevent variation in air pressure from producing a signal. If this housing is made of metal, it can also provide electrical screening. I have tried this with one of my prototypes, but so far I am not sure it makes any difference. But it does no harm, either.

Hans Mortensen
Glumso, Denmark

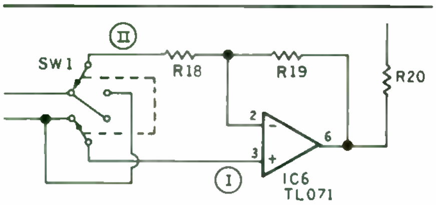

FIGURE 1. The corrected phase selector of the acceleration feedback system.

THE CALDWELL CROSSOVER

The schematic of Jack Caldwell's crossover which appeared with the review of Audio Concepts' Sapphire II speakers (*SB* 6/90, p. 56) was printed in error. We remind readers that the actual values of the parts and the specific brands chosen are vital to the correct performance of the speaker system. The assembled crossover and other parts for the system are available at reasonable cost from Audio Concepts. The design itself is proprietary to Jack Caldwell and any commercial use is prohibited without a license from the designer. We apologize to Mr. Caldwell and regret the error.

BRADLEY CORRECTIONS

In my response to Frank Anzalone (*SB* 5/90, p. 78), I spotted a typo that may prove confusing. The end of the published version of the letter substituted 4kΩ for 4kHz, impedance for frequency, which is an incorrect value.

Also, I may have reversed the order of the 2.0µF/10mH combination with the 6.3µF/0.25mH. I'm not sure if the order in crossover (B) should be that of (C) and vice versa. According to my old notes, it is correct as is. To be sure which was which, I would need to go back and measure the two variations. The key point is that one of them produces a smooth curve above 10kHz like curve C and the other has the nasty impedance peak like that of curve B.

Since I wrote that letter, I have acquired some gear that permits me to measure frequency response. The frequency response curve was measured outdoors, 6′ in the air and 1′ from the tweeter, on tweeter axis. Other than an anomaly at 8kHz, the speaker measures with crossover (C) +0, −3dB from 80Hz to 20kHz. The anomaly is "real" in that it does not show up in other speakers tested. Its cause has not been identified.

The falloff of measured response below 100Hz is typical of all speakers measured with this setup and does not represent actual frequency response. The "bump" at 25Hz is the action of the passive radiator.

Randall Bradley
Hannacroix, NY 12087

GREMLINS II

The gremlins attacked my correction in *SB* 5/90 (p. 3) and substituted 1mΩ (1 milliohm) for 1MΩ (1 megohm).

James Lin
Galveston, TX 77551

MIXED-UP REFERENCES

A number of errors appeared in my article, "A Triamplified Modular System" (*SB* 5/90, p. 46). Most are obvious, but I would like to note the most significant.

The first sentence in the second column on page 46 should have read "and *retuned* the bass enclosures accordingly." The fourth sentence in the first column on page 48 refers to *Figs. 1* and *2*, but should indicate only Fig. 1. The references to *Figs. 3* and *4* in the first sentence in the second column on page 48 should say *Figs. 2* and *3*. The fifth sentence in the second column on page 52 should identify

Fig. 4, not *Fig. 7*. The first sentence in the third column on page 53 should specify *Figs. 4* and *5*, not *Figs. 5* and *6*.

Randy Parker
Lancaster, PA 17601

WOOFER MINITEST

Some time ago, Larry Hitch at Madisound Speaker Components (8608 University Green, Box 4283, Madison, WI 53711, 608-831-3433) sent me three pairs of their dual voice-coil woofers. I have not had enough time to submit a full report, but I wish to make readers aware of these fine drivers.

The woofers sent to me were the 10204, 12204, and 12524—10″, 12″, and 15″ drivers, respectively. All have polypropylene cones with foam surrounds and feature dual 4Ω voice coils rated at 100W per coil. These are well constructed drivers and are suitable for use in sealed or vented boxes. Madisound will supply alignment data on request. Complete Thiele/Small parameters are also available, for those who wish to perform their own calculations.

As usual, Madisound is offering these drivers at bargain prices. They are currently priced at $45 each for the 10″, $49 each for the 12″, and $66 each for the 15″. Speaker builders needing dual voice-coil woofers should give these serious consideration.

Last summer, I helped a friend build a pair of Madisound's Sledgling kits. These are small bookshelf systems using Madisound's 6102 6½″ woofer and a Vifa D20TD-05 ¾″ cloth dome tweeter. The high quality drivers are matched by the Sidewinder coils and Chateauroux polypropylene capacitors used in the first-order crossover. It is quite refreshing to find such high-quality parts in a loudspeaker kit selling for $125/pair, including the enclosure. The quality of parts in the Boston Acoustics A40, at $190/pair assembled, does not come close.

With a low-frequency cutoff of 70Hz, the Sledglings certainly are not going to rattle floors, but within their operating range, they deliver clean sound with a creditable stereo image. Assembly is easy and the finished product is attractive. In any situation where size and price are the main considerations, the Sledglings are an excellent choice. You simply cannot go wrong for the price. Madisound's level of customer service remains extremely high.

Gary A. Galo
Contributing Editor
Potsdam, NY 13676

EQUATION FIX

In *SB* 2/90, Carl E. Richard wrote a short article on the basic Helmholtz vent length formula (p. 26). On page 76, the third equation for finding V_B contains a typographical error. The equation should read:

$$V_B = \frac{1.463 \times 10^7 \times r^2}{F_B^2 \times (L_V + 1.463r)}$$

Nicholas K. Soudas II
Jacksonville, FL 32257

BASS CORRECTION DEVICE

I am interested in electronic bass equalization circuitry, such as that described in Hans Mortensen's article (*SB* 1/90, p. 10). Regarding the circuit in Fig. 1, I'd like to point out that a related (though more general) single amplifier, biquadratic filter was described some time ago by T. Hamilton and A. Sedra ("A Single Amplifier Biquadratic Active Filter," *Proceedings of the International Symposium of Circuits and Systems*, April 1972, pp. 355–359).

Marchand Electronics (1334 Robin Hood Lane, Webster, NY 14580) and I recently developed a bass correction device based on a four-amplifier, biquadratic filter. In addition to improved S/N performance and less sensitivity to loading, the four-amplifier configuration allows adjustable tuning of the filter to suit a range of loudspeakers and listening conditions. The design is published in the April 1990 issue of *Audio*. Incidentally, high frequencies pass through only one of the four amplifiers in this filter.

While it is true that increasing cone excursion at low frequencies increases nonlinear distortion (such as harmonic distortion), there are other factors to consider when assessing bass equalization. First, as Mr. Mortensen noted, much music (with the notable exception of that intended for demonstration purposes) has little energy in the range below 50Hz. Hence, many music lovers will not notice increased nonlinear distortion levels, but will only notice the enhanced "openness" that results from electronically moving the bass resonance to a frequency below that of musical fundamentals or from reducing its Q. Second, if a speaker is designed with electronic equalization in mind, it is possible to reduce nonlinear distortion. That is, if you put a woofer in a small sealed enclosure and equalize the bass, you will get less harmonic distortion than the same woofer in a large enclosure, due to the linearity advantages of the acoustic suspension versus the mechanical suspension.

In any case, Mr. Mortensen's article was timely. CD is placing a greater emphasis than before on the reproduction of low frequencies. Several manufacturers, notably KEF, Thiel, and Celestion, include line-level bass equalizers with some of their speakers. Others, as he mentioned, use motional feedback to help reproduce the low bass. Hobbyists should have access to these useful techniques as well.

Along the lines of bass performance, I was also pleasantly surprised by Glenn DeMichele's cylindrical guitar speaker article in the same issue. Stereo 18″ woofers! I found the design innovative and the writing amiable.

Ralph Gonzalez
Wilmington, DE 19803

Hans Mortensen replies:

Thank you very much for your letter and interest in my article. Unfortunately, I was not aware of the two systems you mention, but I have set my librarian to work on it. When I planned the article, I had the vain hope of mentioning or describing more or less all the available systems, but soon realized that much more work had been done than I expected.

I agree with your observation that putting a woofer in a small sealed enclosure will reduce nonlinear distortion. Nevertheless, it is my experience that an active feedback system gets rid of even

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more distortion—at least that is what happened in my case.

The circuit I describe is simple—not to say crude—but it works. There are, however, still quite a number of problems to deal with. You mentioned one of them, namely tuning the filter to suit various listening conditions. There is also the transient response of a feedback system.

1. Recently, I have changed the crossover networks in my systems: I replaced the third-order low-pass filter on the circuit board with a third-order high-pass filter to get rid of the VLF garbage. This is a very simple operation. Moreover, I added an active second-order state variable low-pass filter. This filter allows easy change of crossover frequency (a stereo potentiometer will do the job) and equally easy change of Q. With closed-box side-systems this should work well. If you want to rid the side-systems of some of the low register, which they cannot reproduce anyway, it is easy to include a first-order high-pass filter in the system. In my system the state variable filter is adjusted to a crossover frequency of approximately 70Hz, and a capacitor in series with the input to the side-system amplifier gives a rolloff of 6dB/octave from approximately 50Hz.

2. The free choice of Q in this setup also allows some control over the transient response, in that a different Q_S will give different step-responses, but still there are quite a lot of parameters to keep track of. It is not at all clear where the efforts should be concentrated, first of all because a "perfect" sub-system will no doubt introduce some time delay for which the side-systems must be compensated. Ideas are very welcome indeed.

Long after I completed the article, I happened to re-read an article by J.A. Klaassen and S.W. de Koning entitled, "Motional Feedback with Loudspeakers" (*Philips Technical Review*, Vol. 29), in which the authors outline some basic theory in connection with the Philips devices. They suggest the transducer should be enclosed in a light, airtight house to prevent variation in air pressure from influencing the output voltage of the Piezo disc. This sounds reasonable, so I tried it on one of my systems. I cut a piece of thin metal off a beer can and glued it to the dust cap covering the disc without touching it. This lid then was connected to ground. Listening tests have been so inconclusive that I haven't bothered to modify the other system.

DRAWING BOARDS?

During the listening tests of my new speakers (KEF B110B + Scanspeak 2010A), I decided to try assembling the front panel using hard rubber, as suggested in *SB* 1/88 (p. 16).

Well, to my ears, at least, the result was losing sound immediacy.

Maybe there was a reduction of colorations I cannot detect, but overall I consider the result negative. These speakers use a very elaborate third-order acoustic crossover, computer optimized, and it seems the loss of coupling between the front panel and the box was causing objectionable vibrations that would decrease coherence in the sound field.

Jorge Omar Oliveira
Campinas SP, Brazil

QUESTIONS ON FIDELITY

I enjoyed the article by Mill Johnson ("Titanium + TPX + Polypropylene = Fidelity," *SB* 3/90, p. 34). I have several questions regarding his plans. Do you have to bi-amplify these speakers? If not, what type of crossover would be appropriate? Also, what type of wiring (diagram) is used for the woofers in the Isobarik version? I plan to use these with my Rotel 100W per channel integrated amp. I am sure it has sufficient power to drive an arrangement such as this.

Willard P. Wilkinson
Mobile, AL 36695

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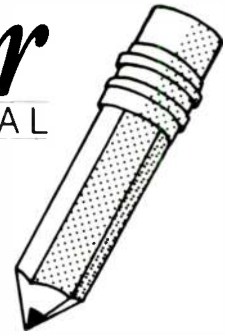

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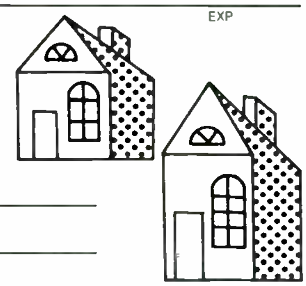

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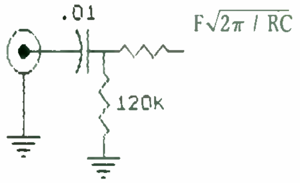

FIGURE 2: The 6dB attenuator.

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Mill Johnson replies:

I prefer bi-amp. I use a Richter Series III electronic crossover/equalizer set for 125Hz on the woofer amp only. The upper set of speakers are rolled off with a 0.01 capacitor to the input of the tube amps, which give a first-order 6dB attenuation (*Fig. 2*). Both speakers are wired in parallel.

IN STOCK, AVAILABLE

In the article "A Modular Three-Way Active Loudspeaker" by Fernando Ricart (*SB* 4/90, p. 36), Mr. Ricart mentioned he had not seen the Peerless 315SWR39 for sale.

I would like to inform *Speaker Builder* readers that it has been on the US market since October 1989 and is in stock and available from A & S Speakers, ITC Electronics, Just Speakers, Madisound, Solen Electronique, Speaker City, and The Speaker Works.

The part number is #831715, and is also known as the CC 12" woofer.

Rune Karsbaek
Racine, WI 53404

THE R-J ENCLOSURE

Although I am a charter subscriber of your superb magazine and I find something of interest in each issue, until now I have never written to you. The "Vintage Design" department reminded me of my interest in the R-J speaker design popular in the early 1950s. I don't recall any articles of the day giving a good explanation of its design features or an accurate description of its construction.

I do recall Edward Tatnall Canby's (ETC) descriptions of the R-J's remarkable performance in his column in *Audio Engineering*. This was the first speaker system that seemed to defy the then immutable bass performance/system volume law. Its major feature was a clever arrangement of a bass reflex port coincident with direct radiation from the speaker. This was done by mounting the single speaker on a sub-baffle spaced behind the enclosure's front panel. The bass reflex port consisted of a slot between this sub-

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baffle and the enclosure front panel. This design feature probably made the system as close to a Helmholtz resonator as is physically possible in a loudspeaker system.

ETC included enough verbal descriptions of the R-J's construction that I was able to build a crude approximation in my kitchen. Not having a real shop or any test equipment, I am afraid the result was quite bad. This was at a time when I made a new speaker enclosure every month or so and my long-suffering wife was convinced speaker enclosures were never finished as furniture.

The only R-J references I have found are from *Radiotron Designer's Handbook* (4th ed., 1952). Two mention Canby's columns, but two others looked promising:

1. Joseph, W. and F. Robbins, "The R-J Speaker Enclosure," *Audio Engineering* 35.12 (December 1951), p. 17.
2. Article giving detailed description of R-J loudspeaker to be published in *Audio Engineering*, January 1953.

I have been unable to find these articles, so my curiosity about the R-J has never been satisfied. Since it came and went so quickly, it must have had a terrible fault. I think it may have been a deep cancellation notch in the bass response, but I would like to know more.

Does anyone recall details of this landmark system or a related "EW" system? Do any R-J construction plans exist today?

Les Shaw
Boulder, CO 80303

WESTERN ELECTRIC's LEGENDARY 728B

I am writing to draw attention to my vintage Western Electric speaker, the 12" 728B. Its impedance curve (derived mounted in a reflex cabinet) is *very* different from that commonly found in speakers and it forms the basis for the T/S design theory. The curve (*Fig. 3*) shows a type of internal construction whose electrical equivalent circuit is far different from that basic to T/S theory.

Interesting features of the WE 728B are its 4" voice coil, its very shallow construction (only 3" deep overall), and a huge solidly cast magnet that hefts between 5 and 10 pounds. It also has a curvilinear cone.

The Western Electric 728B will interest readers because its performance (even 30 years after its purchase) surpasses that of many rated excellent today. It is a 12" speaker whose free-air resonance is about 60 cycles, advertised for a 3 cu. ft. enclosure and having a response from 60 to 10k cycles. Recent close-mike testing has verified its response is reasonably flat from 60 to 5k cycles, falling off 10dB at each end to 40 cycles and 10k cycles. All this in a 10 cu. ft. corner cabinet not yet tuned for an optimum, which cannot be calculated because of its unusual impedance curve.

This Western Electric unit reproduces voices realistically; it especially gives a most musical rendition of my best organ records (Virgil Fox, *Vale of Dreams*, Capitol SP8557). Its rendition of organs, even in an *untuned* 10 cu. ft. enclosure, is more *musically* opulent and satisfying than that of 5 cu. ft. Allen speaker whose cabinet was tuned to the design frequency of 33 cycles.

I conclude that tight bass is peculiar to the usual size of speakers, those of small volume operating with an α ratio greater than 2 (even my Allen, in which $\alpha = 2$, is not so lively as the WE), while the fully resonant musical bass appears only in large cabinets operating at α ratios less than 1.

An astonishing difference is in the curve's Q. My Allen Speaker Reflex has an impedance rise of 14x in its 5 cu. ft. cabinet, but this Western Electric has a rise of only 2½x in a 7 cu. ft. enclosure. I surmised each *very* low Q made the speaker performance independent of the box size, but my exact conclusions had to wait until I could determine the actual equivalent circuit.

Since the impedance curve made it impossible to determine the 720B's V_{AS} in a vented enclosure, I decided to install the speaker in a 3 cu. ft. sealed enclosure and determine the V_{AS} by its rise in resonance frequency. The result was baffling and unbelievable. This speaker *showed no resonance rise in frequency*; its f_R remained the *same* as when it was determined in free air.

Had I not used that same enclosure to test two other speakers and obtained the expected rise of f_R to almost double its free air value, I would have suspected the cabinet was not truly sealed. Also, my test data was done at enough frequencies to determine the entire Q curve, so I am confident the unexpected result is valid, baffling as it is.

Naturally, I wished to study the speaker's construction in more detail. The speaker easily separated into two sections: a light 2 lb. part that carried the cone and the attached voice coil and a heavy piece of the magnetic structure, *15 pounds* of magnetic material. I surmise the material may be Alnico D.

The speaker cone has a 4″ diameter hemisphere molded at its center, about whose periphery the voice coil winding support is cemented edgewise and is ½″ wide. The voice coil is wound on this 4″ dimension form, but the winding is only 3/32″ and its thickness is 0.025″.

The magnet has been finely ground over the entire front face and has a 4″ diameter annular slot into which the voice coil fits. The slot is a magnetic gap of only 0.065″ between centerpiece and periphery. This centerpiece has a 1½″ diameter and 2″ deep hole at its center.

Since the hemispherical dome of the voice coil makes an appreciable volume

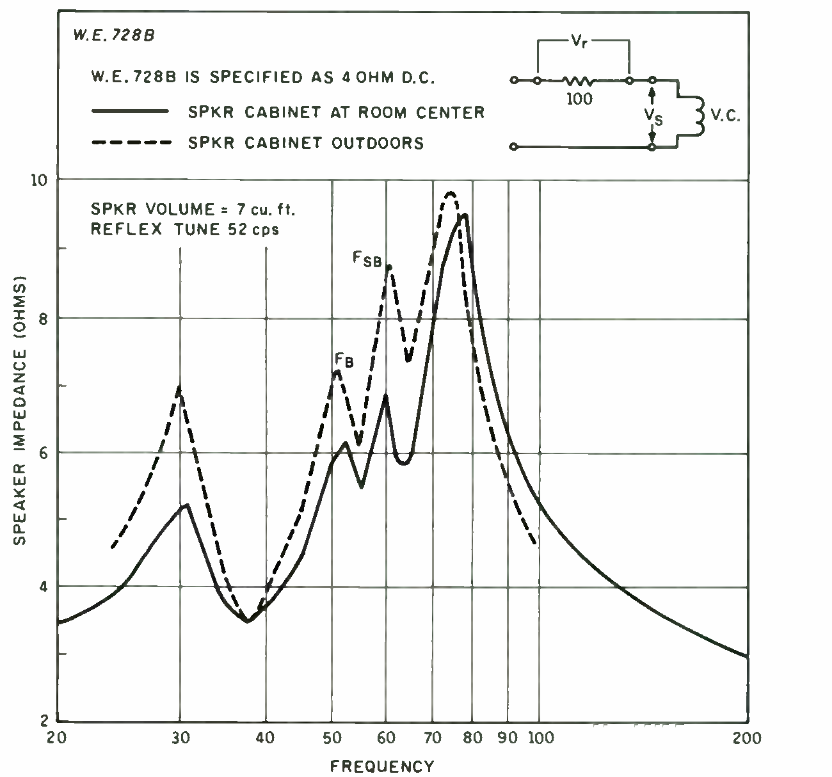

FIGURE 3: Impedance curve of the Western Electric 728B.

of air enclosure when the speaker is assembled, I wondered what this volume would be in the assembled speaker and what would include the magnet center hole. It measures to 12.88 cu. in.

Also, at the rear of the magnet hole is a shield with seven 3⁄16″ holes drilled in it serving to vent the inner chamber. I believe the actual cone-loading is by this inner-contained air volume and that *this* loading is so high that the exterior air-loading is negligible. Therefore, f_R is independent of the enclosure volume or its configuration.

From the measured impedance Q curve, I calculated that Q_{SA} = 2.33, Q_{EA} = 0.9725, and Q_T = 0.692. Also, close-mike testing shows this speaker has a resonance curve that well exceeds any speaker of today. It can be stated as *flat* from 60 to 4k cycles, ±3dB. Western Electric also made an 8″ version advertised with a flat response to 10k cycles.

I trust your staff can determine the design theory for this speaker and maybe can contact an AES old-timer who can tell more about it.

L.T. Medveson
Mentone, CA 92359

RE: DYNAUDIO D28

I have a question/comment to Mr. D'Appolito and any others who regard the Dynaudio D28 as something of a reference tweeter.

In the original 1984 design, as well as in the more recent Swan IV satellites, the Dynaudio D28 tweeter was used for power-handling capacity, easy frequency tailoring, high efficiency, and closer physical alignment of the voice coil with the woofer coils. What other tweeters were tried and had they been judged on the above criteria alone? Were listening tests of music conducted with only the tweeters (through a passive 2 or 3kHz high-pass filter, for example, or better yet, with an active filter at the same frequency)? This is an interesting test that, assuming you trust what you hear, reveals much about the basic sonic character of drivers.

I have used the D28 in many designs, including a copy of the original D'Appolito satellite. My experience suggests that although the D28 is good, it lacks the transparency, detail, and last octave "air" of smaller, lower-mass designs (like its stablemate the D21 or the Scanspeak 20mm units) and even with crossover tweaking to flatten the midband response, something a bit "horn-like" (in this case, not a positive trait) seems to remain.

One of my best-sounding projects was a modified D'Appolito satellite with the crossover changed for flat response with a Scanspeak 2008 and the crossover point shifted up closer to 3kHz to reduce the possibility of power overload. My listening panel generally agreed this was superior to the original, with a more subtle, delicate delineation of detail, even at high volume levels. At the time, I did not have the means to conduct high-resolution frequency response measurements, but the preference was consistent.

Perhaps I am doing something wrong or perhaps I do not have high enough volume requirements, but I believe the D28 is overrated; its basic sound quality does not seem very transparent. My guess is that all things being equal, a smaller diaphragm generally means lower mass and more even dispersion, both of which should lead to better sound. Comments please.

Can someone please explain in detail the theory of operation of a "push-pull" bass system? This method is used in a number of *SB* projects. (Levreault's "A Sixth-Order Vented Woofer System," *SB* 2/87, p. 9), but I have not seen a good exposition of how it works. Are the two woofers actually wired out of phase so they remain acoustically in phase with one of the woofers mounted backwards? Perhaps I have missed an article or letter somewhere.

Mike Chin
Vancouver, BC, Canada V5W 2V7

Joe D'Appolito replies:

Regarding the selection of tweeters, you must recognize that by the time any article gets into *SB*, the project is already at least two years old. All tweeters from my earlier satellite projects were tested exactly as you suggested. The requirement of a relatively low crossover frequency of 2kHz caused gross low-frequency distortion in all candidates except the Morel and the Dynaudio units. When the tweeters are listened to alone, this distortion is most obvious. You have three choices: raise the crossover frequency, increase the crossover slope, or use one of the two tweeters that can meet the requirements. The reasons for the choice I made were clearly explained in both my *SB* articles.

Today, few tweeters can deliver the SPL levels I designed for at 2kHz with acceptable out-of-band distortion levels. One fine tweeter that would most certainly be a candidate in any new design effort is the 29mm Scanspeak D2905. Of course, if you increase the crossover frequency to 3kHz or lower your SPL requirement, many more choices are open to you.

There is nothing magic or exotic about push-pull bass systems. To first-order, they are no different than any system using two woofers, with one important exception. If the suspension nonlinearities that produce distortion are identical in both woofers, then the even-order portion of the nonlinear responses that produce even-order harmonic distortion (second and fourth harmonics, and so on) will be out of phase and cancel acoustically. The voice coil of the backwards-mounted woofer is wired out of phase relative to the forward-mounted woofer voice coil so the fundamental it produces is in phase with the forward-mounted woofer.

TECHNICAL INQUIRY

In Jung's "LF Garbage Filter" article (*TAA* 4/75, p. 14), he shows in *Fig. 3* a general purpose VCVS filter circuit that can be used in several ways (Figs. 3b and 4a,b,c,d) simply by changing the elements in the filter network box (*Fig. 3*). This is nice except for the value of R6. Although not an element in the filter network box, its value appears to be a function of the desired filter characteristic. Furthermore, I do not understand the formulas for R6.

In Bullock's "Sixth-Order Alignments" article (*SB* 1/82, p. 20), R6 is again shown to be a function of the boost frequency, but in his article "Passive Crossover Networks" (*SB* 3/85, p. 14), R6 is not listed as a filter characteristic variable and C2L and R2H occupy circuit positions different from those shown in Jung's circuit.

I am in the process of building two sixth-order equalizers using the KF-6 kits and will soon be in a position to build active, two-way crossovers. As you can see from the foregoing, however, I need some technical guidance.

David J. Meraner
Scotia, NY 12302

Contributing Editor Robert M. Bullock replies:

The resistor R6 in my *SB* 1/82 article can actually have any value other than infinity, even R6 = 0, without affecting the filter shape. I believe I obtained the suggested value of R6 = RB from Jung's op amp book. As I recall, it is supposed to help reduce noise, although I do not remember for sure.

In my 3/85 article, R6 is not discussed at all, but the idea still applies. The way I remember, the general suggestion is as follows: for a high-pass circuit, take R6 to be the same value as the resistor from the positive input of the op amp to ground; for a low-pass circuit, take R6 as the sum of the resistors in the positive input path.

The apparent different placement of C2L and R2H in the 3/85 article as compared to C2 and R2 in the 1/82 article is explained as follows. R2H has not changed placement, but different pads are used in the drawings. There is more than one pad in this position to accommodate components of different sizes. As to C2L, remember it is in a low-pass circuit in the 3/85 article, while C2 in the 1/82 article is in a high-pass circuit. Consequently, you should not expect the same component types in corresponding positions. In general for this circuit, any filter-shaping resistor (capacitor) in a low-pass circuit becomes a filter-shaping capacitor (resistor) in a high-pass circuit.

3-D IN BASIC

I am working on a BASIC routine that helps calculate the formulas in Jean Margerand's article "The Third Dimension: Symmetrically Loaded" (*SB* 6/88, p. 29 and 1/89, p. 27) when used with the alignment table (6/88, *Table 3*). It will give the front, back chamber, and vent volume. Also, it allows for a different vent diameter input to get a practical length. Unfortunately, my math and computer skills

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are weak, so I didn't generate a routine for displacement and frequency response.

I wish to avoid vent noise. Is this possible with small vents and high SPLs? Is there some rule of thumb to have S_V large enough so this does not happen? Also, how is the power handling affected for cases where the back chamber volume is relatively large? Is this analogous to the sealed box α ratios?

The merit of the symmetrical-loaded enclosure seems high as it can allow simplification or deletion of a crossover. May I ask if you would contribute a BASIC program to cover the symmetrical load better? Also, I would be curious to see a model for the variant where the back chamber is vented.

Fred Ireson
Huntington, WV 25701

Contributing Editor D'Appolito replies:

Vent velocity limitations of a symmetrical bandpass subwoofer are the same as those of a conventional bass-reflex speaker. Small states that vent velocity should be no more than 4.5% of the speed of sound. This gives the following formula for the minimum acceptable vent area:

$$S_V > 0.0008 \times F_B \times S_D \times X_{MAX}$$

where:

S_V = vent area in cm²
F_B = front volume tuning in Hz
S_D = cone area in cm²
X_{MAX} = max linear throw of the driver in mm

Bandpass designs have a bigger problem than vent noise. The tuned front volume acts like a low-pass filter only at the lower frequencies where port and front volume dimensions are small compared to the wavelengths involved. At higher frequencies, the vent develops "organ pipe" resonances and front volume standing-wave resonances that pass directly through the vent. (This also happens in conventional bass-reflex systems.)

If these resonance effects are not controlled, bandpass subwoofer response at 500Hz to 1kHz is often equal to its low-frequency response. Even with proper design, the total high-frequency attenuation of the front volume acoustic low-pass filter rarely exceeds 20dB. Thus, some form of crossover is still advisable with bandpass designs.

The sixth-order symmetric bandpass design consists of a fourth-order system with a series LC circuit to provide high- and low-pass electrical rolloff and acoustic rolloff. The sixth-order system has many advantages over the fourth-order one; for the same bandwidth, it can be up to 6dB more efficient. Also, the LC circuit controls low-frequency out-of-band cone motion and attenuates unwanted high-frequency resonances. Finally, because the Q of the series LC circuit is easily controlled, it is usually possible to obtain flat response in the passband. This is in contrast to the fourth-order design where passband ripple is almost unavoidable.

I hope to publish an article on the sixth-order design in *Speaker Builder*. It will include design software, as the design process is too complex to be reduced to tables.

The doubly vented bandpass design produces an inherently nonsymmetric response. This greatly complicates the design process because no low-pass prototype exists on which to base the design. Higher-order doubly vented systems must be designed via computer iteration or optimization. The models for these systems are well-known, but are quite complex. I know of no simple design procedure and do not recommend their construction without some design software, as your chances of producing a poor design experimentally are excellent.

NO ISO-NULL?

In his article in the 5/90 issue of *Speaker Builder* (p. 10), Bill Schwefel describes an "Iso-Null." I take this to mean he expects a standing wave to develop between the two drivers in the constant pressure tunnel. The frequency of the standing wave and its corresponding peaks and nulls depends on the spacing between the two drivers.

I was under the impression that because the drivers move in unison, the tunnel be-

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tween them truly maintains a constant pressure (hence the name Isobarik). If the pressure is constant, then a standing wave can't be in the tunnel. Also, there should not be any such thing as an Iso-null or an Iso-peak and the length of the tunnel should not be critical. Might you say a few words about this?

Len Moskowitz
Teaneck, NJ 07666

Bill Schwefel replies:

Thank you for your interest in my Korean Wonder speakers. As you suggest, a standing wave does develop between the two drivers mounted in an Isobarik tunnel. The peaks and nulls will move around as you alter the spacing distance. The peak will depend on the high-frequency response of the driver and the null on the spacing distance.

The Iso-null (Iso-dip would be a better term) will be equal to half a wavelength of the spacing distance. In my speakers, this occurs at roughly 2.5kHz for a distance of 2¾" (13,560/2/2.75 = 2,465Hz).

The two drivers will tend to move in unison as you lower the frequency, but not like perfect dancing partners. Even as low as 100Hz, there will be some (not much) cancellation due to standing waves. These standing waves (please excuse the pun) will not simply "stand up" at certain frequencies. They will come and go gradually based on the distances involved.

You can observe this effect by generating a constant 1kHz sine wave through your speakers and then walking around the room. You will notice the 1kHz tone becomes louder and softer. Even slight head movements will make a large difference in apparent sound intensity. It does not suddenly jump high or low; it moves gradually based on relative distances, just like the Iso-null.

Try an experiment. Fasten two speakers together face to face and seal them with a gasket. First hook them in reverse phase so they play in unison. Then switch the leads on one speaker and play them in phase. You will notice the speakers tend to cancel each other's output and sound garbled. This garbled response is the inverse of the Iso-null.

The term "Isobarik" means constant pressure and is a somewhat accurate term for closely spaced woofers operating below 1kHz. When operating in this lower range, you do not need to consider the Iso-null because it normally falls beyond the operating range of the driver.

BEER CLARIFIER

In regards to Bill Schwefel's article "The Beer Budget Window Rattler" (*SB* 3/90, p. 10), how did he determined the upper crossover frequency for the midrange pair spaced 9" center-to-center?

The wavelength chart (*Table 6.2*) in Vance Dickason's *The Loudspeaker Design Cookbook* gives 9" = 1.5kHz as the highest frequency, not 2.7kHz.

I hope you can help clear this up for me. Thanks.

Steve Anderson
North Highlands, CA 95660

Bill Schwefel replies:

Thank you for your interest in my Beer Budget Window Rattler. Your letter raises an important point. *Table 6.2* refers to an asymmetric system with only one vertically aligned tweeter and mid/woofer. It does not apply to the center-to-center distance between midrange drivers mounted in a symmetrical 3/2 D'Appolito arrangement.

LETTER WRITERS AHOY . . .

We need your cooperation in the matter of your welcome letters to authors and other readers. Please enclose a stamped and addressed envelope if you expect a reply. If the author/reader lives outside the USA, please include two International Postal Reply coupons (available at your post office) instead of stamps on your envelope.

In questioning authors, please leave room in your letter for replies which should relate to the article, be framed clearly, and written legibly. Please do not ask for design advice or for equipment evaluations.

In *Speaker Builder* 4/84 (p. 7), Mr. D'Appolito cautions readers not to exceed a spacing distance of one full wavelength. Measure this distance from the center of the tweeter dome to the center of one midrange driver. On the Beer Budget system, this distance is 4½" and equates to 0.9 wavelength at 2.7kHz.

I would like to lower the crossover to 2kHz, but other things must be considered. This would increase the chance of tweeter failure. It also would move the upper crossover too close to the lower crossover point of 700Hz.

MODIFIED LISTENING ROOM

Ralph Gonzalez and I would like to thank *Speaker Builder* and Mr. Koonce for the comprehensive review of our program, "The Listening Room" (*SB* 6/90, p. 63). Since the article was written and because of customer requests, I have modified the program to remove all placement limitations. The speakers, as well as the listener, can now be moved to any point in the listening area. Also, we now supply a math coprocessor version for machines equipped with such and have customized the code for both versions to provide for faster plotting. A printed manual is being supplied with the program.

I am very responsive to user suggestions and often add a requested feature on the same day it was suggested. Indeed, I look forward to suggestions that will make the program more useful.

Bill Fitzpatrick
Sitting Duck Software
PO Box 130
Veneta, OR 97487

L-R Crossover

continued from page 35

higher order of filter, which makes the summed acoustic transient response of a step function appear far from ideal. Whether or not this is an audible effect is debatable. However, D'Appolito stated a flat on-axis frequency response is a primary consideration and it was my goal for this project.

Building Speakers

continued from page 18

ferent types of music when making the comparisons. *Table 1* shows the responses of some students in June 1990.

The excitement on the students' faces brings the whole course into focus. All the electrical and Thiele/Small enclosure theory, the mathematics, the homework, the schmutzing, the wiring and assembling, and the soldering and testing have produced a product the students will care about and use often.

Servo Subwoofers

continued from page 31

CONCLUSION. The two systems, using acceleration and velocity sensors, are servo subwoofers that should satisfy any critical home listener. They will handle music power in small living rooms (15 by 19 feet) with 100dB or more without strain. The distortion levels, staging, and transient handling are unmatched by any system to which I have listened in private homes or audio store listening rooms.

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WANTED

Magnum Dynalab or Yamaha CT-7000 FM tuners, must be in good condition and reasonable. Eduard, (213) 395-5196.

KLH, Advent or Proton FM table radios. Will consider inoperable radios. Michael Vahldieck, 12108 Elmhurst Pkwy., Milwaukee, WI 53226, (414) 475-5163 7-10 pm CST, or send a postcard and I will call you.

Electro-Voice components; (2) 16Ω T350 "Ultra-Sonax" drivers with mounting brackets, (1) 16Ω T35 "Super-Sonax" driver with mounting bracket, (1) 16Ω MT30 or 848HF driver, one 16Ω X336 crossover. David Hasek, 209 Eareckson Ln., Stevensville, MD 21666, (301) 643-1814, 7-9 p.m.

Wanted (cheap-for personal use): dbx 3BX (4BX?): Commodore 64 or 128, Atari, Apple II, MacIntosh computers. Angel Rivera, 1845-52nd St., #32, Brooklyn, NY 11204, (718) 837-3203.

Two JBL JET8A speakers in good condition. Buy or trade. Have a McMartin 100W commercial amp. N. Oshana, 101 Treble Rd., Bristol, CT 06010.

Cheap for personal use: dbx 3BX (4BX?); Commodore 64 or 128, Atari, Apple II, MacIntosh computers. Angel Rivera, 1845 52nd St., #32, Brooklyn, NY 11204, (718) 837-3203.

Help needed for information on modification of transmission line IMF (Fried) speakers. Contains (1) KEF B-139 woofer, (1) KEF B-110 midrange (1) 1" tweeter, (1) ¾" super tweeter. Stan Kaleta, 7658 North Linden Ln., Parma, OH 44130, (216) 845-5347, collect.

Old Klipsch catalogs, JBL literature, books; Bergh and Dean, *Light Emitting Diodes*; old books dealing with point-contact transistors; old Hewlett-Packard and Tektronix catalogs; *Throne of Merlin*, (R.C. Schaller). D.R. Schaller, 6704 Schroeder Rd., Ste 6, Madison, WI 53711.

FOR SALE

Antique buffs: 15" electromagnetic woofer Scott model 505 tube power supply frame marked Magnavox part 925 spec D-1021 1271 (2) KEF T-27 tweeters, (2) Cerwin Vega 12" cast frame 122W4 needs reconing, best offer or trade on all speakers. Merle Weddle, 1530 Celebrity Cir. W., Hanover Park, IL 60103, (708) 289-0794.

Raw drivers. Audio experimenter selling off assortment of custom and OEM speakers, ADS, AR, Audax, Bolivar, Cole, Isophone, JBL, Jordan Watts, Polk, Dyna, Foster, MB, Dalesford, KEF, etc., send SASE for free list. Joseph Waclaw 5224 West Nelson St., Chicago, IL 60641, (312) 777-0943.

Magnavox CDC552 carousel CD player, new Christmas 1990, with remote, $215 ppd.; Phase Linear 4000II preamp, $215; Souther SLA-3 linear tonearm, $175; ADC PSX-30 phono cartridge (retail $150), unopened, $35; Knight audio generator in custom rackmount, $35. Andy Sommersberger, 2506 So. 20th, Sheboygan, WI 53081, (414) 458-2057.

Nice walnut cabinets, 2 cubic feet, with cutouts for 8" speakers, black grille cloths, $100/pair; 15" Jensen Flexair woofer, $75; pair of Jordan 5" aluminum cone speakers, very rare, $300; Celestion HF 1300 pressure tweeters, $30/pair; MK II only one, $20; VIFA C17WH 6½" speakers, $30/pair. Jerry, (212) 515-4853.

Sony SL/TT 2000, $250; Parsec Beambooster, $40; new TDK P6-60MP 8mm tapes, $4.50; GE Control Center 4, $79.98; Recotron FREDII, $50; Nakamichi LA-100, $50; Sony AC-M110, $30; Blaupunkt Frankfort, $99; AudioSource LS-ONE, $89 pair, Sony DT-4 with acc., $165; Toshiba T-1000 laptop with additional 768K board, $650. Michael, (703) 641-5824.

Complete plans for folded horn, long path, compact cabinet, front and rear loaded new concept, mouth exhausts to front or floor, awesome bass, 103dB efficiency. Send $35 Canadian in postal money order. Jerry Moody, 128 Sherwood Dr., Brantford, ON, Canada, N3F 1N9, (519) 759-3826.

Ribbon drivers (4) 30″ long units similar to Strathearn of Gold Ribbon designs, made by Carver. Have crossover data, $100 per section. Kevin, (216) 941-6490, leave message before 5:00 EST or call evenings.

Pair Gauss 4060 2″ drivers, new voice coils, $700 or trade for pair JBL 2245. Doug Miller, (504) 367-5388, home, (504) 585-2410, work, 24 hours.

MIT PC-18 speaker cables with Esoteric Gold Bananas, one 12′ pair, $150; two 8′ pairs, $90 per pair; dbx 228, $50. Larry Cartwright, 2723 Darlington Rd., Beaver Falls, PA 15010, (412) 846-7419.

10 ga., multi-strand silver-plated wire, Teflon insulation, $1/ft; perforated aluminum stators for ESL per Hermeyer, *TAA* 2/77, holes line up for transparency, perfectly flat, 80 pieces 7″ X 18″, $150; heatsinks for stereo Pass-40 amplifier Wakefield 435-A, eight for, $80. Prices in US dollars only. Nick Mastrobuono, 50 Copperfield, Sarnia, ON N7S 5K8, (519) 336-9160.

Teflon sleeving, 0.028″ inside diameter (22 ga.), extra-thin wall (6 mil wall thickness) for insulating component leads and conductors. Clear, non-shrinking, high-temperature. Make your own custom Litz wire, 40 cents/foot plus $1 postage and handling. B. Hefner, PO Box 6601, Newport News, VA 23606.

Paradise VGA card, new, $100; Western Digital 20M hard card, $25; 287-6 coprocessor, $50; 1″ MDF 2′ X 5′ sheets, 5/$10 pickup only (central CT); used; Admiral dual channel pulse generator, $25; Sherwood-Draco micro/CPU 100 tuner, programmable call letters, $350; REGA II with Ortifon om30, $250. All negotiable. John, (203) 872-9210 EST.

Magnepan Tympani 1D, $500/pr; PS Audio IV-H preamp, $200; Robertson 4010 amp, $350; $500 for pair; Kenwood KT-8300 tuner, $100; Terk Pi FM antenna, $30. All like new with manuals and original boxes. Ed, (412) 351-1052.

Ampex tapes 7″ plastic reels 2400′ per reel, highest quality, data grade, exceptional for voice or music, eleven cases, 40 reels per case, sacrifice at $100 per case plus $5 shipping. Sy Roden, Route 5 Box 331-B, Hartselle, AL 35640, (205) 355-8251.

SEAS MP12VL poly midranges, (4), $60; Duntech electronic crossover with bass EQ, $200; Superphon DM-220 amp, $475; Superphon CD Maxx preamp, $250; woodstyle WS-602 (black), one only, $15; SEAS P17RCY 6½″ poly woofer, $15. Rick Craig, (704) 536-6821, 9-12 pm EST.

Crown IC-150A preamp, $195; JVC JT-V77 top-of-line tuner, $75. Both for $225. Darren, 6633 Glenway Dr., W. Bloomfield, MI 48322, (313) 851-1296.

Dynaco ST-410, weight 50 pounds, excellent operating condition, volume control kit added, brushed aluminum face panel, lettering came off, original manual, 300W/channel at 4Ω, nice workhorse amplifier, shipping negotiable, $225. Don Lockett, (602) 749-1527, Tucson, AZ.

Kinergetics KBA-75 Class A amp, excellent condition, $775; Carver ESC-U speaker controller, as new, $140; Noble stereo pot, new, $15; Monster Discust, CD damping disk, brass, $10. Doak, (504) 738-1235, 6-11 p.m. CST.

AR SP-11B II, $3,200; cj Premeire 5s, $3,000; SOTA Star Sapphire Black Lacquer, $1,200; Electronic Flywheel, $200; audio garage sale—from Litz wire to raw drivers send SASE. Records send SASE. Douglas S. Robinson, 104 Lincoln Ave., Waterloo, NE 68069, home (402) 779-2589, work 8-5, (402) 779-2531.

(2) Dynaudio D-52AF, new, $90; (2) Madisound 1054, new, $45; Fisher 500C, $50; dual 1019 changer, $20. All plus shipping. John Duncan, (208) 375-5292.

Altec A-7 VOT speakers; Altec 604-E speakers in studio rosewood formica cabinets with 411-8A subwoofers in matching cabinets; pair 415-A diacone 15″ woofers; pair ER-15; pair 421-II; Eumig rackmount preamp; Cronin DC-300A; VFX-2A; Phase Linear 400-II; 500-II; E-V 1823M horn drivers; Altec 511; 802, 808, 806 drivers. David, (914) 688-5024.

Toroidal power transformers: Avel-Lindberg 40-3021, dual 22V 2.8A secondaries, dual 115V primaries. Totally encased. Great for preamps or small power amps. Limited supply at $15 plus $3 shipping each. Jerry McNutt, 801 S. Gay St., Auburn, AL 36830, (205) 821-8426.

McIntosh MC-2205 power amplifier 200W/channel with cabinet, $900; NAD 2200PE power amp 100W/channel, $275. Roger Daniels, Rt. 5, Box 612, La Grange, TX 78945, (409) 247-4260 before 9 p.m. CST.

Highly modified Audio Concepts speakers, all Dynaudio, 21W54MPS, D52, D28, damped, diffractionless cabinet, poly crossover, Audioquest wire, etc. Must pick up western MA, northern CT area, $550; (413) 525-1678 after 5 p.m. EST.

10-foot TL subwoofers with 24W75 Dynaudio based on Roger Sanders' design, *SB* 3/90, but folded. Made of 1.5″ MDF. Very massive! Good workmanship ideal for electrostats or ribbons, $750/pr; DeCoursey crossover as per Roger Sanders' *SB* 3/90, 500Hz with 30Hz filter, $250; or both for $900. Will deliver in southern California. Eduard (213) 395-5196.

NAD 1130 preamp, like new in box, $140. Paul Rodzevik, 27 Heyward St., Mohegan Lake, NY 10547.

Rare and unusual KEF model K-2 Celeste MK II two-way speakers made in 1966 in England. Solid wood construction, never abused, first class condition, woofer is oval shape, truly collector's items. Good deal for the right person, cash price, $200/pr. I pay shipping. Ed, (305) 891-2267.

CLUBS

AUDIOPHILES IN THE DAYTON/SPRINGFIELD, OHIO AREA: We are forming an audio club. Please contact me if you're interested in construction, modifications, testing, recording or just plain listening to music. Ken Beers, 1756 Hilt Rd., Yellow Springs, OH 45387, (513) 767-1457.

THE CATSKILL AND ADIRONDACK AUDIO SOCIETY invites you to our informal monthly meeting. Join our friendly group of audio enthusiasts as we discuss life, the universe and everything. No matter what your level of interest, experience, or preferences, you are welcome. Meetings are generally held once a month, on a weekday evening. Contact CAAS at 756-9894 (leave message), or write CAAS, PO Box 144, Hannacroix, NY 12087. See you there!

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

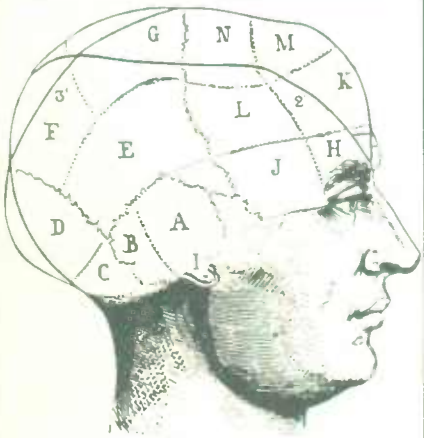
THE INLAND EMPIRE AUDIO SOCIETY our former name, has now been changed to the **SOUTHERN CALIFORNIA AUDIO SOCIETY** (SCAS). Our effort is now inviting music lovers, audiophiles, hobbyists and other interested parties throughout the southland to join us in our pursuit for that elusive sonic perfection and truth at our meetings and seminars and through our official speaker, *The Reference* newsletter. For information write or call: Frank Manrique, President, 1219 Fulbright Ave., Redlands, CA 92373, (714) 793-9209.

THOSE INTERESTED IN AUDIO and speaker building in the Knoxville-East Tennessee area please contact Bob Wright, 7344 Toxaway Dr., Knoxville, TN 37909-2452, (615) 691-1668 after 6 p.m.

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

ELECTROSTATIC LOUDSPEAKER USERS GROUP is now a world-wide network for those interested in sharing valuable theory, design, construction, and parts source information. If you are interested in building, or have built, your own SOTA ESL we invite you to join our loose-knit organization. For information, send an SASE to: Barry Waldron, 1847 Country Club Dr., Placerville, CA 95667.

THE HI-FI CLUB of Cape Town in South Africa sends a monthly newsletter to its members and world-wide subscribers. To receive an evaluation copy of our current newsletter, write to: PO Box 18262, Wynberg 7824, South Africa. We'll be very pleased to hear from you.



FILL IN THE BLANKS JOIN AN AUDIO CLUB

Learn about the latest equipment, techniques and recordings through group meetings, tours, and newsletters. Ask questions. Share viewpoints and experiences. Stretch your mind.

If there's no club in your area, why not start one? Our club ads are free up to 75 words ($.20 per word thereafter). Copy must be provided by a designated officer of the club or society who will keep it current.

PIEDMONT AUDIO SOCIETY. Starting an audio club in the Raleigh-Durham-Chapel Hill area of North Carolina. Interested in designing, building, and modifying speakers and electronics (solid state and tube). Beginners and old hands both welcome. Kevin Carter, 9009 Langwood Drive, Raleigh, NC 27612, (919) 870-5528.

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, A B listening tests, equipment clinics, recording studio visits, and audio fun. The club journal is *LC, The SMWTMS Network*. Corresponding member's subscription available. Call (313) 477-6502 (days) or write David Carlstrom, SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

MEMPHIS AREA AUDIO SOCIETY being formed. Serious audiophiles contact J.J. McBride, 8181 Wind Valley Cove, Memphis, TN 38125, (901) 756-6831.

AUDIO SOCIETY OF MINNESOTA. Audiophiles, music lovers, scratch builders, record collectors, tube freaks, digital freaks—we've got 'em all! Monthly meeting, tours, audiophile concerts, special guests, etc. Now in our 12th consecutive year! Write ASM, PO Box 32293, Fridley, MN 55432.

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

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 cassettes. All operation is via the mail. SFORZANDO, c/o E.A. Rawlings, 5411 Bocage St., Montreal, Canada H4J 1A2.

PACIFIC NORTHWEST AUDIO SOCIETY (PAS) consists of 60 audio enthusiasts meeting monthly, second Wednesdays, 7:30 to 9:30 p.m. at 4545 Island Crest Way, Mercer Island, Washington. Be our guest, write Box 435, Mercer Island, WA 98040 or call Bob McDonald, (206) 232-8130.

WASHINGTON AREA AUDIO SOCIETY Meetings are held every two weeks, on Fridays from 1900 hrs. to 2130 hrs. at the Charles Barrett Elementary School in the city of Alexandria, Va. Prospective members are welcome but must register in advance in order to be admitted to the meetings. No exceptions please. If interested please call Horace Vignale, (703) 578-4929.

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

THE BOSTON AUDIO SOCIETY invites you to join and receive the bimonthly *B.A.S. SPEAKER* with reviews, debates, scientific analyses, and summaries of lectures by major engineers. Read about Apogee, Nytal, Conrad-Johnson, dbx digital, Snell, music criticism and other topics. Rates on request. PO Box 211, Boston, MA 02126.

THE ATLANTA AUDIO SOCIETY is dedicated to furnish pleasure and education for people with a common interest in fine music and audio equipment. Monthly meetings often feature guest speakers from the audio manufacturing and recording industry. Members receive a monthly newsletter. Call: Chuck Bruce, (404) 876-5659, or Denny Meeker, (404) 872-0428, or write: PO Box 361, Marietta, GA 30061.

CHICAGO AREA ENTHUSIASTS WANTED for audio construction club. Call Tom, (312) 558-3377 or (708) 516-0170 evenings for details.

THE OREGON TRIODE SOCIETY is seeking men and women who are interested in good sound reproduction and the music it re-creates. We are 80+ members strong and meet bi-monthly in various locations in the Portland area. Our bi-monthly newsletter is *Positive Feedback*, a vital forum on audio and a host of related subjects. For information on our next meeting and newsletter, contact Richard Eggerston, 3623 S.E. Hawthorne, Portland, OR 97214, (503) 238-1957 or Ian Joel (503) 233-1079.

LONDON LIVE D.I.Y. HI FI CIRCLE has formed in London, England. We meet quarterly, with an open agenda of anything to do with any aspect of audio design and construction. For information contact Dick Bowman, 081 520 6334.

TUBE AUDIO ENTHUSIASTS. Northern California club meets every other month. For next meeting announcement send a self-addressed, stamped #10 envelope to Tim Eding, PO Box 611662, San Jose, CA 95161.

WANT TO START OR JOIN CLUB IN CENTRAL ILLINOIS (Peoria, Bloomington, Champaign, Kankakee area). Speaker building and audio in general. Trade info and parts. (815) 657-8488 evenings or weekends.

Advertising Index

Pox Humana

There's no power like DC power.

by Fernando Garcia Viesca

From my college days, I can recall several wild characters, but a fellow I will call Luis is etched in my memory. He was our indigenous Edison. He had some far-fetched ideas, but he was most remembered for his staunch support of DC current—pure, unblemished DC without a trace of corruption by AC, never having been in the form of "unpure" AC. This effectively ruled out any power supply made with rectifiers; power had to come from batteries, solar cells, or DC generators—and nothing else.

We cannot be sure how this addiction started. Legend has it he became enamored with the simple and predictable DC circuit analysis in his first electronics courses, making him hate all the complexities of AC. Imaginary numbers, polar-rectangular conversions, negative reactances frequency-domain responses, and phasors were absolute heresy to him. Never mind the proven advantages of AC current like transformers. He wouldn't listen.

What about the AC components inside any electronic equipment? "Well," Luis countered, "this is a sublimated form of AC since it has been produced from DC." He also said that in most instances this AC voltage had to coexist with DC voltages in the form of bias. There was no way to knock any sense into him.

He was also a music lover and an audio fanatic. Therefore, he designed his own "purist" power supply, a contraption that could have been devised only by a genius or by a madman. It used salvaged components. A 1 HP electrical motor, by means of belts and pulleys, drove an alternator from a '66 Plymouth, that in turn charged an automotive-type lead-acid battery. From there, through a maze of cables, it supplied power to his hi-fi setup, a highly modified auto stereo with custom circuits designed to run at this voltage.

The equipment was by no means cheap. Quite the contrary, much design effort was built in to operate it within the power constraints. Among other things, it was one of the first times I saw carefully matched bridge amplifiers and solid-state DC-to-DC converters, active biasing to center the bias accurately and maximize signal swing. Grounding layout was first-rate, with a heavy ground bus bar to simulate a chassis (shades of automotive environment). About the only piece of equipment spared from being DC powered was the turntable, on the basis that AC power did not act on the signal path. In those years, turntables all used small synchronous motors. If the newer DC brushless turntables had been available, I'm sure Luis would have attempted to power it from the same source.

Doomsday, however, was very near. One night Luis was bragging with some fellow students, "There is no power like DC power."

The other roommate countered, "All this nonsense about DC purity is just that, nonsense. Do you know that the alternator you use to power up your stereo is just what its name implies, an alternating current source?"

Luis was speechless. "But it produces DC voltage."

"True. It has an internal three-phase rectifier."

He was shown the press-fit rectifiers assembled into the heavy alternator plate. Luis was absolutely numb. Never mind that the battery would actually store and supply the current, it once had been AC current! He had the shocked look of a man suddenly discovering his wife has been unfaithful.

In the following months, I recall that Luis grew distant, and though his love for electronics did not wither, it was tarnished. He still studied and passed the last three semesters and graduated with a degree in electronics engineering. However, his sharp wit had vanished.

I don't know what happened to Luis. The last I heard, it was rumored he was living a secluded life in the Sierras, working out the curative properties of peyote and other hallucinogens with the local witch doctors.

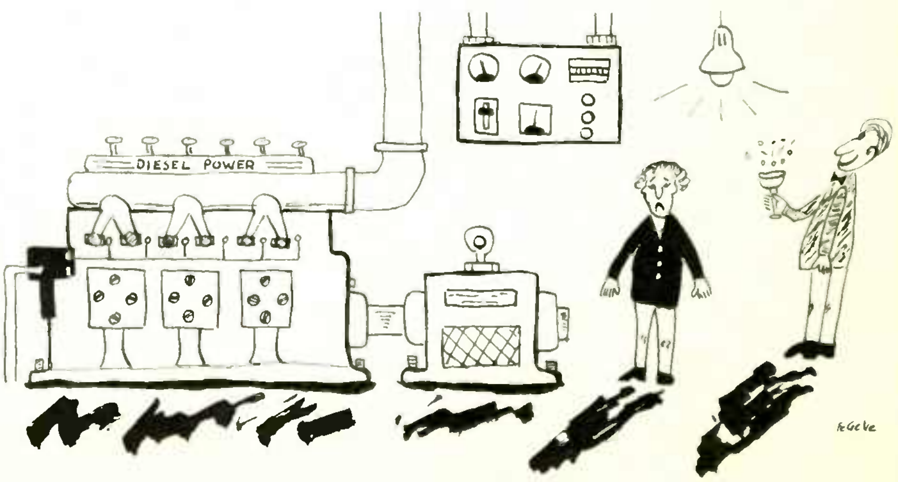
Several years later, I started repairing professional audio equipment because of my love for audio and because it provided a nice economic sideline. I once had a terrible struggle with a particular device. (Electronic circuits may work on paper, but in the real world, they defy all the laws of physics.) The circuit in question was a simple one: the phantom power supply used in certain microphones in professional studio equipment. I could not find the culprit, but I had a severe noise problem when the phantom-powered microphone to certain inputs was hooked to a 24 x 16 mixing console. This would not happen to ordinary dynamic microphones.

The studio owner wanted to use (and with good reason) his expensive Neumans on any input. This problem was compounded by the fact that the console manufacturer had gone out of business. Mixing consoles are some of the most expensive pieces of equipment in a recording studio. However, they are relatively simple (on paper), and many garage-companies built them and often went broke. Therefore, I had no technical support except for an error-ridden schematic.

I attempted all sorts of cures: more ground paths, capacitor decoupling, rerouting wires, external power supply, and anything short of calling the ghost busters—to no avail.

At one point, I seriously thought about giving up, but my pride and the fact that I could not bill the customer if the unit was not fixed prevented that. On a particularly frustrating evening, once again asking myself "what ifs," an idea occurred to me: what if I use batteries to power it? To my amusement, as soon as I tried it, everything worked fine. Although a less-than-ideal situation, it worked for the studio owner until he finally got rid of the haunted console.

I have thought about this situation often in the years since. Although I realize I must have overlooked a fact or done something very wrong, I cannot figure out what. Nevertheless, the whole episode now makes me smile and think: "You were right, Luis. There is no power like DC power."



BEFORE I CAN SHOW YOU MY TRI-AMPED, VACUUM TUBE-POWERED, ELECTROSTATIC SPEAKERS AND SURROUND SOUND, WE MUST ALLOW THE 280 HP POWER SUPPLY TO WARM UP...

The MDM 85 is a mid range 75mm soft dome unit of extremely high standard, both from a design and technical viewpoint.

It incorporates the renowned Morel double magnet and Hexatech voice coil techniques, and results in a unit of above average sensitivity with extremely low distortion and high power handling capability.

With an output level of 96dB distortion in the area of 400-800Hz is slightly over 1% falling to 0.015% from 1Khz.

There are two different types available, one with a rear enclosure and one without (MDM 85NE). The type with the rear enclosure can be fitted into a cabinet as an integral unit.

The MDM 85NE without the rear enclosure can only be fitted into a system having a separate housing to enclose the unit. A volume of 0.7 litre is recommended for this housing, which is essential to prevent inter-reaction with the bass unit compressions and expansions. This housing must be filled full with damping material, such as fibreglass or rock wool.

The Thiele small parameters are given for both types under specifications. The contribution of this unit to a suitably designed system will be evident in the clarity and detail given in the 500-5000Hz region.

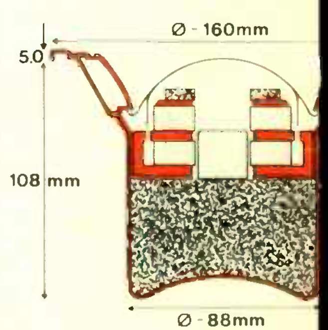
Specification

MDM 85 (with enclosure)

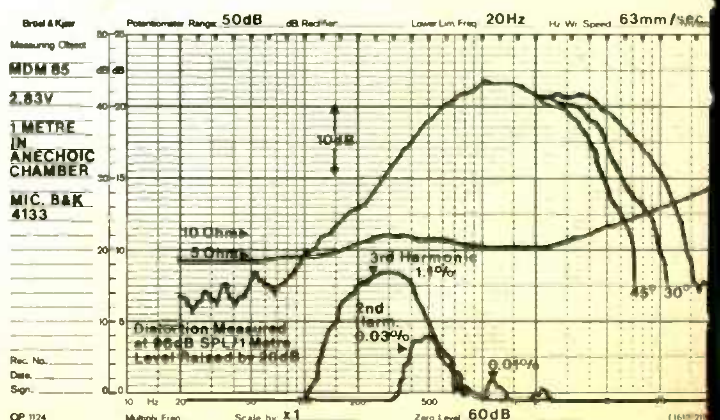
Overall Dimensions	Ø - 160mm x 113mm
Nominal Power Handling Din	300W
Transient Power 10ms	1500W
Voice Coil Diameter	75mm (3") Hexatech Aluminium
Voice Coil Former	Aluminium
Frequency Response	300-5000 Hz
Resonant Frequency	250 Hz
Sensitivity	92 dB (1W/1M)
Nominal Impedance	8 ohms
Harmonic Distortion for 96 dB SPL	<1%
Intermodulation Distortion for 96 dB SPL	<0.25%
Voice Coil Inductance @ 1 Khz	0.2mh
Air Gap Width	1.05mm
Air Gap Height	3.0mm
Voice Coil Height	6.0mm
Flux Density	1.0T
Force Factor (BXL)	4.6 WB/M
Rdc	5.2 ohms
Rmec	37.90
Qms	0.29
Qes	2.66
Q/T	0.20
Vas	0.33 litre
Moving Mass including Air Load	7.0 grams
Effective Dome Area	63.50 cm^2
Dome Material	Chemically Treated Fabric
Nett Weight	1.25 kg

Variations to specification for MDM
(without enclosure)

Overall Dimensions	Ø - 160mm
Frequency Response	250-
Resonant Frequency	
Rmec	
Qms	
Qes	
Q/T	
Vas	
Nett Weight	



Specifications given are as after 24 hours of running.



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Morel operate a policy of continuous product design improvement, consequently specifications are subject to alteration without prior notice

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