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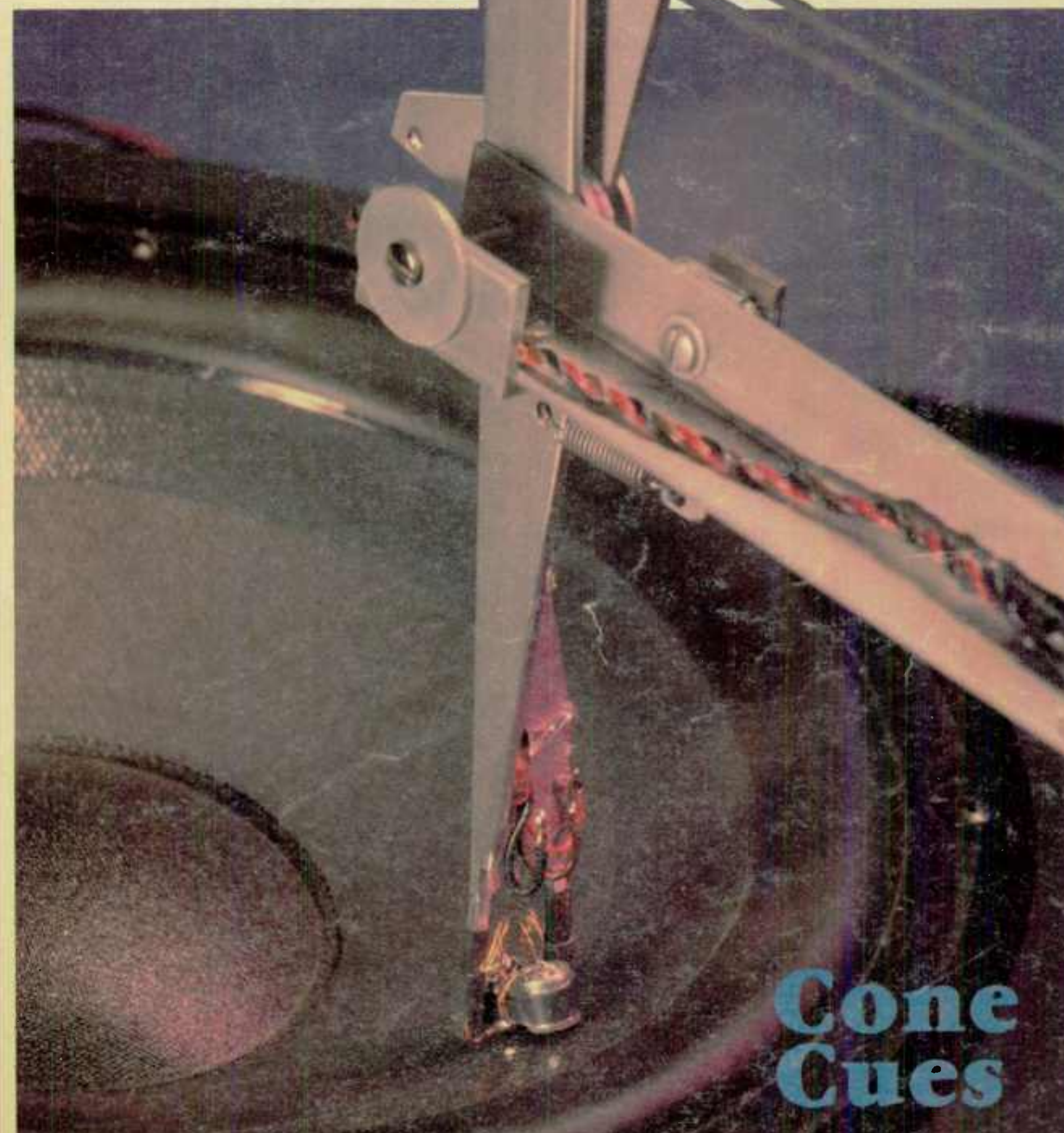
Speaker Builder

THE LOUDSPEAKER JOURNAL



Madisound Presents

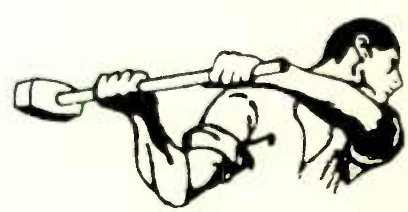
Sledgling

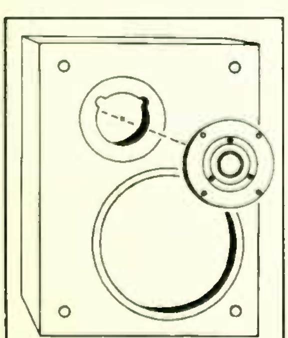
The Sledgling is the best little bookshelf speaker kit on the market today. This kit gives you the quality sound of commercially produced speaker at a fraction of the price. The Sledgling speaker is small enough to fit unobtrusively in any den or be used as a rear channel speaker in a four channel audio or video system. (Don't tell anyone, but this is a great car system too!) We chose a high quality six and one half inch polypropylene woofer and a Vifa treated textile dome tweeter with ferrofluid for a very clean sound. We have utilized the best possible components in the crossover network with Sidewinder coils and Chateauroux Metallized Polypropylene capacitors on an epoxy circuit board.

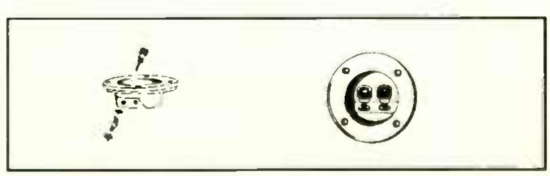
The kit is designed to be as easy as possible to assemble. The cabinet is already finished. The holes for the drivers and the input cup are pre-cut; the crossover is preassembled, and the grill cloth is stretched on the frame. The assembly of this kit does require some soldering ability.

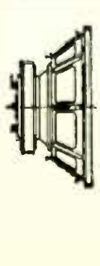
Specifications:

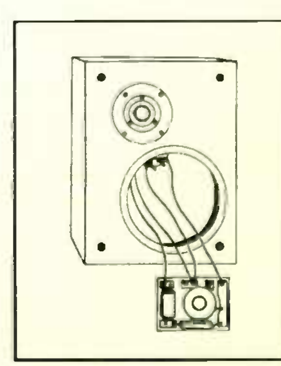
Impedance	4 Ohms
Sensitivity	90 DB
Power Rating	50 Watts
Freq. Range	60 - 18000 HZ +/- 2db
Woofer	6 1/2 "
Tweeter	3/4" Dome
Crossover	6 DB w/ Sidewinder Coils and Polypropylene Capacitors
Cabinet	Walnut Wood Veneer or Black Lacquer 9"x12"x6"
Price	$130/pair Walnut $125/pair Black











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

Good News

The availability of dual filters (high-pass and low-pass on the same chip) has been announced by **DECOURSEY ENGINEERING**, for use in their crossovers. These filters are plug-in, to be used in the Model 120 crossovers and can be supplied to operate on any frequency normally used in an audio system. Attenuation slopes available are 24dB/octave or 30dB/octave. Price: $56.25 each, 100Hz and higher; $61.88 each, less than 100Hz.

Model 120 stereo crossovers will be supplied with these filters at the following prices (add $5.63 for any filter less than 100Hz): 120-B4 biamp, $197.50; 120-T8 triamp, $322.50; 120-Q8 quad-amp, $442.50.

Contact: Decoursey Engineering Lab., 11828 Jefferson Blvd., Culver City, CA 90230.

Fast Reply #KC18

EUPHONIC TECHNOLOGY, currently incorporating Philips' improved digital-to-analog converter and digital filter in its ET650PX MkII CD player, is now making these ICs available to hobbyists and experimenters, on a limited basis.

The TDA1541A, Philips' new high-resolution dual 16-bit DAC, delivers improved low-level linearity, signal-to-noise and channel separation.

The SAA7220P/B digital filter utilizes a new design which also offers improved dynamic range.

Each IC is pin-to-pin compatible with its previous version. All Magnavox and Philips-based CDs, including the CDB-650, 460, 465 and 560, and 471–473 models, can benefit from the new circuits. They are available as a set, including one 24-pin and 28-pin gold-plated, machined pin socket; the price is $89.95, plus shipping.

The MkII CD player is available from selected dealers and company-direct; price $1,295.

For more information contact Euphonic Technology, 19 Danbury Rd., Ridgefield, CT 06877, (203) 431-6434; or 800-444-1428, for credit card orders.

Fast Reply #KC138

Several new products have been announced by **MONSTER CABLE**, a manufacturer of high-performance audio interconnect and loudspeaker cable.

The T-Series Turbine connector is an RCA type, redesigned with a massive ground connection utilizing a series of 12 diagonal cuts along its circumference. Multiple points maximize contact area and allow a more complete transfer area of the audio signal. The connector also features the Monster split-center pin construction and 24k hard-gold plating. The price is $35/pair.

A new Interlink CD cable is offered to provide improved performance for entry-level CD systems, which come equipped with ordinary connecting wire. It features two high-performance strandings, plus a shield, to handle the sonic qualities most CDs are capable of delivering. Price: $29.95/pair, 1m; $39.95/pair, 2m; $76/pair, 20 ft.

Monster Video 2 is designed for improved video picture quality, signal transfer and protection from interference, for VCRs, VHS and Beta players, laserdisc players, video-based digital audio recorders, digital-to-digital connections, TV receivers, HD monitors, projection TV, or satellite dish equipment and programming. The construction uses a multi-stranded copper center conductor, double shielding of braid and conductive plastic, and is wrapped with low-loss dielectrics. Suggested retail price: $14.95/meter length, with RCA connectors; $16.95/meter length, with F-type connectors. A Video 2/Interlink 300 cable connection kit is $29.95/meter set.

For more information contact: Monster Cable Products Inc., 101 Townsend St., San Francisco, CA 94107, or call Gary Reber, Los Angeles Division, (213) 322-8200.

Fast Reply #KC664

SYN-AUD-CON announces "The Audio System From an Electronics Viewpoint," a grounding, shielding and installation workshop, January 27-29, at the Holiday Inn, Anaheim, CA. The workshop staff is Allen Burdick, Benchmark Media Systems; John Lanphere, Altec Lansing; and Edward Lethert, SECO. A $100 fee includes supplies, manual and luncheons.

For registration details contact: Synergetic Audio Concepts, Rt. 1, PO Box 267, Norman, IN 47264, (812) 995-8212; FAX, (812) 995-2110.

Fast Reply #KC41

SCIENTIFIC DESIGN SOFTWARE announces that Computer-Aided Crossover Design, currently available for IBM format computers, will soon be released for use with the Apple Macintosh.

CACD is a circuit optimization program designed to develop both passive and active crossover networks for loudspeaker systems. CACD greatly reduces the trial-and-error process of determining crossover topology and component values, and includes the generation of an equivalent impedance circuit model of any driver, or driver/enclosure combination, so all network designs utilize the actual driver load and driver natural frequency response. It also predicts the response of the network and driver combination and the input impedance for amplifier load verification.

The software's data library of 750 drivers and a library of standard circuits reduce hand entry of data. The program includes a reference book outlining loudspeaker system design, documentation and manual. Suggested retail price for the IBM version is $349.95.

For further information: Scientific Design Software, PO Box 3248, Chatsworth, CA 91313, (818) 718-1201.

Fast Reply #KC14

SPEAKER BUILDER

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A Note To Contributors

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

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

ACOUSTAT announces the new Spectra Passive Woofer. The SPW-1, utilizing four 6.5-inch woofers in a tuned enclosure, is designed to increase the dynamic range of their Spectra series speakers.

A modified 100Hz, 12dB/octave crossover relieves the electrostatic panels of the bottom two octaves, allowing a 10dB increase in dynamics with greater bass impact. A network bypass switch in the woofer allows the use of an external electronic crossover for biamped systems.

The SPW-1 is available with beige or black cloth and a wood finish of light, dark or black oak. The woofer has a retail price of $600.

For further information, Acoustat, 613 S. Rockford Dr., Tempe, AZ 85281, (602) 967-3565.

Fast Reply #KC166

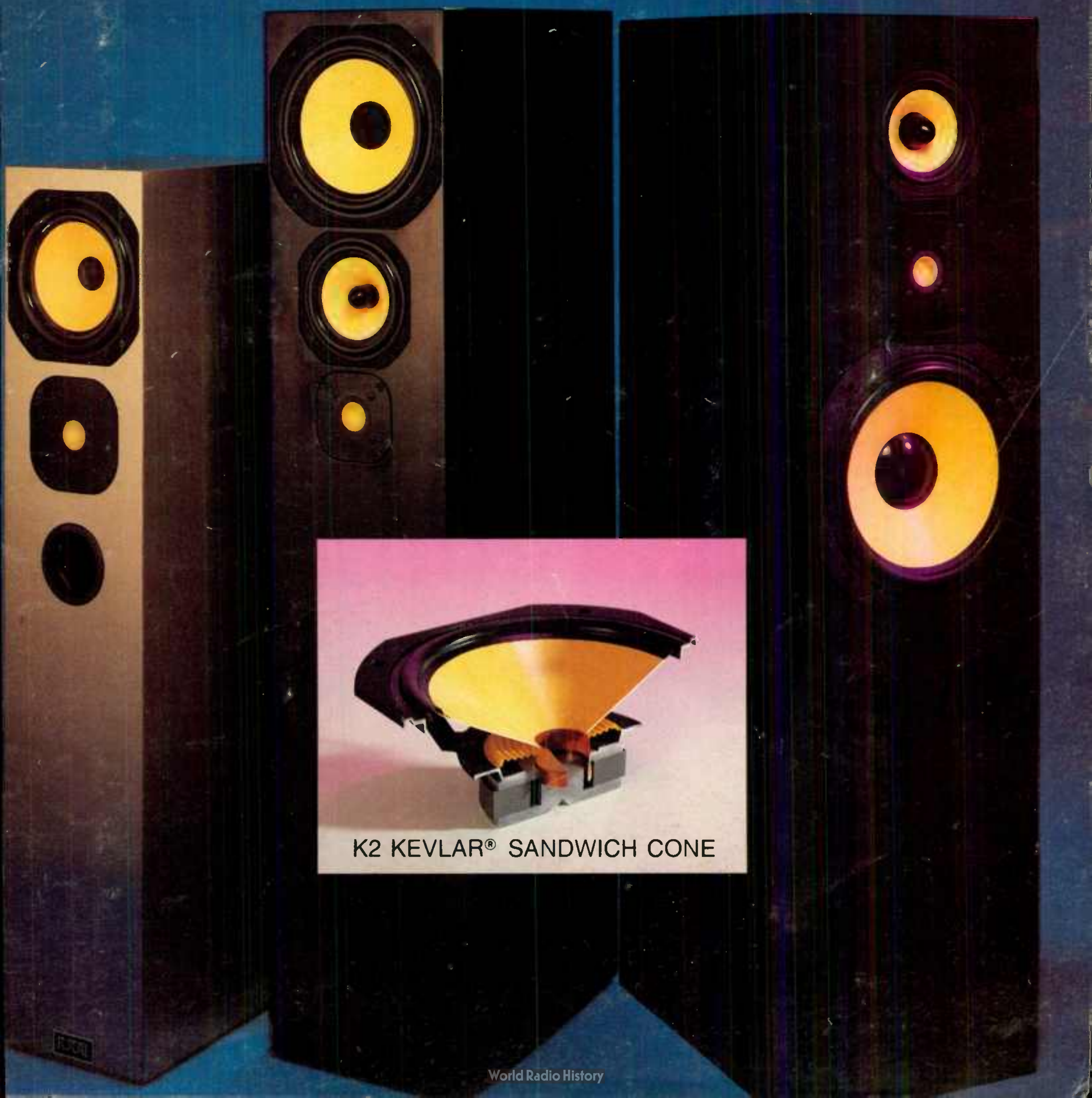
ACOUSTICAL MAGIC, manufacturers of speaker cabinet insulation for 30 years, has developed Flexible Borosilicate Ceramic Mineral Coatings, which reduce the colorations in speaker cabinets.

The coatings, when applied to enclosures with cone drivers, are claimed to improve transparency with musical passages and individual instrumental lines, with a fast, open sound similar to ribbons and ESLs. The product may be brushed on; drying time is four hours; two coats are recommended; nontoxic and nonpollutant. Prices are $17/quart and $58/gallon, plus shipping.

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

Fast Reply #KC27

Fast Reply #KC147



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About This Issue

Bruce Edgar's interview with **Keith Johnson** leads off this largest *Speaker Builder* issue ever. Johnson's thoughts are backed by years of construction, experimentation, recording and most of all, creative listening. **David Lang** looked back at one of the early formats of the electrostatic and figured out a way to clone it. His sheathed conductors are woven on commonly available fluorescent lamp diffusers, giving ESL enthusiasts yet another way to realize their favorite format (page 18).

One of our readers in France, **Jean Margerand**, did his homework on a new way to load drivers in vented enclosures and the first installment of his work begins on page 29. **David Weems** opined that chimney tiles could work as low-resonance enclosure walls. His guess, along with an arduous technique for cutting the stuff, make good reading beginning on page 37.

Just how many henrys is that coil, exactly? Contributing Editor **G.R. Koonce** has devised a measuring technique that does not require a bridge (page 43). The **Reg Williamson/Alan Watling** team from Britain gives us a fine implementation of **Peter Baxandall**'s instrument (*SB* 5/88, p. 9) for diagnosing how your amps and speakers are getting along (page 47).

Dick Pierce will introduce you to Chrome Dome on page 46. **Jim Frane** updates his Polk mod (p. 60), **Peter Muxlow** reports on some woofer magic from England (p. 58) and reviews the new Borwick speaker and headphone book on page 50, where **Gary Galo** also reviews some design software in the review department.

Although our early renewal offer ends as you receive this issue, there's still time to renew before the first big 1989 issue comes off the press in January.

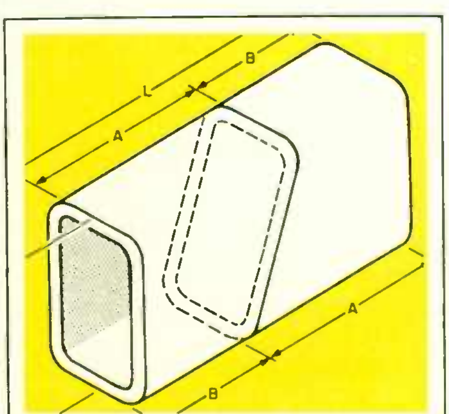
Cover photo by **Keith Johnson.**

Speaker Builder

THE LOUDSPEAKER JOURNAL

VOLUME 9 NUMBER 6 NOVEMBER 1988

DEPARTMENTS



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FEATURES

Editorial

A PROGRESS REPORT

In late 1987 we made several decisions based on the belief that *Speaker Builder* had come of age. We decided to make the magazine a bi-monthly, recruited an advertising salesperson, and strongly committed ourselves to promote the magazine to new readers.

We were also sure that *Speaker Builder* had become part of a new segment of the loudspeaker business in the United States. We were convinced the industry needed a communication medium that could keep up with the news and new developments in loudspeaker technology. We founded *Voice Coil*, a monthly newsletter edited by Vance Dickason.

The industry has responded positively to the publication which is now read by about 40% of the industry.

For more than a year we had been actively searching for a new home for our publications. We were fortunate to find a suitable location in the West Peterborough section of town and also fortunate to find contractors who did the necessary renovations on schedule so we were able to move into new offices at the end of May.

Staff cooperation and enthusiasm were major factors in the smoothness of the move. Not only did we move our offices, graphics equipment and darkroom and our computer systems, we moved almost 100 pallets of back issues of the magazines from overage, soon-to-be demolished chicken coops to dry, accessible storage only a block from our new work facilities. On magazine moving day a large forklift lifted pallets out of the chicken palace, onto an overlong flatbed truck, across town to their new home.

While all this was going on, we settled into the new quarters and struggled to keep on schedule. Rally Dennis, our new ad representative, stepped up his year-long campaign to bring more vendors into *Speaker Builder*'s pages. Advertising contracts and pages began to multiply.

In our efforts to find new readers, we had a lot of help. Our friends at both *Stereophile* and *Absolute Sound* were a major source of assistance in several ways—and continue to be helpful. We used new means to reach readers from related enterprises, like car stereo buyers and those interested in woodworking.

We began 1988 with some 4,700 subscribers. As I write this the US/Canada total is just over 9,300. We have had the pleasure of moving from an occasional second color to full color on our covers—which will continue in 1989. This issue is 88 pages, our largest ever.

But lots of magazines get larger and add more advertisers. The hard part is keeping the quality of the content high—or making it even better. Believe it or not, for *Speaker Builder* that is the easy part. The work our authors have produced for these last six issues is as good or better than that in any of the magazine's previous eight volumes. And the quality is a direct result of their imagination and diligence.

If this sounds like patting ourselves on the back, for me at least, it is rather an expression of my profound admiration and gratitude for the work of the staff and the work of the dozens of authors who make the whole enterprise possible.

We have exciting plans for 1989, *Speaker Builder*'s tenth anniversary. The variety of the articles we are readying for publication in the upcoming issues is the equal of any we have ever done. And the ideas are fresh and new. Two new interviews are in the works, construction pieces abound, and two of them are unique designs. I look forward to 1989 with great enthusiasm and anticipation.

CONTAMINATION CONCEPT

If you were not present at the Audio Engineering Society's most recent meeting in Los Angeles (Nov. 3-6), you missed a paper delivered by Deane Jensen and Gary Sokolich entitled "Spectral Contamination Measurement." Let me be the first to call your attention to it.

Mr. Jensen is well-known for his Jensen transformers but has also pioneered the use of the Hewlett-Packard computer system for sound analysis work. The test facility required for this new form of distortion analysis is complex and expensive but the results of the Jensen-Sokolich preliminary work show great promise for isolating some new clues as to how amplifying devices color signals passing through them.

Briefly, they use a Wavetek Waveform generator to produce a "picket fence" series of signals about 30Hz wide, alternating with frequency spaces of 30Hz with no signal in them. Along with an H-P computer to control the process with a number of filters, attenuators and amplifiers, they produce nonlinear distortion products in the areas between the signal which Jensen and Sokolich call "spectral contamination."

I will not venture to use the term "breakthrough" about this paper. It is too early for that. Those of you who have labs at your disposal for duplicating this work, are strongly encouraged to do so. Doubtless Jensen and Sokolich will welcome your input, and those of us who cannot participate will wait expectantly on the sidelines for your field reports.

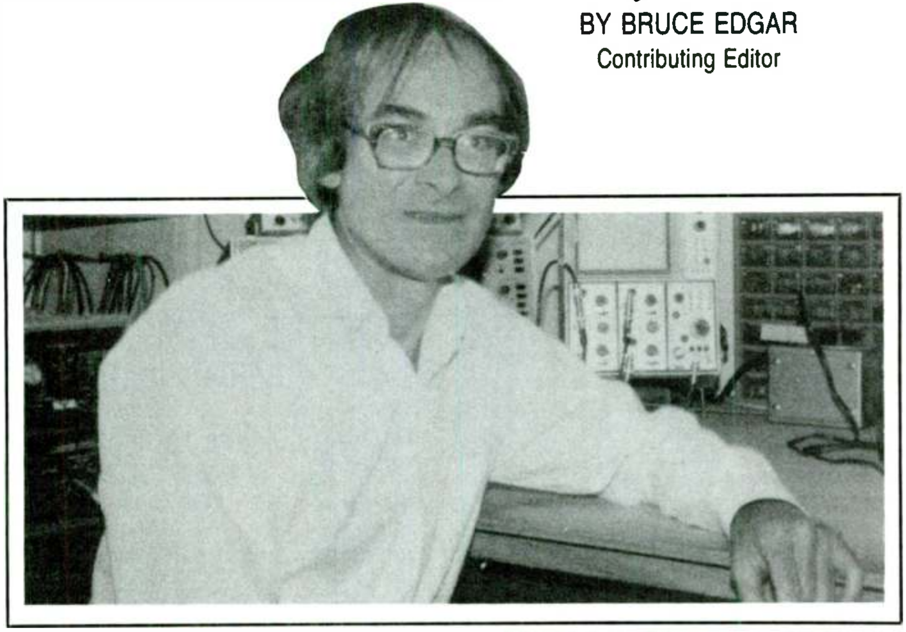
The paper gives enough samples of distortion products not heretofore quantified in just this way—to give hope to those of us who hear anomalies in sound reproducing equipment that are not fully accounted for by the currently available distortion measuring methods—that these mysteries will begin to have scientifically reliable verification.

—E.T.D.

(Preprint 2725, I-2, available from the AES, 60 East 42nd St., New York, NY 10165, $4.00 for members and $5.00 for non-members, ppd. in the US.)

A CONVERSATION WITH KEITH JOHNSON

BY BRUCE EDGAR
Contributing Editor



In some way, every loudspeaker reflects the personality, that is, the personal likes and dislikes of its designer, especially if the end product is tuned by ear. In the case where one particular person is connected to a series of speaker designs, it is always interesting to ask this designer for the rationale behind his design choices and to gain some insight into his thinking and creative processes.

Thus it is refreshing to talk to a designer-engineer like Keith Johnson, who can give a clear explanation of how his designs have evolved and pass on useful information to *Speaker Builder* readers.

I visited Keith in his small but tidy workshop/office, located in a rural area south of San Francisco. He maintains a very active consulting business in loudspeaker design, instrumentation, and digital circuit design, besides being the recording engineer for Reference Recordings. Keith is the only loudspeaker engineer I know who has two Grammy awards to his credit.

ABOUT THE INTERVIEWER

Dr. Bruce Edgar is a Space Scientist/Project Engineer for a Los Angeles based aerospace company and a Contributing Editor for SB. His interests include horn loudspeaker design, woodworking, and the history of loudspeakers.

Speaker Builder: Keith, you've been involved with audio for a long time and have achieved a certain amount of fame as a recording engineer. But how did you become involved with loudspeakers?

Keith Johnson: Probably, it started when I began recording, which goes back to high school in Dallas. I did a stereo recording of a pep rally using a much modified Pentron tape recorder. I also built a pair of Williamson amplifiers and installed a pair of 5-inch Goodmans speakers on open backboards.

SB: That might have been one of Ted Jordan's original full-range designs (*SB* 2/84).

KJ: I believe it was. It was an exquistely good little driver. It was my first encounter with speech reproduction that sounded definitely like speech. They eventually died after much use. I moved to the San Francisco region where I built a three-track recorder. I came across a number of BBC Goodmans monitor drivers that I liked and proceeded to build enclosures for them.

SB: What kind of enclosures were they?

KJ: They were sealed enclosures that had active equalization to extend the bottom end. I experimented with cone treatments that I learned from an engineer at Ampex, placing little bits of tape selectively on the cone to smooth out the response.

SB: How did the three-track machine originate?

KJ: In the early fifties, stereo recording at the laboratory level used three tracks. It was Ampex' first multitrack endeavor. As a high school student, I had a chance to see that system. I liked it very much and decided to build one for myself, as kids do when they get started. I started to rebuild the Pentron recorder. Eventually, all that was left of the original machine was the metal frame and the capstan bearing. But it gave me a chance to pick off the mechanical parts one at a time and make improvements on them. The playback system used three Williamson amplifiers with three 8-inch monitor speakers.

SB: How did you figure out where to put the tape on the cone?

KJ: You do that by playing pink noise through the speaker on a partially sealed box so the back wave won't confuse things. Then you touch parts of the cone symmetrically with all five fingers and find positions that make significant differences in the sonics of the pink noise. And the larger the difference, the more likely you have placed your fingers on a nodal structure—the peak part, not the interference part. By this mapping procedure, it is possible to figure out where all the nodes are. You can make little holes at the nodes, or you can try putting little bits of tape on the peaks to break up the nodal structure, although it will appear in a different way. By playing games with these tricks, it is possible to take a speaker with cone cry colorations and more or less make them appear to go away. I still do the same things to my speakers today.

SB: Where did you go to college?

KJ: I went to Stanford and received an Engineer's Degree in E.E. I also took pipe organ instruction at Stanford and grew to like music a great deal. That's why recording and speakers became more serious endeavors because at that time there wasn't anything good available in stereo—it was all mono then. I could go out and make a three-track recording of a concert and play it back on my three monitors. And there wasn't anything close to it available commercially.

At that time the three-track machine was not very portable. Because there were many recording opportunities, I made the portable three-track machine, that in its upgraded form we still use today at Reference Recordings.

SB: What was your research topic at Stanford?

KJ: I was involved in building transducers; the prime one was a magnetic flux switcher which was used in oil exploration. My job was to make switchers out of thin films, a technology which proved to be useful to me later in making recording heads and microphone diaphragms.

SB: What microphones were you using?

KJ: I was beginning to build microphones because what was available were either limited in frequency range or very noisy. By luck I encountered some Sennheiser FM microphones. The FM electronics were well thought-out and very quiet. I had to rebuild the diaphragms and chose thin film techniques. In the process, there was an immediate improvement in the sonic quality, even though the frequency response had not changed very much. That was my initial realization that something audible was going on with the mass of a driver. I noticed the same phenomenon with small electrostatic speakers that were just starting to appear. They had a far more articulate top end than did the cone speakers at that time.

SB: Did you experiment with electrostatics?

KJ: Yes, I played around with some little German imports and the Radio Shack Electrostat 3. They didn't work well, but they gave me some background on how they could work if done right. Later, I built a three-channel electrostatic system

PHOTO 1: The three-track tape recorder used by Keith Johnson in recording sessions for Reference Recordings.

which clearly showed up the cone cry problems of the 8-inch monitors.

SB: What do you mean by cone cry?

KJ: I mean the resonant breakup modes on cone speakers. I used the 8-inch monitors for years. They added to my knowledge that there is every indication that you can hear "mass"—that is you can hear how "heavy" a diaphragm is.

I learned about the properties of cone papers, as well as other cone materials, the interplay of porocity and fiber length, and the different paper fabrication processes. I also experimented with the subtle ways that resins in the cone material can make changes in the acoustical response.

SB: Do you imply by the term "mass," a high frequency rolloff?

KJ: No, it's not. If it were that simple, the Japanese would not be doing the opposite by building speakers with extraordinary heavy materials. They have gone off on a fantasy of building diaphragms out of the heaviest, stiffest, and hardest materials they can find—like carbon fiber, titanium, and so forth.

SB: So how do you hear mass?

KJ: For example, I could put voice coils on an egg shell, a tuna can, and a cardboard box and build a very complex equalizer to make that crude "speaker" system have a very flat response. But even with very flat response curves, it would still sound like its constituent parts; because the human brain can recognize sounds of familar objects even though they are as much as 40dB down.

SB: So you claim that any particular material or mechanical structure induces its own colorations that the ear-brain system can detect.

KJ: Yes, but in the presence of complex signals, with practice you can hear these colorations roughly 40dB below the peak. If you were to add or subtract these sounds 40dB down from the frequency-response curve, you will not see them. They are invisible to normal measurements.

I have coined a word, "eigensonics" to describe the phenomenon. The term eigenvalue comes from the German word for solutions to differential equations for mechanical vibrations. For example, a ringing bell has eigenvalues. And the ear-brain system is remarkably sensitive to these eigenvalues or eigensonics which can be familiar sounds like those of plates, domes, strings and bells.

SB: What are the problems with massive drivers?

KJ: When you supply an impulse of energy to a massive driver, it will store that energy for a period of time; so you will have plenty of time to recognize the things that are going wrong after the signal has changed. When this behavior is damped and low Q, it is more likely to be excited by program frequencies. On a lightweight driver, the impulse is over quickly. The natural behavior may have a higher Q, however its narrowband character makes it less likely to be excited by the program. That's one of the reasons that I like the sound from lightweight drivers over heavier ones. There is a certain neutrality about lightweight parts.

SB: What was the next stage in your career?

KJ: I set up Gauss Electrophysics with a partner, Paul Greg. We both combined inventions to start the company. He had some of the basic video disc concepts worked out, and I had the focus gap recording head with a beamed rf bias that made very high-speed tape duplication possible. We put a few machines out in the field that worked, and overnight that machine set standards for the whole prerecorded tape industry. All of a sudden, the industry was dealing with endless tape loops, continuous duplication without hand-loading, and machines that could run faster than record presses.

SB: How far did you take the Gauss business?

KJ: I stayed with them while Ed May, who had come over from JBL, was starting the Gauss loudspeaker efforts. His son, Dick May, was head of marketing for Gauss. I learned much about speaker manufacturing, design, and testing from Ed. Later, I sold out of Gauss and resumed consulting in the early seventies.

SB: What type of consulting did you do?

KJ: One of my clients was a speaker manufacturer who had a contract with a department store chain for home stereo speakers. It was the perfect chance for me to design drivers from the ground floor up, apply some of my knowledge, and not be picky about the end results.

SB: Who made the drivers?

KJ: The drivers were originally made by CTS and later by others here and abroad.

SB: What were the problems associated with the speakers?

KJ: They were not the problems we have today. At that time the major difficulties were getting a tweeter to go high enough and not be a midrange, a woofer that would go down low enough and not be a midrange, and a midrange that did not have bunch of resonances in the midrange. The other problems were with tooling that was designed for table model radio speakers and with speakers that didn't have room for excursion. Most of the problems were mechanical.

SB: What did you learn from designing these systems?

KJ: I learned about the properties of cone papers, as well as other cone materials, the interplay of porocity and fiber length, and the different paper fabrication processes. I also experimented with the subtle ways that resins in the cone material can make changes in the acoustical response.

I don't have any speaker that I like. They all have their problems as well as their virtues. Much goes back to eigensonics. If you know what the problem is, it just glares at you.

PHOTO 2: Keith examines a driver for excursion and linearity.

SB: Were you able to rid these cheap speakers of their acoustical ills?

KJ: Yes, in a way, but it took me one year to work out a 12-inch woofer. It had a remarkably flat response, and the breakup-mode ripples were probably no more than 2dB. It had a glass polyamide voice coil with a very thick top plate and a small magnet—what I called a "mystery magnet special."

SB: It was probably way underdamped.

KJ: Yes, but everybody is aware of the BL factor or force factor limitations of that design. It was made that way by economic choice. In all other aspects, it was a very serious woofer and will play very well without a crossover. My development time did not add much to the cost of the woofer. The driver ended up in many rack systems, and similar versions were copied by others.

In the bass-reflex version, the panel resonances in the voice region forced me into the compromise of setting the port tuning to a lower frequency. The boomy bass was set at the same level as all of the other obnoxious faults, so that no one fault was compounded upon by others. But that's what goes on in the design of cheap three-way rack systems.

*During a break, Keith pulled out a driver that a client had sent him to check out (*Photo 2*). The way Keith manipulated the speaker with his fingers, checking the suspension linearity and excursion limits, reminded me of a doctor gently palpating a patient's abdomen with his fingers to make a diagnosis.*

SB: What other activities were going on?

KJ: I was also consulting on instrumentation so that loudspeaker design was little more than a hobby. Meanwhile recordings were happening. I needed monitor speakers that were easily transportable and that I could use to evaluate recording sessions. The old BBC monitors could not survive in that environment. So I started to work with catalog speakers like everyone else.

SB: What type of monitors did you use?

KJ: For a long time, they were all 8-inch two-ways, but with larger volumes so that the drivers would not be straining to compress the enclosed air. At any one time, I would have three pairs of different configurations. One pair would be for listening in a room, another pair would be under construction trying to sound better than the pair I hated, and a third pair would be stashed away in case something broke.

SB: Do you find yourself going through love-hate relationships with your speakers?

KJ: Oh yes, I don't have any speaker that I like. They all have their problems as well as their virtues. Much goes back to eigensonics. If you know what the problem is, it just glares at you.

SB: What did you do to improve the catalog speakers?

KJ: First I took the speakers as they were and learned how they behaved and misbehaved. One of the best improvements was to map out the cone and put tiny holes where the nodes were and also put small selective damping materials on the main vibration peaks. It took days to map out a cone.

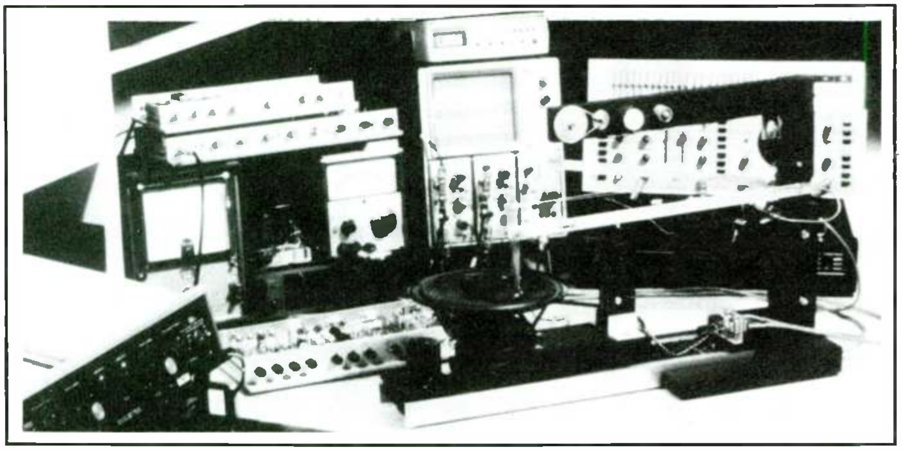


PHOTO 3: The scanning microphone mechanism for mapping out the nodal structure of a diaphragm.

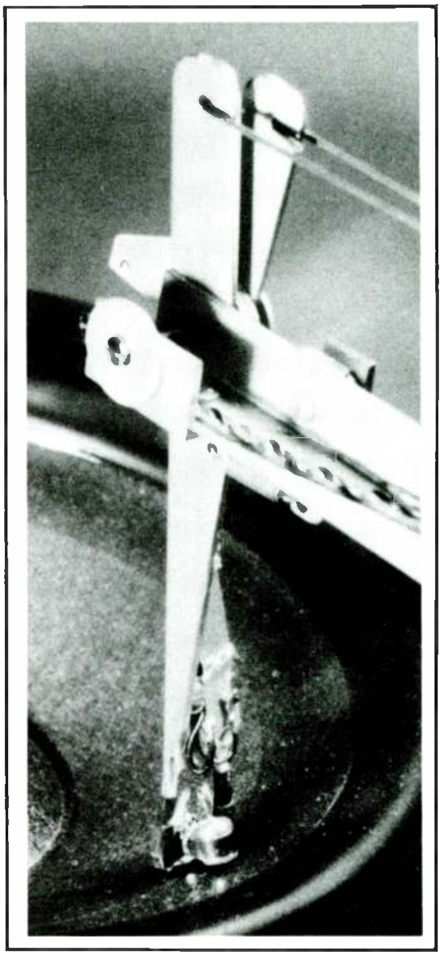


PHOTO 4: A closeup of the differential microphones.

SB: Did you accomplish that with your finger technique?

KJ: At first I did it by hand, but it became obvious that it was a very clumsy way to map a cone. So I built a special differential microphone which allowed me to scan a cone.

SB: What is a differential microphone?

KJ: For example, a ribbon microphone is often a differential microphone. The pickup pattern is a figure-eight with one lobe 180° out of phase with the other lobe. In the case of my microphone, the physical resolution is roughly 1/32-inch for node definition. At first I used my thin film techniques to make the diaphragms. The present ones use matched electrets.

SB: How did you use the differential mike to detect the nodes and peaks?

KJ: If you scan radially along the cone with pink noise excitation and with one lobe pointing down, you can tell which part of the cone is radiating. With a noise generator and a spectrum analyzer or a fast sweep oscillator, the resonant modes become obvious. To find the nodes, position the mike sideways so that the null in the pattern is pointed toward the cone. Where the cone is moving like a piston, there will be little output from the microphone. However at an interference node, the output becomes a sharp spike.

SB: Do you find that these nodes are the size of a pin or are they larger?

KJ: They vary. The most common locations are the center of the dome dust cover, and at the edge where the dome terminates at the cone. Also there are a series of nodes about two-thirds down the cone and along the cone, as well as the surround boundary. This mapping technique gives us a powerful tool for changing cone behavior. At that point, how to change and what to change becomes proprietary knowledge.

SB: What other experiments were you doing with your monitors?

KJ: Probably the most interesting was the comparison between paper and polypropylene cones, where both systems are equalized for close comparative frequency response. In a recording session you have to hear as much as possible from your monitors. So a recording session hinges on them as much as on microphones. And one of the big surprises was how much difference cone materials make in monitor performance.

Of the ones that we have used in recording sessions, the least satisfactory monitor used a polypropylene cone. Its major difficulty was that we could not hear important details in playback, and so the recording would be overmiked or miked too close. Playback on electrostatic speakers made it very obvious that the microphone placement was too close.

Usually, I use a fast-frequency sweep for doing an approximate response. I can do that in a room like my shop here. The sweep is fast enough that the first wall reflection does not affect the results. That procedure works well for frequencies down to 200Hz. Below 200Hz, I do measurements outdoors with a slow sweep and a pen plotter.

SB: What are the problems with polypropylene cones?

KJ: The main one is distortion. Although I haven't done a detailed stress-mode analysis, apparently the material is under stress such that it undergoes plastic deformation rather than elastic deformation. So it has a distinct hysteresis loop problem. I have observed that behavior trying to close a servo loop on a polypropylene woofer. Under high power, it behaves differently than at low power levels.

We are back with the problems of heavy moving structures. It takes a long time to settle after excitation so they have a murky sonic character.

SB: Describe your present monitors (*Photo 5*).

KJ: My pine box monitors have laminated walls, and the one I like the best has

ESOTAR TWEETER

T330D

DYNAUDIO
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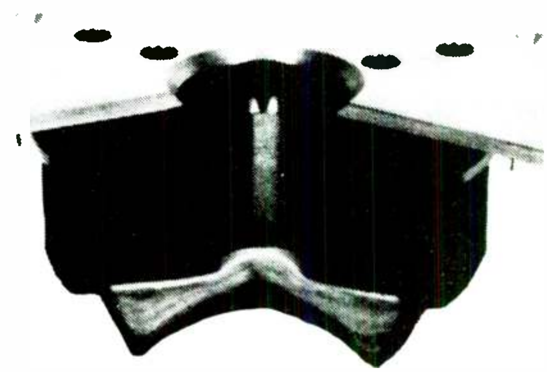
Preliminary Data Sheet

Cone:	
eff. cone area S_D	8.5 cm²
moving mass M_{ms}	0.5 g
lin. volume displacement V_d	6.0 cm³
mechanical resistance R_{ms}	4.7 kg/s
lin. excursion peak to peak X_{max}	0.7 mm
max. excursion peak to peak	3.2 mm
Frequency response	2500-22000Hz
Harmonic distortion	<0.6%
Intermodulation distortion (1000Hz)	<0.05%
Magnet System:	
total gap flux	421µWb
flux density	1.9 Tesla
gap energy	193 mWs
force factor BxL	5.2 Tm
air gap volume V_g	0.16 cm³
air gap height	2.5 mm
air gap width	0.75 mm
Net weight	1.6 kg
Overall dimensions	140 x 66 mm
Power Handling:	
nominal DIN	300W
music DIN	1200W
transient 10 ms	2000W
Q-Factor:	
mechanical Q_{ms}	0.32
electrical Q_{es}	0.29
total Q_{ts}	0.15
Resonance frequency f_s	500Hz
Sensitivity: (2.83V RMS) 2.5 bis 20kHz	1W/1m 92dB
Voice Coil:	
diameter d	28 mm
length h	3.2 mm
layers n	2
inductance (1kHz) l_e	0.09 mH
nom. impedance Z_{vc}	8Ω
min. impedance Z_{min}	6.4Ω
DC resistance R_e	5.2Ω

ESOTAR speaker drivers made by DYNAUDIO are designed and manufactured with only one aim: to reach the summit of quality.

ESOTAR speaker drivers are carefully made in the best and well-known DYNAUDIO tradition, not allowing any compromise whatsoever: research and development, material and craftsmanship.



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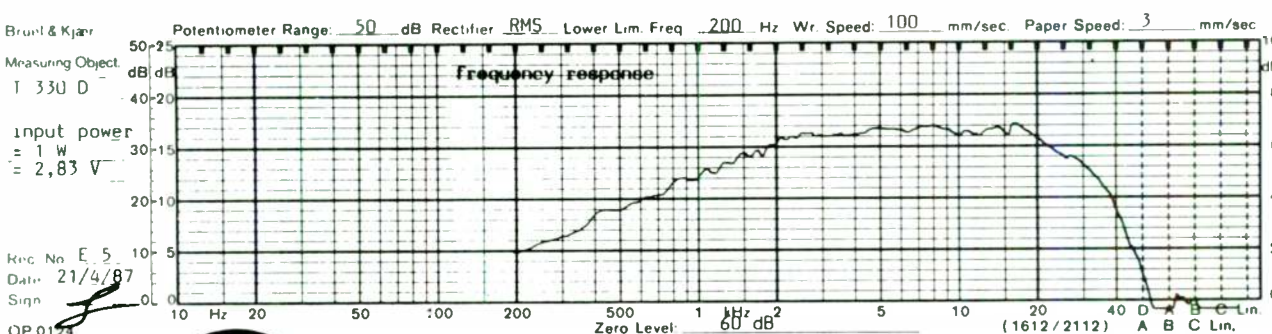


PHOTO 5: Keith Johnson's monitor speakers for his recording sessions. The pine box monitor is on the left, and the "black box" monitor is on the right. Notice the Q-Tip used as a phase divider on the ribbon tweeters. The pipe organ that he is rebuilding is in the background.

a Polydax poly-paper woofer and a Mitsubishi ribbon tweeter (with the horn removed). It has been used for some of our best vocal recording sessions. It has been very truthful in telling us how close or how far the microphones should be or what kind of imaging we should obtain. The speaker plays tremendously loud when it wants to with no signs of stress.

SB: What about your "black box" monitors?

KJ: The black box monitors have formica siding and a biamped system with a servo-controlled woofer. The woofer cone has been tweaked with ridges and holes. They have exciting dynamics and present very good instrument staging. But they also have some strong voice colorations that we have learned to live with. Tam Henderson, our producer at Reference Recordings, just hates them. Everytime I show up with them, he just moans and groans about how awful they sound. Yet the black box monitors have been the most predictable as to the success of the end product.

SB: How do you test for the frequency response of your speakers?

KJ: Usually, I use a fast-frequency sweep for doing an approximate response. I can do that in a room like my shop here. The sweep is fast enough that the first wall reflection does not affect the results. That procedure works well for frequencies down to 200Hz. Below 200Hz, I do measurements outdoors with a slow sweep and a pen plotter.

SB: What kind of a microphone do you use?

KJ: I use a B&K with a ¼-inch head, because I can trust it off axis with my clumsy measurements. With the ½-inch head you have to be right on with the axis alignment, or you don't get true high-frequency measurements.

SB: How do you handle imaging?

KJ: Imaging is very much related to out-of-phase or random-phase behavior off axis. Imaging is absolutely paramount in a speaker. Because when it works, it is absolutely magical with a good recording and a death knell with a bad recording. In a speaker that doesn't image well, a bad recording will not sound too differently from a good recording because the speaker doesn't tell you what's happening.

The classic example of speakers that don't image too well are the old studio monitors. In fact, often the better the recording, the worse it sounded on them.

SB: What was the problem?

KJ: The main culprit was the midrange horns that have diffraction and internal reflection effects that create weird phase characteristics, particularly off axis. Their designers had only one objective in mind: a flat on-axis response.

SB: How do you test for imaging?

KJ: When two stereo speakers are connected in mono, the sound should appear to come from the center. If you reconnect for stereo and reverse the polarity to one of them, then the sound should appear to come from the sides or the rear. On some monitors, you can't tell the difference between a stereo and a mono connection except that you lose some bottom end with the latter. Or you reverse polarity in one speaker, and you can't tell the difference.

SB: Could you say that consumer speakers have better imaging?

KJ: Oh yes. Interestingly, the rack systems, certainly the ones I designed, have very good imaging. There was a while there when the consumer had much better imaging capabilities than the recording engineer in his studio. And it became apparent when chorus effects in modern synthesizers were not being well-reproduced in studio monitors. That was one reason why the little black Yamaha monitors have become so popular in the studios. They have good imaging, and they predict reasonably well how a recording will sound on a variety of systems.

SB: When did Onkyo approach you?

KJ: It was about two years ago. They had heard about me from a number of sources. I was contacted by David Birch Jones of Onkyo USA, who wanted Onkyo to get into the US speaker market. The US marketing arm had already reported back that the Japanese product would be unsuitable for the US market. And also Onkyo would have to start from scratch and get an American designer to deal with products that would work here. Onkyo decided to set up a separate speaker product line which became known as Precise.

Things started quickly. I had to set up listening sessions with American and English speakers, literally training the Japanese engineers how to listen to the speakers and what was important for the American market. We went through several days of hard listening. For a while no one was hearing the difference. Eventually, we got some consensus. The best material we found that could show how it could work were *Star of Wonder* (Reference Recordings) for spatial effects, *Serendipidy* (Reference Recordings) for clean jazz sound, and *Simply Red* (Elektra) for vocal reproduction. On a popular Japanese three-way system, the records would sound flat with little dimension or depth and with the instruments appearing to hang on a clothes line between the speakers. On most competitive American speakers, the whole room would come alive.

SB: What was wrong with the Japanese speakers?

KJ: We were back to the problems with standing waves and breakup modes on woofers and hard domes.

SB: What was the next development step?

KJ: I worked out a set of designs in the Precise line. I specified the magnet weights, voice coil geometries, and cone materials.

SB: Did you have much problem in specifying the drivers?

KJ: No, that part was easy. It was old hat to me because I had taken so many speakers apart. The difficult problem was to correlate the Japanese measurements with what you hear.

SB: What measurements do they use?

KJ: They do the traditional ones—frequency response, group delay, and power response in a reverberant room. Those three measurements will tell you many things of how a speaker will sound. But you are looking for the little squiggles in the response where it is rolling off.

SB: When you look at a response, you typically see many ripples. How do you tell which ones are harmless and which ones will color the sound?

KJ: That is where instinct and knowledge come in. Given an 8-inch speaker, I know that the surround will take off at 200Hz. If it has a dome, it will take off at 3-4kHz. How fast the response rolls off will tell me the amount of phase shift that occurs and the type of crossover to use.

SB: Do you have a favorite crossover type, or is it dictated by the specific case?

KJ: It is dictated by the specific case. The rolloff of the driver tells you a little bit. Is it fast or slow? Are there resonant points? Where are the nodes? What does it sound like as a raw driver? All these factors play important roles. Then, you try to guess how the phase shifts will occur, because you want to treat how the acoustical crossover will work, not the electrical.

SB: Are there cases where the standard crossover circuits work?

KJ: Only when the drivers have a wider response than the crossover region. Unfortunately, in a practical system that is not likely to be encountered. After all, why sacrifice the bottom end of a woofer to give it two extra octaves of top end so that a classic crossover can be used? It is better to use the driver right to the hairy edge, get the best bass response, and then deal with a nonstandard crossover to make the system work.

SB: Do you have computer programs to generate crossovers, or is it done by cut and try?

KJ: It's both. I have charts I use. Onkyo has a very good computer program that was used to generate the best computer crossover match for each system. Then they actually built them up. That was our foundation. Then we started to tweak components to see what made audible changes.

So far with only one exception, the computer generated crossover has not been that close. The computer can't take into acount the eigensonics of the driver. The computer doesn't have a way of making subjective decisions.

SB: Computer people will not like to read that.

KJ: Well, you have to have the physics and engineering background to understand how things work, and the computer will give you a good start. Then you have to have the instinct from having done it many times before and seeing how drivers misbehave in ways you can't predict. The final point that makes a good audiophile system is the countless hours of listening, evaluating, and making subjective judgements.

SB: But can't you get yourself into little traps with subjective judgements?

KJ: Oh yes, but experience and sometimes creativity will keep you out.

SB: Do you try to have continuous phase transfer or constant group delay between drivers?

KJ: Group delay is what I like to have. The computer works that out quite nicely.

SB: Do you try to design for constant group delay with physical alignment or the crossover?

KJ: I do both because of the interplays between forward-backward alignment, baffle size, and crossover.

SB: In the Precise line, you start off with a series of two-way speakers, continue with a two-way with a passive, and finish with a three-way design. What was the philosophy that decided that product mix?

KJ: The primary one was market philosophy because you have to start with

PHOTO 6: The Precise Acoustic Laboratories line of speakers developed by Keith Johnson. The Monitor 3 is the small speaker to the left in the front. Next to it is the Monitor 5. In the back row from left to right is the Monitor 7, the Monitor 9, and the Monitor 10.

a product that people want and can afford. The Monitor 3 competes with the entry level speakers that many manufacturers make. But it is a serious loudspeaker in many respects and has had as much engineering work done on it as any of the other products.

SB: What are the main differences as you go up to the Monitor 5 and Monitor 7 models?

KJ: The Monitor 5 has essentially a bigger enclosure and a slightly different crossover balance. The tweeter has a larger magnet with a hollow pole to extend the lower frequency response. The 8-inch woofer has a bigger magnet and a more sophisticated moving structure. The Monitor 7 is essentially the same speaker as the Monitor 5 but in a bigger enclosure. The Monitor 9 is an 8-inch two-way with a passive radiator and larger woofer and tweeter magnets.

mounted passive radiator. It has a little bit less bottom end compared to a rear-mounted drone, but it has a tighter sonic character. Also, when people pick up a speaker, they tend to grab from the rear to the front and may inadvertently destroy a rear mounted drone.

SB: Why did you choose to make the Monitor 10, the top of the line, with a time-aligned® configuration?

KJ: Primarily, it was a dollar and cents decision. The Monitor 10 is a complicated system with many more piece parts. The Monitor 10 has an approximate time alignment with a more complicated crossover.

SB: What materials have gone into the cabinets?

KJ: All the speaker cabinets are made from a high density particle board (MDF). As one goes up the product line,

If you take a driver and mount it on just a baffle board, you will be amazed how much sound is transmitted via the baffle board as measured by an accelerometer. An easy way to control the panel vibrations is to cut a slot in the wood and fill it with compliant material.

SB: Are the enclosures sealed?

KJ: No, the whole line is ported or passively tuned.

SB: I don't see any ports on the front.

KJ: The port is on the back, primarily for cosmetic reasons.

SB: Did you run into any problems with the passive radiator on the Monitor 9?

KJ: You better believe we did. Passive systems are very difficult to design, of all the various types of tuning. But they are the simplest to make. The problems were in the various surround resonances in the woofer and passive radiator, cone-drone interactions, and standing waves in the enclosure. We have the development of the speaker now to the point where the listener won't hear any of the woofer-drone interactions.

SB: Is there a reason why the drone is on the front?

KJ: We wanted the drone to reinforce any wave front activity from the woofer. The speaker sounds better with a front-

there is also more sophisticated cabinet bracing.

SB: How did you decide where to put the bracing?

KJ: We drive the speaker and look for misbehavior with the drivers in place, or with a mini-speaker inside.

SB: How do you detect panel vibrations?

KJ: I detect panel resonances with an simple accelerometer. I move it around to find the most significant vibration peak. I've equalized the accelerometer output to compensate for panel radiation resistance and accelerometer frequency variations.

SB: The Precise literature describes slots in the front panel. Can you fill in the details?

KJ: As you can see on my pine box monitors, the baffle board is mounted on hard rubber. That's done because if you take a driver and mount it on just a baffle board, you will be amazed how much sound is transmitted via the baffle board as measured by an accelerometer. An easy way to control the panel vibrations is to cut a slot in the wood and fill it with compliant material.

SB: Where do you put the slots in the Monitor line?

KJ: The front panel is a separate piece from the rest of the enclosure, attached with a form of low-temperature hot melt glue that doubles as a glue and a damping gasket. We only use it on the two-ways because we want to damp out the objectionable cabinet resonances that usually show up in the crossover region. On the three-way, the driver range is more restricted, and we deal with vibration problems in different ways.

SB: Where is the crossover point between the midrange and the woofer on the Monitor 10?

KJ: I don't remember exactly, but it is in the neighborhood of 125–150Hz. The computer said that it should be 200Hz, but subjective listening moved it down.

SB: That low crossover point means that you must drive the midrange hard.

KJ: The mid is pushed hard. But it is a very good full-range reproducer at 6.5-inch diameter. At 125Hz and at high power levels, it is really moving! To keep the distortion down, the voice coil is moderately long. Yet it can't weigh very much because it has to extend into the high frequency range. So the only way to make it is to use thin wire in the voice coil and a very big magnet. It is not a very cheap driver.

SB: What about the midrange power handling capabilities?

KJ: The driver can handle high power levels. The voice coil uses a glass polyamide former which can withstand tremendous temperatures. We do that for speed of response because we've learned that if a speaker is fast with low mechanical loss, then you have a subjective jump factor in the bass that creates excitement. It's one of the areas that polypropylene cones do very poorly.

SB: So you believe in "fast" woofers?

KJ: Really fast woofers have large BL products that can give large push. That's an expensive woofer to make because you are not relying on the weight of the cone to keep it under control. You are relying on the motor structure and linear compliance to make it work. This type of woofer does take a bigger cabinet volume because you don't have the cone mass for low tuning.

SB: I've noticed that you like soft domes.

KJ: So far I've never heard a hard dome I've liked. The main problem is that they are too heavy and take too long to settle, which is audible.

Of course soft domes have compromises too. The soft dome, by its very nature, does not move all of its parts at the highest frequencies. At the upper cutoff, it becomes an annular radiator with the center making no contribution at all. Thus it's faster than a rigid system. So the soft dome comes closer to a ribbon driver which I like best.

SB: So would your ideal tweeter be a ribbon?

KJ: Oh yes, I can say absolutely from my experiences in recording sessions. But ribbons are very hard to make and use expensive magnets.

SB: After all your monitor speaker work, are you close to your ideal speaker?

KJ: It is close, but no potatoes as they say. The more I play with speakers and deal with the compromises of physics, the more I am convinced that you would like to have a driver that weighs nothing, has infinite stiffness, has different speeds of sound in it, and doesn't require any objects in front of it. If you could build a system with all those features, it would be a good foundation.

SB: Would your ideal system have only one driver?

KJ: Not necessarily, it would depend on what one could get away with. The ideal system would reproduce voice in a room that would sound like the person was there. That's the top priority. It also has to be able to shake the place and produce spatial effects.

SB: Getting back to the present, how has the digital revolution affected speaker design?

KJ: Digital recordings have affected speakers in both positive and negative ways. In digital recordings, you can have random phase all the way down to near DC. On an analog record the maximum energy at low frequencies has to be more or less in phase. With digital recordings, you can realize a more ethereal, large and powerful bottom end without channel-to-channel phase restrictions. An analog recording will capture the real bass aura or bloom better, since it doesn't let go or cut out in the final decay.

As you go up in frequency, digital recordings tend to advertise or separate various drivers. It made it very difficult to identify a good speaker from a not so good one, that is they can sound more alike. For example, cymbals sometimes have a phasy, hissy character as a result of time-delay modulation going on in the filter in your CD player. That phenomenon for a time discouraged work on tweeters, because if you build a better tweeter you have nothing to show for it. On an analog recording, you can hear a tweeter merge into the rest of the speakers, versus standing out and scratching at you as found with some digital recordings.

Most of the people who do digital encoding are now aware of the problem. And there are new encoders on the market that have decimation or symmetrical analog filters to correct the problem. The Apogee filter, for example will take away that brittle dull sound and will put more emphasis on building better tweeters.

A moving membrane speaker...made out of composite materials...worked out by complex computer programs.

SB: Where do you think loudspeakers will go in the near future?

KJ: I think more speaker systems will go the way that audiophile systems are heading. Manufacturers and designers are looking at new materials and ways to apply physics. Cones, domes, electrostatics and so on, have been around for a long time, and there is only so much you can do with them. I think the next really new breakthrough will be akin to achromatic lenses, in which the moving parts will be designed with CAD techniques. It is possible to design a moving membrane speaker. It may be a strange device, made out of composite materials that can only be worked out by complex computer programs. Such a system may be an order-of-magnitude improvement in sonic capabilities, compared to what we have now. The complexity of that design would be on the order of designing a zoom lens. Sometime in the future if someone perceives a need for such a design, I think a very fine system will result.

SB: Keith, thank you for a very enjoyable conversation.

ACKNOWLEDGEMENTS

I thank Lucette Nicole for suggesting this interview and for her support and Fred Buechler for photography assistance.

AMBER:
A SHEATHED CONDUCTOR ESL

BY DAVID LANG

Early Greek philosophers used to rub the yellow resin, amber, on various surfaces to give them the power to attract other small objects. Our word, electricity, comes from the Greek word for amber (elektron). This is the earliest description I am aware of which describes the electrostatic phenomena. Amber seems an appropriate name for my first completely successful electrostatic loudspeaker (ESL).

My design is a method of ESL panel construction which fulfills several goals. These include excellent sonics, ease of construction, low cost, improved dielectric stability over previous designs, and finally, significant design flexibility. This ESL is not an entirely original design. I have drawn heavily on work by Janszen[1] Hermeyer[2] Sanders[3] Wagner[4] Strickland[5] and Malme[6]

This sheathed conductor design, suitable for the home constructor, has not been published previously. Briefly, its origin dates back to 1953 with Janszen and Hunt's elegant description of the sheathed conductor concept. A popular commercial tweeter was developed from Arthur Janszen's work. Recently, James Strickland of Acoustat Corporation modified Janszen's design and has manufactured a highly successful, full-range electrostatic driver. My design incorporates Janszen's "sheathed conductor" stators with a variation of Mr. Strickland's rigid stator supports.

The design is, in essence, a PVC insulated wire stator supported by a rigid plastic grid. The diaphragm is 0.25-0.5-mil Mylar film, mounted on a removable Plexiglas frame. This allows removal of the diaphragm element without risk of panel damage. This also facilitates experimentation with various diaphragm spacer thicknesses and diaphragm coatings. Furthermore, should the diaphragm ever be damaged, replacement is extremely simple.

The associated electronics for my electrostatic loudspeaker system are very similar to the Sanders system. However, these panels work quite well with stock

ABOUT THE AUTHOR

David Lang, 29, is an orthopedic surgeon with a longstanding interest in high-fidelity sound reproduction and acoustics. His current passion is ESL theory, construction and design.

1. Janszen, Arthur, R.L. Pritchard, and R.V. Hunt, "Electrostatic Loudspeakers," Office of Naval Research; #NR-014-903, 1950.
2. Hermeyer, David and Roger Sanders, *Audio Amateur Loudspeaker Projects*, Marshall-Jones, 1985; specific articles.
3. ibid.
4. Wagner, Ronald, *Electrostatic Loudspeaker Design and Construction*, Tab Books, 1987.
5. The following patents have provided a significant source of construction techniques and understanding of the principles involved in electrostatic loudspeakers. The most relevant patents include #4323736, 4324950, 3931469, 3135838, 3136867, 3160715, 3345469, 3389226, 3992585, 3773984, 4316062, 3935397, 3941946, 4105877, 4160882. Janszen's patents are most relevant to the sheathed conductor design, #2631196, #2896025 and #3014098. All patents are available through the United States Department of Commerce Patent and Trademark Office, Box 9, Washington, DC 20231; $1.50 each. Alternatively, most large libraries have a patent library where you might be able to photocopy the patents.
6. Malme, Charles, "A Wide Range Electrostatic Loudspeaker," *JAES*, January 1959.

or second market "Acoustat" type biformers, or direct-drive tube output stages.

DESCRIPTION. All the materials you need to construct the sheathed conductor ESL panel are readily available through home improvement centers and electronic supply houses (see list of suppliers and parts at the end of this article). The construction sequence begins with the wire stator grids. Two are needed per panel. The diaphragm-frame assembly will then be described. Finally, I present a suitable enclosure and electronics. However, you should not limit your imagination or building endeavors to the details which I provide. The suggested dimensions were decided upon only for my available space.

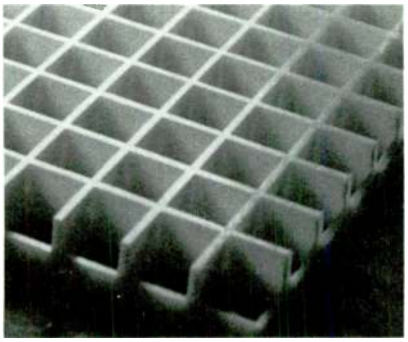


PHOTO 1: "Egg crate" material used for stator supports.

SHEATH CONDUCTOR STATORS. Rigid support for the stators is provided in part by two sheets of fluorescent light screen intended for use with suspended ceilings. The material is commonly known as "egg crate," available at most home improvement or building supply centers for $5–6 per 3 by 4-foot sheet. One sheet is enough for each ESL panel.

The egg crate is approximately ½-inch thick and has a grid-like structure composed of ½-inch square openings (*Photo 1*). Be sure you purchase all your egg crates from one supplier. I have encountered slight variations in specifications between brands which would lead to mismatched panels.

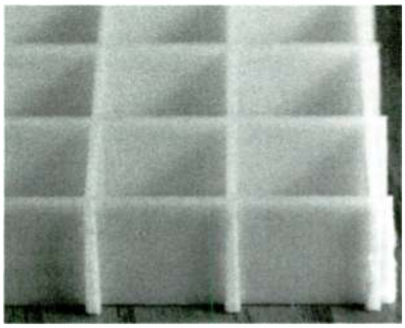


PHOTO 2: Cut egg crate with 1-2mm of overhang at edges of the panel.

The stators can be made any size but I suggest trying my dimensions first. The sound is incredible and the investment is minimal.

Note that your grid panels are tapered. That is, a definite inside and outside exist. Note that the cross section of a single slat in the grid is roughly trapezoidal. Be sure to lay out your grid panels so that the wider opening in the grid faces outward on both the front and back panel stator grids (*Fig. 1*).

Carefully cut out the egg crate with a large diagonal wire cutter, leaving 1–2mm of overhang (*Photo 2*). So all stator wire ends exit from the same end of the panel, the stator width must be an even number of squares. I will describe this in detail later. My stator grids are 18 by 74 squares, approximately 10 by 42 inches.

Decide which side of each grid is going to be the outside and where you will place your mounting screws, and mark these squares with a permanent marker on the side which will face the diaphragm. Make sure the marked squares match in location on all egg crate panels. Finally, file the edges of the egg crate smooth and sand off injection points on the side which will face the diaphragm.

I made the mounting holes for the stator grids by filling individual grid squares with epoxy cement, and then, when dry, I drilled holes to receive nylon bolts, which will pass through the entire thickness of the ESL panel. Tape off the grid squares, designated to form mounting holes, on the front side of the stator grid (*Photo 3*). Lay the stator grid, outside surface down, on a smooth surface covered with wax paper. Using any medium-time curing epoxy (Devcon works well), fill each marked square approximately half full. This is easily done using a 20 or 30cc Luer tip syringe to facilitate filling the squares. Anticipate using ½ to 1cc of epoxy per square. Allow approximately two hours for the epoxy to dry. Peel off your masking tape from the grid as soon as the epoxy is cured, since it is extremely difficult to remove once the epoxy is fully hardened (*Photos 4* and *5*). Note that the mounting squares in *Photos 3-5* are placed very close together for illustration purposes only. My system shows the actual spacing (see *Photo*).

PHOTO 4: Inside of stator support showing detail of epoxy-filled square.

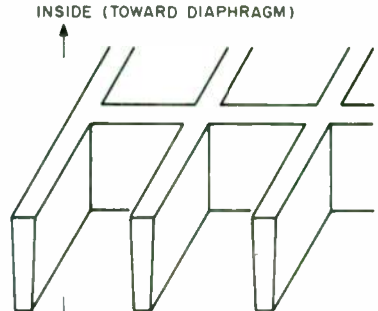


FIGURE 1: Egg crate tapered grid panels.

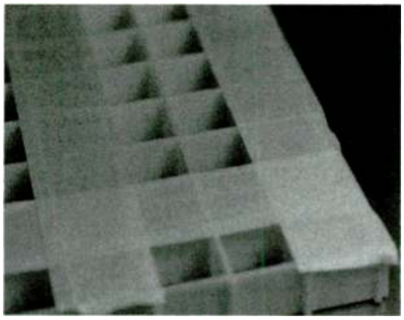


PHOTO 3: Mask off squares which will be used as mounting holes (note two strips of tape are closely-spaced for illustration).

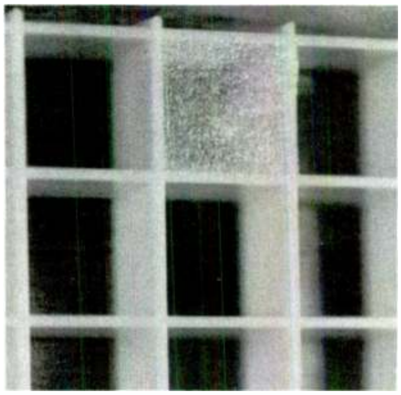


PHOTO 5: Outside detail of epoxy-filled square prior to drilling mounting holes.

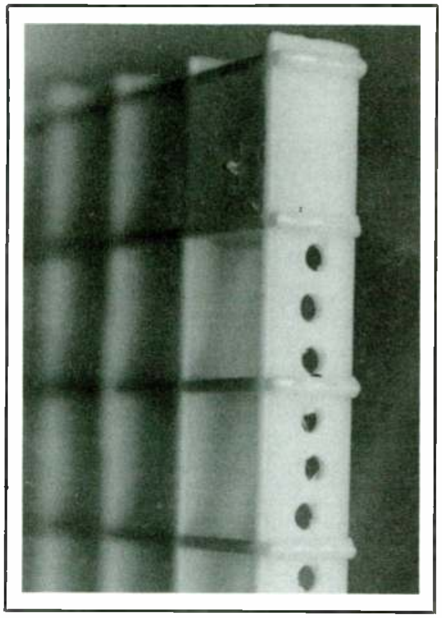


PHOTO 6: Drill holes at each end of the stator to receive mil-spec wire.

Next, drill holes at each end of the egg crate grids to accept mil-spec wire. Drill three, uniformly spaced holes per egg crate square, as shown in *Photo 6*. Drilling is most easily done with a hand-held hobby grinder, such as a mototool fitted with a drill. Drill slowly; the heat generated will tend to melt the plastic. After drilling, smooth all holes free of plastic flashing with a small metal file. This step is necessary to protect the mil-spec wire insulation. Check your mounting hole squares. Be absolutely sure they match on all panels.

To complete the construction of the stator grid supports, drill the appropriate mounting holes, which will receive nylon mounting bolts, in each of the epoxy filled squares. Use this drilled egg crate grid to guide the drilling of all remaining grids to ensure the holes match.

WINDING THE STATORS. The holes drilled at the end of each egg crate grid accept mil-spec wire. This wire is manufactured by Belden and comes in 500 or 1000 foot spools, approximately $40 per 1000 feet. The mil-spec wire is critical for uniform PVC insulation thickness. Plain PVC insulated, hook-up wire may work, but I have not tried this type. Belden #9924 is appropriate according to specifications outlined by Janszen and Hunt.

Weave the stator in three or more sections. Use of shorter wire segments makes the winding easier. Furthermore, this allows for future modifications as described at the end of this article.

Starting from the bottom of each stator, wind your mil-spec wire as follows.

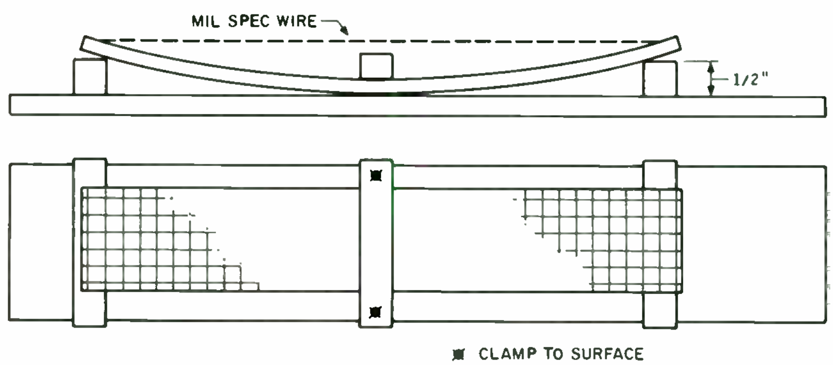


FIGURE 2: Bending jig for winding stator.

Unwind enough wire from your spool to cover ⅓ of one stator, approximately 60 feet. Tie a square knot one foot from the end of the wire. Enter the first hole with the free end of your wire and pass it through the corresponding hole on the opposite or top end of the egg crate grid, pulling the entire length of wire through the holes until your knot pulls tight against the grid's lower end. Then, while maintaining tension on your first pass, thread the free end of the wire back through the second hole at the top of the grid and exit the second hole at the lower end of the grid. Maintain tension on the first pass until the loop at the upper end locks in place. It sounds complicated, but it is really quite easy to do.

To remove slight kinks in the wire and ensure uniform wire tension, I built a tensioning jig to use while winding my stator grid (*Fig. 2*).

Cut three 1 by 1-inch wood strips, as long as your stator's width. Clamp each end of your stator grid firmly to your work surface. Lay the stator grid on these spacers and clamp the third spacer of the same size at its mid-point. This will place a bow in your stator. Wind your stator as noted above. The tension used should be just enough to straighten the wire but not enough to bow the stator grid further than it is bowed with the jig. Terminate and begin each section of your stator with a square knot. Be sure to leave one to two feet of wire exiting each section of your stator. This length will be needed for making connections. If you are unsure of your final cabinet design, leave two or more feet of wire at each end.

Release your stator from the bowing jig. Then, using the struts you used to clamp your stators in a bowed position, clamp the ends of your stator grid assembly to a flat, wax paper covered surface. This releases the bow and tensions the wires, and removes all minor wire kinks. The mil-spec wire should lie flat against the diaphragm side of the grids. *Be sure that you have wound your stators with the mil-spec wire on the inside of the egg crate grids.*

GLUING THE STATOR. This step requires the use of extra long setting epoxy (at least four hours working time) and several 20cc Luer tip medical syringes. Mix only 20cc of epoxy at a time to prevent costly waste. If you mix more, it

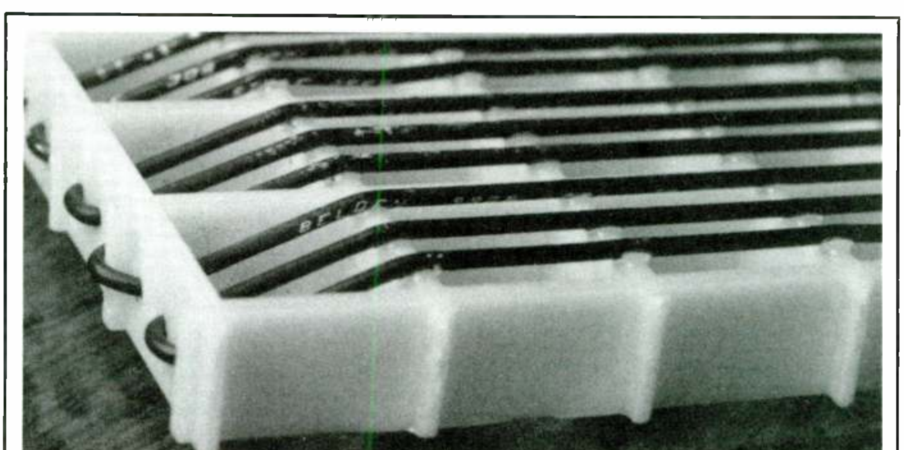


PHOTO 7: Wound sheathed conductor stator: be sure to leave room for nylon mounting screws between the wires. Alternately, do not wind wire into outermost grid squares—adjust panel width as needed for all wire ends to exit one end of panel.

tends to harden prematurely because of the heat generated during its curing. Double check your flattened stator.

Apply a thin bead of epoxy over *every spot where the wire and stator grids cross.* Failure to do this will cause the finished panel to rattle. This step is easily accomplished by applying a continuous bead of epoxy with the syringe, over the width of the stator following one of the stator grids. Any extra epoxy will run over the wire and into the grid squares.

The key at this point is to get a tight smooth joint between the wire and stator grid without any sharp bumps of epoxy facing the diaphragm. The epoxy will flow only if you use extra long setting epoxy (*Fig. 3*). Don't try to do too much with a single large epoxy batch; it is far easier and the results are more uniform with many small batches. Keep the stator clamped in place 24 to 48 hours to allow the slow working epoxy to set. You can remove any extra epoxy from the outside of the grid squares by gently chipping it out with a screwdriver once the epoxy is hardened. Unclamp your stators. You will see that they are now roughly flat. Be sure no epoxy "bumps" or sharp points face the diaphragm side of the stator.

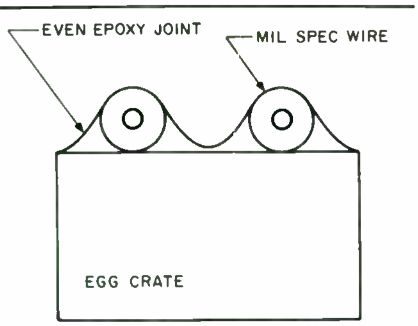


FIGURE 3: Stator, side view.

DIAPHRAGM-FRAME ASSEMBLY.

My design has one significant advantage: the diaphragm-frame assembly is easily removed or replaced. The frame is constructed of single thickness, one-tenth inch Plexiglas or lexan sheet. It functions both as a spacer and as an insulator for the diaphragm. Plexiglas is the preferred material as it is inexpensive and readily available. Be sure to cut all your frame pieces from a single piece of Plexiglas, since different sections often vary slightly in thickness. Using a single piece for each stator avoids this problem.

Cut and assemble the insulator frame first (*Fig. 4*). I have tried two methods of cutting the Plexiglas. A carbide tip saw blade on a table saw works best. It chips the edges slightly but this is not critical.

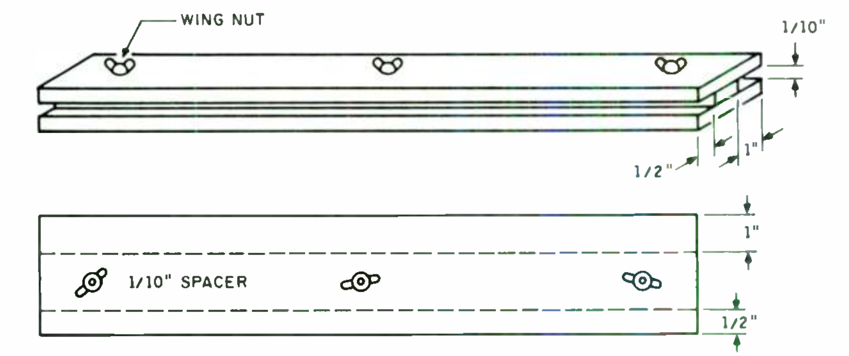


FIGURE 5: Slotted Plexiglas breaking guide.

The sharp edges can be sanded with 240-grade sandpaper.

If you lack a table saw, you will need a four-foot metal straight edge, a special Plexiglas scoring tool (Red Devil is best) and a long slotted breaking guide. The scoring tool is readily available. Plan on at least 25% waste with this method because of breakage. Make the slotted breaking tool from pine, four stove bolts and a 1/10-inch spacer (*Fig. 5*). The slotting tool works but requires practice. Score to a depth of at least one-third the thickness of the Plexiglas to ensure a uniform fracture line. Insert the scored side of your Plexiglas sheet into the appropriate slot of the breaking tool. Using one deft, firm snapping motion, you will break the Plexiglas along the scored line. With practice you should be able to break ½–1-inch widths with minimal waste.

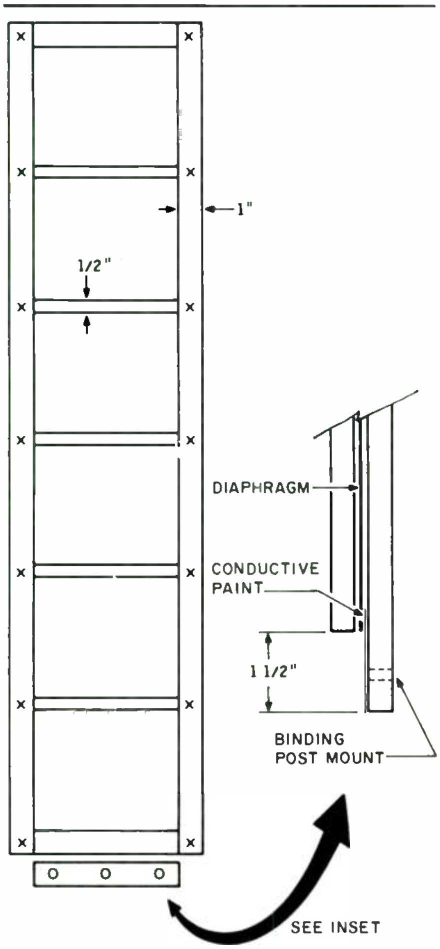


FIGURE 4: Diaphragm frame (dimensions can vary) and termination ends (inset).

After cutting all required insulator-spacer pieces, glue two spacer frames per ESL panel, according to *Fig. 4*. Plexiglas joints are best glued with polyester solvent, applied with an eyedropper. First, cover a flat surface with aluminum foil. Glue your diaphragm frames on the foil. Any gaps at the joints can be filled with baking soda and solvent. After gluing the diaphragm frames, sand the joints smooth. Drill your mounting holes in the diaphragm-frame using the stator holes as a guide. Drill your binding post holes at the same time.

MAKING MYLAR CONDUCTIVE.

You will need a piece of plate glass, two inches larger in all dimensions than your diaphragm frame, for hand rubbing Mylar with graphite. Edges of the plate glass should be sanded and the corners rounded by your glass cutter. It might be helpful now to review previous articles by Roger Sanders and David Hermeyer for a more detailed description of how to rub the Mylar with graphite. Fine powdered graphite works, but extra fine graphite works best.

Buy cotton balls at your local drug store. Clean your plate glass surface with glass cleaner and a lint free cloth. Vacuum the surface and spray lightly with 3M anti-static spray. You won't be able to eliminate all of the dust under the Mylar, but do remove all large particulate matter from the glass. Any large pieces of debris will tear the thin diaphragm when rubbing in the graphite.

Wear medical gloves when handling the Mylar. This will avoid fingerprints on its surface. Fingerprints don't take graphite and leave "dead" spots on the

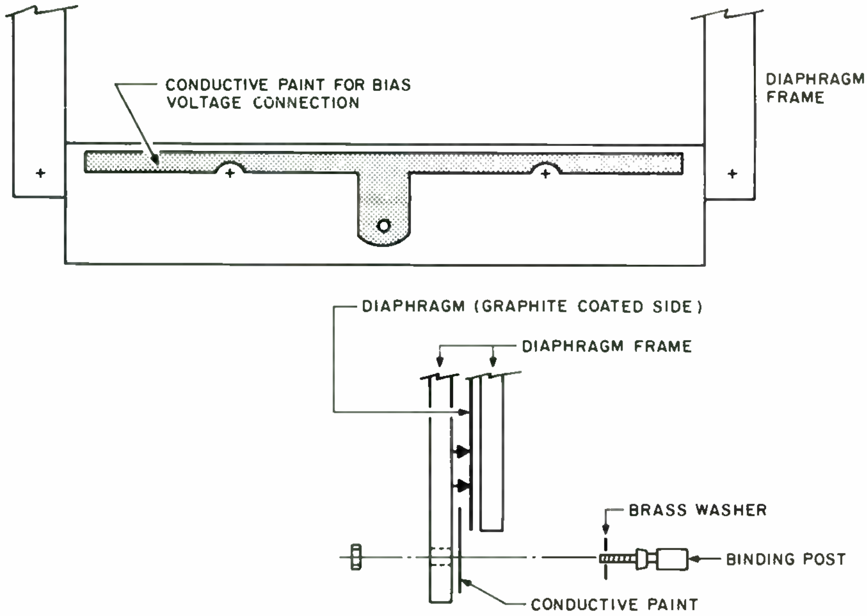


FIGURE 6: Conductive paint.

diaphragm. Cut an appropriate size piece of Mylar from your stock and tape it to the plate glass with masking tape. The Mylar should be taut to eliminate large wrinkles prior to applying the graphite. A stretcher frame is not necessary.

Next, apply a small amount of graphite to the Mylar. Spread it evenly over the entire surface. Using a cotton ball and gentle but firm pressure, rub the graphite into the Mylar using smooth circular motions. Avoid scrubbing or wiping in wide sweeps. Remove any large particle from between the glass and the Mylar that significantly tents the Mylar. If the Mylar tears, don't worry—that's why you ordered extra. Just try again. Remember ESL building is fun!

I use the Hermeyer method to determine when the appropriate amount of graphite has been applied to the Mylar. Get out your vacuum-tube voltmeter (VTVM) with alligator clips on test leads. Grip a new penny in each lead and measure the resistance at various points along the diaphragm with the pennies spaced one half inch apart. Typical resistances will measure from 15-500k. Rub extra graphite on the area that will provide the high voltage bias connection.

As an alternative to rubbing on graphite powder, one fellow ESL builder recommends EMI/RFI graphite spray, available from G.C. Electronics. It is a graphite emulsion similar to paint, which sticks well to Mylar and its application is controllable. I have no experience with this product but I plan to try it soon.

GLUING THE DIAPHRAGM-FRAME ASSEMBLY. First decide which side of each frame will face the diaphragm and mark them accordingly. Vacuum all excess graphite and wipe the mylar surface with a lint-free cloth. Turn your Mylar diaphragm over and tape it in place on the glass, eliminating all wrinkles. Be sure to lay the Mylar diaphragm graphite rubbed side down on the glass.

Now apply a ⅛-inch continuous bead of long working time epoxy around the entire diaphragm frame and insulator strips. Place the bead of epoxy in the center of the ½-inch strips and ¼ inch from the inner edge of the outside of the frame. Lay the diaphragm frame epoxy side down on the Mylar. Weigh it down with a spare piece of Plexiglas and 20-30 pounds of books. Allow the appropriate curing time for the epoxy you have decided to use.

Then, release the weight and lift the diaphragm frame assembly carefully and turn it with the Mylar side facing up. If you have glued the appropriate side, you should have an even graphite-rubbed Mylar surface facing you. Trim the Mylar to conform to the limits of the epoxy which has spread out over the spacers. Any free edge of Mylar not cemented will rattle if not trimmed.

Using silver or nickel conductive paint (see Parts List), paint a strip approximately ¼-inch wide along the width of your diaphragm along the lower section between the Plexiglas strips. Use *Fig. 6* to guide your painting. This conductive strip of paint will provide a means of attaching the diaphragm to your electrical connections from the periphery of the stator. After the paint has dried, glue the matched insulator spacer frame to the graphite coated side of the Mylar diaphragm with epoxy cement and allow the cement to cure.

If you have not done so already, drill mounting holes around the periphery of the spacer diaphragm assembly which will correspond to the mounting holes in your stator grids. Do this very carefully taking care not to crack the epoxy joints you have just made. Drilling these holes ¹⁄₁₆-inch larger than your nylon bolts makes assembly of the ESL simpler.

FINAL ASSEMBLY OF THE ESL PANEL. Pass one nylon bolt through each of the stator grid mounting holes and lay the stator grid on a table, wire side up. Next, carefully place your diaphragm spacer assembly on the wire stator, passing the nylon bolts through the drill holes; and finally, place the back stator, wire side down, over the bolts. The entire assembly should fit together like a sandwich (*Fig. 7*). All of your stator and diaphragm mounting holes should line up.

Then, thread the appropriate size nylon nut over the protruding ends of the

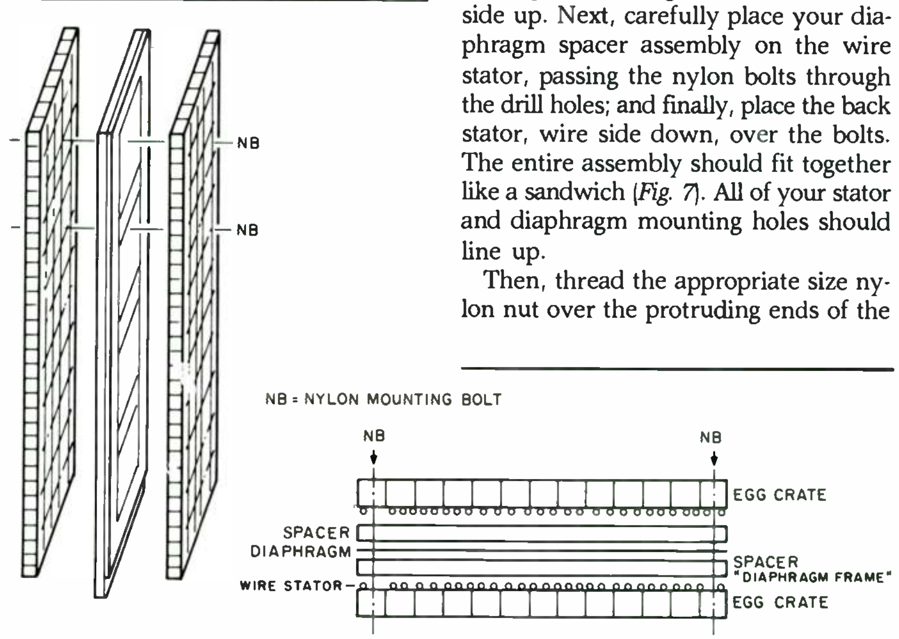


FIGURE 7: Final assembly.

nylon bolts and tighten them finger tight. Be very careful not to overtighten the panels; this might damage the insulation of the wires. Next, connect all of the wires wound on one side of your panel, that is, one stator, and connect them to a single termination of your choice (*Fig. 8*). A similar termination should be used for the opposite stator connections. To avoid stray capacitance, cut your stator wires only as long as necessary to make the terminations. Five-way gold-plated binding posts work well.

To complete construction of your electrostatic sheathed conductor panel, you must shrink the diaphragm to an appropriate degree of tautness. This is most easily done with a 1,000W hair dryer or heat gun. I have found it easier, with most consistent results, to shrink the diaphragm after the ESL is assembled. You could however, decide to stretch the diaphragm with the diaphragm frame assembly separated from the sheathed conductor stators. The stretching is completed by carefully warming the diaphragm with your hair dryer.

Observe the surface of the Mylar diaphragm with a light reflecting on its surface. As the Mylar begins to shrink you will note the very small wrinkles disappear and a perfectly flat, shiny surface will result. Heat shrink only a small portion of your stator at a time to assure uniform results. Take extreme care not to overapply heat in a concentrated area. The plastic egg crate grid could soften or melt if overheated. The Mylar appears to be fairly heat resistant and can be reshrunk many times without ill effects.

After heat shrinking your entire diaphragm, set it aside for ten to fifteen minutes to cool. Using a light reflecting on its surface, recheck the diaphragm for wrinkles that may have formed as the diaphragm cooled. Reheat and shrink the entire diaphragm as described above. This completes the construction of the basic electrostatic sheathed conductor panel.

ELECTRONICS. We can drive electrostatic loudspeakers in several ways. I refer you to a very simple drive system (*Fig. 9*), as described by Roger Sanders (*SB* 2/80). The high voltage parts are quite difficult to find, but are available through the suppliers listed at the end of this article. In selecting your step-up transformer for bias voltage, select one capable of very high voltage output. The electrostatic loudspeaker as described is capable of handling 7–10kV supplies without arcing. This enables the loudspeaker to deliver high sound pressure

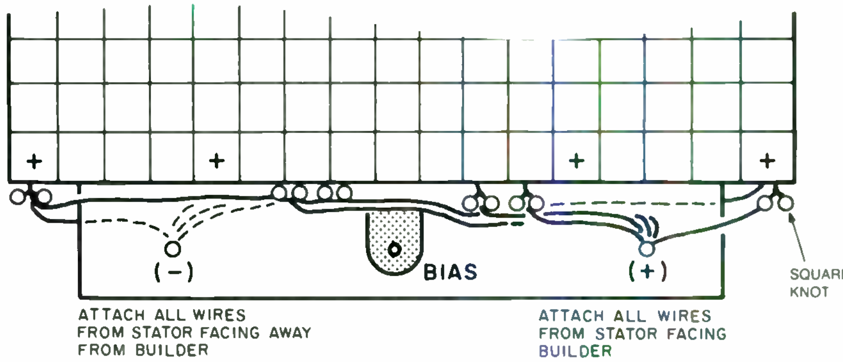


FIGURE 8: Electrical connections.

levels even with low voltage power amplifiers.

THE ENCLOSURE. The sheathed conductor electrostatic panels sound most transparent and preserve the best spatial imaging without enclosures. I have purposely avoided grille cloth and any obstruction at the front or back radiating surfaces of the panels. My enclosures consist of side pieces of hardwood, bolted through right angle aluminum extrusions to provide only a vertical support for the diaphragm. These supports are then bolted to a two-inch thick hardwood piece, which provides a rigid base. The design of any enclosure is entirely up to the builder but I strongly recommend you avoid any obstruction of the radiating surfaces of the diaphragm. The egg crate grids give a modular, modern appearance to the panels.

Place your electrical connections in a safe position away from your wood supports. These high voltages will conduct along a wood surface.

Spacing the loudspeakers four to five feet apart, "toed-in" slightly, and with the loudspeakers three to four feet from the nearest wall, provides the best imaging. Like all dipole radiators, these electrostatic loudspeakers are highly directional. Finding a satisfying position for the speakers requires some practice but these guidelines are a reasonable starting point.

One easy way to determine the best position for your electrostatic loudspeakers is to place a light bulb in front of you at the listening position and adjust the electrostatic panels until the light bulb's reflection is visible in both Mylar diaphragms. This produces a listening window approximately one person wide.

FULL RANGE OR BIAMPED. I agree with Roger Sanders that a full-range electrostatic loudspeaker entails significant compromise, or otherwise must be exceedingly large to provide high sound pressure levels at low frequencies. The results are far more consistent and superior sonics are more easily attained with biamping. The crossover frequency range is quite wide. I am currently using 200Hz as my cutoff point. I have also tried 450Hz, as Mr. Sanders suggests, and this appears to work well.

THE SOUND. I was quite startled at the quality of sound I heard through my sheathed conductor electrostatics. Without a doubt, the improvement in definition and imaging is the most significant that I have attained, and for the least amount of money. I highly recommend this project, as well as those others have presented.

FURTHER IMPROVEMENTS. My design is simple to build and has excellent sonic qualities. I have, however, explored several means of improving the sound quality and dispersion characteristics of the loudspeaker. My preliminary research entailed reviewing the work of Mr. Janszen and Hunt. As noted in the references, a variable width electrostatic loudspeaker array would be preferable for improved high frequency dispersion. That is to say, electronically decreasing the area, specifically, the width of the radiating diaphragm, with increasing frequency, should increase the dispersion of the loudspeaker at high frequencies.

Mr. Wagner, in his book, *Electrostatic Loudspeakers—Design and Construction*, elegantly reviews this topic and the calculation of the minimum width of the array for 180° of dispersion at any given frequency. I am currently designing a variable width, sheathed electrostatic loudspeaker with a theoretical dispersion of 180° that is a size similar to Mr. Wagner's suggested project. My current, although uncompleted project, is approximately two by four feet and theoretical-

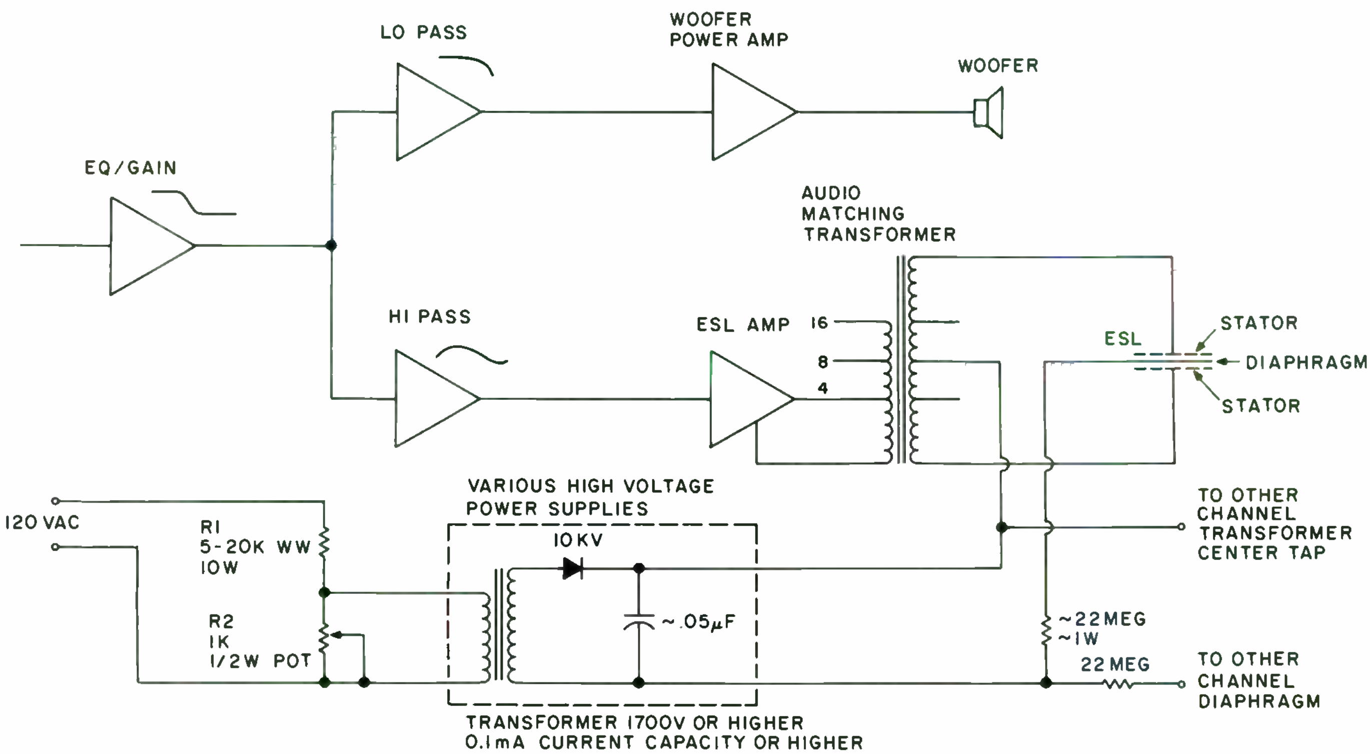


FIGURE 9: Associated electronics (per Sanders).

ly should radiate over 180° at all frequencies up to 17.5kHz.

Janszen and Hunt's article on the sheathed conductor theory touches on the design and reviews its advantages. The design entails a segmented diaphragm with segmented stator wiring which roughly doubles the radiating surface area of the diaphragm with every halving of frequency. The beginning width is ¼-inch at 17.5kHz.

A concern with segmented diaphragms is the potentially difficult load presented to any amp and stepup transformer. This area is probably worth pursuing despite its technical problems.

The reader may also wish to follow the Janszen and Hunt suggestions about improving ESL efficiency by constructing a diaphragm with a narrower stator-to-diaphragm interval. I have not experimented with this and am currently using a stator-to-diaphragm width of 1/10-inch. Mr. Wagner's text discusses this topic in detail.

FINALLY. I would like to call readers' attention to the electrostatic loudspeaker builder and users group, STATIC DISCHARGE. The purpose is an exchange of methods, ideas, tools and parts sources for those trying to build their own version of what they believe is the finest sound transducer possible. Since the group is new, it has no specific forum or agenda and activities have included only communication among members. Those interested should be willing to participate actively in the exchange of ideas and construction techniques, but extensive experience is not required. Those interested can contact the club, c/o Neil Shattles, 829 Glasgow Dr., Lilburn, GA 30247.

TABLE I

ASSOCIATED SYSTEM COMPONENTS

Nikon trapezoidal subwoofers
Magnavox CDB-650—with POOGE-4 modifications
Hafler DH-101 preamp—POOGEd
Old Colony/Jung two-way electronic crossovers with outboard power supply
Hafler DH-120 subwoofer amp
Nelson Pass A-40 ESL amp

PARTS SOURCES

I have listed only the parts not available at typical hardware or home improvement centers.

Newark Electronics
500 N. Pulaski
Chicago IL 60624
(312) 784-5100.
(PVC insulated mil-spec wire: #9924 from Belden Wire Corporation)

Hillman Fasteners
6156 Russellman Road
Cincinnati, OH 45248
1-800-543-1332.
(nylon bolts and nuts)

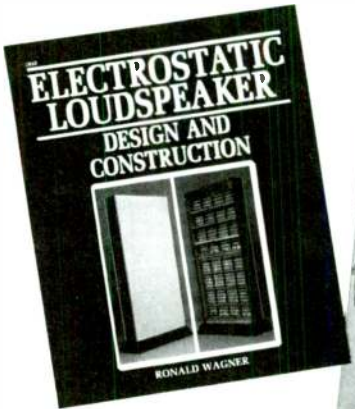
Devcon Corporation,
Danvers, MA 01923.
(epoxy adhesive)

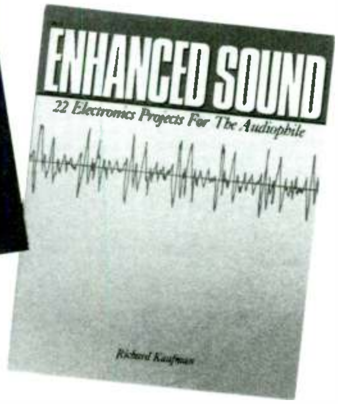
G.C. Electronics
available through Newark Electronics or large electronic supply houses.
(nickel conductive paint, part #22-207)

G.C. Electronics
(EMI/RFI spray graphite coating;, part #10-48-07)
Graphite coatings are available at your local hardware store.

Large plastic wholesalers:

Midland Plastics Inc.
Brookfield, WI
(414) 781-6520.
(Mylar film, 0.25 mil)
Note: 0.25-mil Mylar is difficult to find; 0.5-mil is an acceptable alternative. A possible alternative to Mylar are the Dupont Clysar shrink films, available through Dupont.





TAB BOOKS

T-5 IC VOLTAGE REGULATOR SOURCEBOOK WITH EXPERIMENTS *by Vaughn Martin.* This sourcebook provides the knowledge needed to fully understand, design and service modern power supplies. It introduces the components found in all power supply designs, transformers, diodes and capacitors as well as concepts like AC, DC and filtering. The book covers bipolar and MOS transistors, ICs used in regulators, switch mode designs and output indicators. Diagrams, data sheets and illustrations. 256pp., softbound. Each **$14.95**

T-6 ELECTROSTATIC LOUDSPEAKER DESIGN AND CONSTRUCTION *by Ronald Wagner.* An exceptionally well written and illustrated builder's book. 256pp., softbound. Each **$15.95**

T-7 BASIC ELECTRONICS COURSE [2ND ED] *by Norman H. Crowhurst.* This highly recommended textbook defines literally every aspect of electrical behavior and state-of-the-art electronics. Included are fundamental building blocks of theory and practice such as Kirchhoff's Laws, properties of resistance, electron flow, magnetic fields and power calculations. 430pp., softbound. Each **$15.95**

T-8 ENHANCED SOUND—22 ELECTRONICS PROJECTS FOR THE AUDIOPHILE *by Richard Kaufman.* Twenty-two cost efficient projects are offered which can increase the capabilities of your audio system, and improve its performance. Discussions on theory, design applications and construction techniques are also included. A sampling of the projects...surround sound decoder, passive image enhancer, auxiliary input switch, tone controls, amplifier bridging circuit, modern infrasonic filter, Linkwitz-Riley crossover, improved dipole, helical antennas, and more. Over 100 illustrations. 1988, 170 pp., softbound. Each **$10.95**

McGRAW-HILL

MH-1 HANDBOOK FOR ELECTRONICS ENGINEERING TECHNICIANS [2ND ED] *by M. Kaufman and A.H. Seidman.* A comprehensive compendium of electronic facts. Component selection, circuit analysis, power supplies, IC uses and characteristics, op amps, transistors, batteries and tubes. A one-volume encyclopedia on how components work and how to choose the best of them for your application. Includes seven new chapters on topics such as logic analysis, fiber optics, active filters, digital test gear and microprocessors. 752pp., hardbound. Each **$45.95**

MH-2 HOW TO MAKE PRINTED CIRCUIT BOARDS *by Joel Goldberg.* All the topics on making your own circuit boards are covered both in theory and in practical advice: design and layout, artwork preparation, photo layout, silk screening, and etching. Each **$8.90**

MH-3 LEARNING ELECTRICITY AND ELECTRONICS THROUGH EXPERIMENTS *by J. A. "Sam" Wilson and Milton Kaufman.* Using four perfboards and typical "junk box" parts, this book demonstrates theoretical principles in 54 circuits you can build as you learn. Each **$10.95**

NATIONAL SEMICONDUCTOR

N-5 VOLTAGE REGULATOR HANDBOOK. A manual which simplifies selection of three-terminal regulators depending on your needs, and which offers information on heatsink selection and design, along with transformer and filter specifications. 312pp., softbound Each **$7.00**

N-6 LINEAR DATA BOOK 1. First book in a comprehensive 3-volume set of specifications and applications. Topics covered, with example circuits, include voltage regulators, op amps and buffers, voltage comparators and instrumentation amplifiers. 1988 Edition. 1,250pp., softbound. Each **$16.95**

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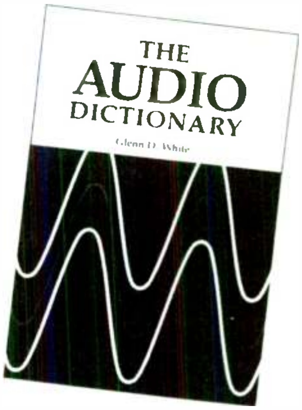

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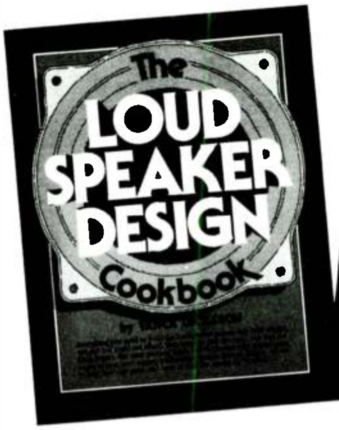

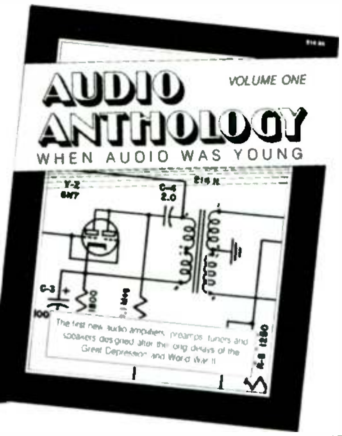

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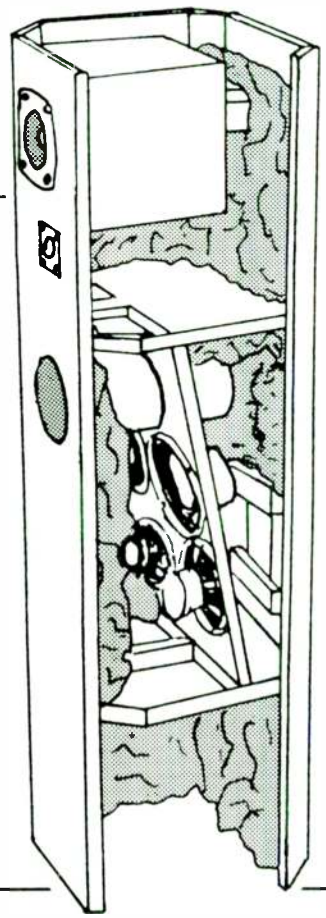
Part I

THE THIRD DIMENSION: SYMMETRICALLY LOADED

BY JEAN MARGERAND



Many articles in this magazine have been devoted to the design of closed or vented systems. But among other possible bass driver loadings, the symmetrically-loaded system, or acoustic super woofer (ASW), is less popular. Although this principle appeared in the US during the fifties, it was not widely applied.

Twenty years later, the French firm, Elipson, a specialist in resonant compartments systems, resurrected this type of loading. Very recently, KEF, known for well-designed bass sections, revived this technology with its model 104-2.

Now I will present reasons to show this type of loading is also a good choice for home builders.

In building systems, you are probably looking for an innovative system which improves the performance of your home system. For your first project you may have chosen a simple closed-box design. It's simple to build, and once you choose the driver, other than the damping Q, you can modify only one parameter: the volume.

Perhaps you progressed to building a vented system, allowing the determination of two parameters: the enclosure volume and the vent tuning frequency. According to Small, you also have a third parameter, the leakage factor, Q_L.[1]

I propose a box with which you can have even more fun. With the symmetrically-loaded system, you may play with three parameters, two compartment volumes, and the vent tuning frequency (see *Fig. 1*). As a result, you can now build a complete bass-frequency response with a natural acoustic, high cutoff frequency.

In this article, I review the essential theories you will need to design such an enclosure. I also provide simplified design formulas and alignment tables to help you with day-to-day design work. Finally, I computed parameters for 27 different drivers for possible use in symmetrically-loaded systems.

I have applied the results to build Delta, a three-way, mid-sized 56-liter enclosure, equipped with four 4.8-inch diameter bass drivers, mounted push-pull, in a slim, esthetic column. [*Look for the full construction details of Mr. Margerand's system in Part II, next issue.—Ed.*]

ABOUT THE AUTHOR

Jean Margerand, 33, graduated from Ecole Nationale Superieure Des Arts et Metiers, Paris and also has a marketing degree from Lyon. He is employed by Electricite de France, and has been building loudspeakers as a hobby for 15 years.

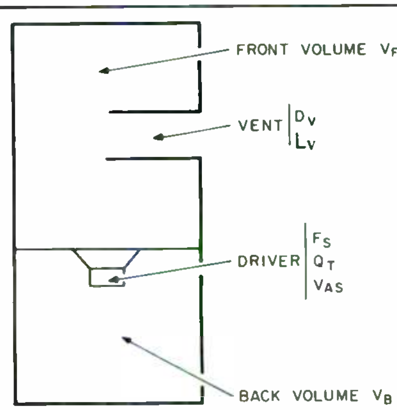


FIGURE 1: Principal scheme.

BACKGROUND THEORY. In 1982, two Frenchmen, Philippe Augris and Dominique Santens, first proposed a theoretical study of this loading type (see *Fig. 2*). Original by their application, it is directly derived from the Thiele and Small approach. From a solid theoretical basis, they built an extremely simple mathematical model allowing every amateur to design and construct such an enclosure.[4] Let's examine this theory.

In this model the leakage through the enclosure walls is assumed to equal zero, and the damping effect is ignored.

In a symmetrically-loaded system, the bass driver is located inside the enclo-

1. Small, R.H., an anthology of articles on loudspeakers from *JAES*, Vol. 1-25, pp. 271-303, 316-343.

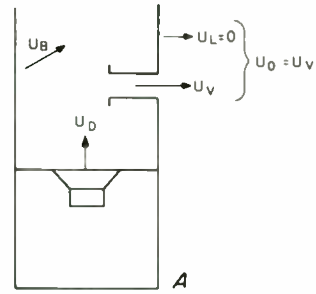
2. Marec, Guy, "Mise au point des filtres separateurs passifs—Du reve a la realite," *L'Audiophile*, No. 16, pp. 27-33.

3. Marec, Guy, "Mise au point des filtres separateurs passifs—Du reve a la realite," *L'Audiophile*, No. 18, pp. 15-23.

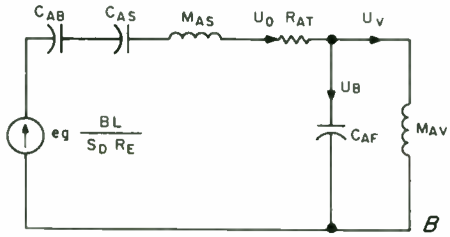
4. Augris, P. and D. Santens, "Optimisation des enceintes a charge symetrique," *L'Audiophile*, No. 23, pp. 47-54.

5. Knittel, Max, "Impedance Compensating Crossover, *SB* 1/83, p. 11.

Hypothesis:
- neglected leakage $U_L = 0$
- vent and cavities neglected damping



Electrical model:

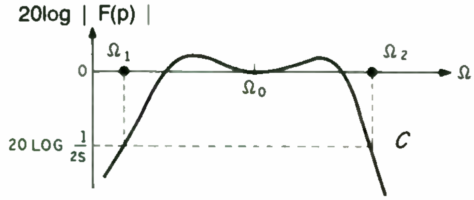


Back cavity: C_{AB}
Front cavity: C_{AF}
Vent: M_{AV}

Driver: C_{AS}, M_{AS}, R_{AT}, R_E, S_D, Q_{ES}, B_L
Amplifier: e_g

Transfer function F(p) and relative acoustic pressure: P_A/P_o (dB):

20log | F(p) |



$$F(p) = \frac{B^2(p/\Omega_o)^2}{D(p)} \quad \text{with:}$$

$$p = j\Omega,\ \Omega = 2\pi F,\ \Omega_o = 2\pi F_o$$

$$\Omega_1 \Omega_2 = \Omega_o^2,\ B = (\Omega_2 - \Omega_1)/\Omega_o = 1/2s/Q'_T$$

$$D(p) = (p/\Omega_o)^4 + 2sB(p/\Omega_o)^3 + (B^2 + 2)(p/\Omega_o)^2 + 2sBp/\Omega_o + 1$$

$$(P_A/P_o)_{dB} = 20\log \left| \frac{p}{2\pi\Omega\, P_{ref}} \times \frac{e_g\, B_L}{S_D \times R_E} \times \frac{1}{M_{AV}} \times F(p) \right|$$

$$\text{with} \quad \frac{p}{2\pi r\, P_{ref}} = \frac{1.18}{2\pi \times 1 \times 2 \times 10^{-5}} = 9400$$

Efficiency: $\eta = P_A / P_E$ with P_A: acoustic power; P_E: electrical power

- closed or vented enclosure: $\eta_o = \frac{4\pi^2}{C^3} \times F_s^3 \times \frac{V_{as}}{Q_{ES}}$
- symmetrically-loaded enclosure: $\eta_1 = \frac{1}{B^4} \times \eta_o$

Sensitivity (in acoustic pressure)

- closed or vented enclosure: S_o
- Symmetrically-loaded enclosure: $S_1 = \frac{1}{B^2} S_o$

Diaphragm displacement:

$$x_D = P_E^{1/2} \times \varsigma_x \times K_x \times X(p) \text{ with } P_E = e_g \times R_E/(R_g + R_E),\ K_x = \frac{V_B}{V_{as} + V_B} < 1$$

$$\varsigma_x = \left[\frac{V_{AS}}{2\pi\rho_o C^2 F_s Q_{ES} S_D^2}\right] \quad X(p) = [1 + (P/\Omega_o)^2] / D(p)$$

FIGURE 2: Mathematical model for symmetrically loaded system.

sure. Its backside is loaded by a closed volume V_B, and its front face by a vented compartment, V_F (*Fig. 1*). This system, tuned by the vent, acts as a fourth-order, pass-band filter. The slopes of the filter on each side are 12dB/octave and the frequency response is absolutely symmetrical in a logarithmic frequencies axes; hence, the name. The frequency response of such a filter is characterized by the following parameters.

- S—damping factor of the system, $0 < S \leq 1$.

It is related to the maximum modulation (ϵ) in the reproduced band (see *Table 1*).

- F_0—central frequency of the pass-band filter. This frequency is reproduced without any attenuation.
- ($F_1 < F_2$)—these two frequencies are located on each side of F_0, such as $F_1 \times F_2 = F_0^2$. For these two frequencies the output level is equal to 20 log $1/(2 \times S)$, compared to the level of the signal at F_0; except in the case where $S = 0.7$, F_1 and F_2 are different from the -3dB cutoff frequencies F_L and F_H.

Figure 3 gives the different frequency response shapes according to S value. Practically, for reasons I will give later, I have chosen the values of S between 0.4 and 0.7. For a given driver, the response of this fourth-order filter depends on three construction parameters, V_F, V_B, and F_B, the vent tuning frequency. In fact, theory proves that V_F and F_B are coupled. Modifying one or several of these parameters allows the speaker builder to change the frequency, the transient responses and the power capa-

TABLE I

S	Σ(dB)
.4	2.70
.5	1.25
0.6	0.35
0.7	0
>0.7	0

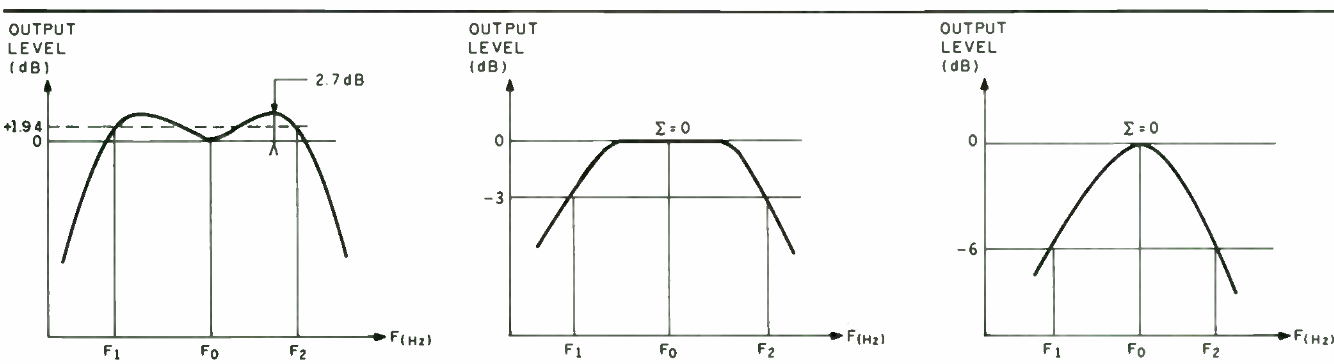


FIGURE 3: Frequency response according to S value.

bility of the system. I suggest we detail the influence of each parameter.

FRONT VOLUME. The front volume, V_F, is independent of the reproduced bandwidth. It depends only on the value of S and on the basic characteristics of the chosen driver according to the following formula:

$$V_F = (2 \times S \times Q_T)^2 \times V_{as}$$

where: Q_T: speaker mesh;
V_{as}: equivalent acoustic compliance air volume; V_F and V_{as} are expressed in the same unity (in liters, for example).

BACK VOLUME. You draw the frequency response and the system efficiency (η) from the back volume value. For a given value of S, you cannot fix in the same time and independently, the bandwidth characterized by $(F_H)_{-3dB} - (F_L)_{-3dB}$ and the efficiency. Moreover, as in the case of an infinite box where we can define Q_{TC}, in the same way we can define here Q'_T, the total suspension factor of the driver loaded by the back volume. The formula giving Q'_T is:

$$Q'_T = Q_T \sqrt{1 + \frac{V_{as}}{V_B}}$$

We can draw the back volume:

$$V_B = \frac{V_{as}}{\left(\frac{Q'_T}{Q_T}\right)^2 - 1}$$

V_B and V_{as} are expressed in the same unity, in liters, for instance. Q'_T must be greater than Q_T. Practically, we cannot choose a value of Q'_T from Q_T too near, or V_B will grow too large. At the limit, if $Q'_T = Q_T$ then V_B is infinitely great. Then the system would be equivalent to a classic vented box, where the acoustic output would issue from the vent only.

VENT TUNING FREQUENCY. The theory[4] shows that:

$$F_0 = Q'_T \times (F_S/Q_T)$$

Furthermore, the tuning of the system is realized when F_B, the vent tuning frequency related to the front volume, V_F, is equal to F_0, the resonant frequency of the driver loaded by V_B, the back volume; so, $F_B = F_0$.

From the Helmholtz resonator theory, we draw the relation between the following parameters:

$$F_B^2 = \frac{3 \times 10^4 \times S_V}{L_V \times V_F} - 0.9\sqrt{S_V}$$

as:

$$F_0 = Q'_T \times \frac{F_S}{Q_T}$$

we have:

$$L_V^2 = \frac{3 \times 10^4 \times S_V}{V_F \times F_0^2} - 0.9\sqrt{S_V}$$

where:

S_V = vent section (cm²)
L_V = vent length (cm)
V_F = front cavity volume (liters)
F_B = vent tuning frequency (Hz)

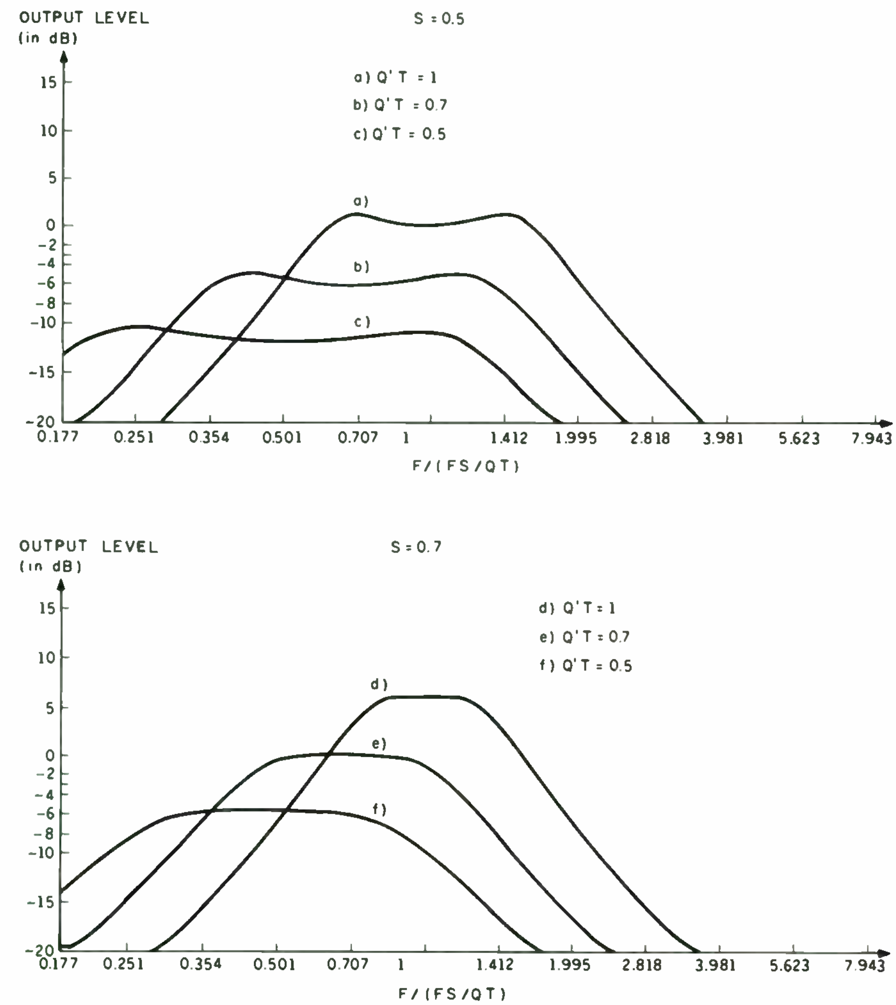


FIGURE 4: Frequency response according to S and Q'_T values.

REMARKS. We have just studied all the parameters. We can now see their interaction. Philippe Augris and Dominique Santens have drawn networks of frequency response curves (*Fig. 4*) according to various values of S and Q'_T. To keep the generality of the results, these curves give the value of the output versus the normalized frequency, $F/(F_S/Q_T)$. When you examine these networks of curves, you can see that:

1. Once you choose a value of S, the −3dB bandwidth is identical for any value of Q'_T. But the smaller Q'_T is, the more the bandwidth shifts toward the low frequency, but the efficiency also falls. The zero dB corresponds to the efficiency of a flat baffle mounted driver.

2. The bandwidth versus the normalized frequency, $(F_H - F_L)_{-3dB}/(F_S/Q_T)$ is a function of S according to *Table 2*.

If we choose smaller and smaller values of S, we have at the same time a wider reproduced band (for example, if S goes from 0.7 to 0.5, the bandwidth in-

TABLE II

S	$(F_H - F_L)_{-3dB}/(F_S/Q_T)$
0.4	1.7172
0.5	1.2712
0.6	0.9560
0.7	0.7206

creases by a factor of 1.76), more shifted toward the low frequencies, but also with a lower efficiency. Finally, remember that choosing a smaller value of S results in bandwidth modulations and also a worsening transient response.

3. The volume, V_I, of the front compartment is proportional to the square of S. Going from an S value of 0.5 to 0.7 increases the value of V_I by $(0.7/0.5)^2$ or 1.96. This is the price we pay to get better frequency response linearity and as a result, better transient response.

4. According to R.H. Small, the efficiency η_0 of a given driver is identical, whether it is mounted on a flat baffle, in an infinite box, or a vented box. This efficiency η_0 is only a function of the driver's basic characteristics. The symmetrically-loaded system is based on the same theory but with a slight difference. R.H. Small says:

> "If deliberate mass loading of the driver is employed in the system, e.g. placing a restricted aperture in front of the driver, the system reference efficiency will be less [or more] than the basic efficiency [η_0] of the driver."

Thus, you should not be surprised to find alignments characterized by a different efficiency than η_0 (greater or smaller).

PROS AND CONS. *Cutoff Frequency:* for some drivers for which the ratio F_S/Q_T and V_{as} are low, this type of loading may allow a lower cut-off frequency than the vented box for the same volume; no free lunch, though, since the sensitivity will fall.

Power Handling: this is greater than that of a vented box because the excursion for a given acoustic power is smaller, at least in the reproduced bandwidth. This is caused by the $\kappa\chi$ factor, which is lower for this loading than for the vented box (*Fig. 2*).

Asymmetrical Distortion: because the driver cone is loaded identically on its two faces, any asymmetrical distortion is diminished.

Simplified Filter: the high cutoff frequency, F_H, of the system is purely acoustical. More often, you can remove the inductor in series with the woofer which often damages the transient response. And, the moderate slope of the cutoff (12dB/octave) produces a correct transient behavior.

Efficiency: is not fixed and constitutes one of the design parameters of the system. The efficiency may be superior to the nominal efficiency η_0 of the driver.

Sensitivity Matching: since the extended low-frequency response results in a low efficiency, matching the woofer to a high sensitivity midrange driver may involve using bi- or triamplification to raise bass level. Furthermore, because the high cutoff frequency is between 100 to 200Hz, you will need to find a midrange driver which will reproduce these frequencies at an acceptable level. Finally, you cannot use this system for a two-way design.

Construction: enclosures for symmetrically-loaded systems are more sophisticated than for a closed or vented enclosure. But the internal partitions give us a bonus of increasing the cabinet's rigidity, and the drivers are more protected within the enclosure.

Diffraction: on the front face, you see only the 10 or 15cm diameter vent aperture, rather than the usual 25cm diameter bass driver of a conventional enclosure. So it is possible to design a very narrow enclosure with rounded edges by mounting the driver on an angled board inside the enclosure. Finally, this construction may reduce the number of parallel faces and the effects of standing waves.

DESIGNING A SYSTEM. To accurate-

TABLE III

ALIGNMENTS ACCORDING TO S & Q'$_T$ VALUES

(FH-FL)/(FS/QT)=1.7172 S=.4

Q'T	FL/(FS/QT)	FH/(FS/QT)	PA(dB)
0.7887	0.3073	2.0245	-8.0
0.8117	0.3230	2.0402	-7.5
0.8354	0.3394	2.0566	-7.0
0.8598	0.3565	2.0737	-6.5
0.8849	0.3744	2.0916	-6.0
0.9108	0.3931	2.1103	-5.5
0.9374	0.4126	2.1298	-5.0
0.9647	0.4329	2.1501	-4.5
0.9929	0.4541	2.1713	-4.0
1.0219	0.4761	2.1933	-3.5
1.0517	0.4991	2.2163	-3.0
1.0825	0.5230	2.2402	-2.5
1.1141	0.5479	2.2651	-2.0
1.1466	0.5738	2.2910	-1.5
1.1801	0.6008	2.3180	-1.0
1.2145	0.6288	2.3460	-0.5
1.2500	0.6579	2.3751	-0.0
1.2865	0.6881	2.4053	0.5
1.3241	0.7195	2.4367	1.0
1.3627	0.7521	2.4693	1.5
1.4025	0.7859	2.5031	2.0
1.4435	0.8209	2.5382	2.5
1.4856	0.8573	2.5745	3.0
1.5290	0.8950	2.6122	3.5
1.5737	0.9340	2.6513	4.0
1.6196	0.9745	2.6917	4.5
1.6669	1.0164	2.7336	5.0
1.7156	1.0598	2.7770	5.5
1.7657	1.1047	2.8220	6.0
1.8172	1.1513	2.8685	6.5
1.8703	1.1994	2.9166	7.0
1.9249	1.2491	2.9663	7.5
1.9811	1.3006	3.0178	8.0

(FH-FL)/(FS/QT)=1.2712 S=.5

Q'T	FL/(FS/QT)	FH/(FS/QT)	PA(dB)
0.6310	0.2600	1.5312	-8.0
0.6494	0.2731	1.5442	-7.5
0.6683	0.2867	1.5579	-7.0
0.6879	0.3010	1.5721	-6.5
0.7079	0.3158	1.5870	-6.0
0.7286	0.3313	1.6024	-5.5
0.7499	0.3474	1.6186	-5.0
0.7718	0.3642	1.6354	-4.5
0.7943	0.3817	1.6528	-4.0
0.8175	0.3999	1.6711	-3.5
0.8414	0.4189	1.6900	-3.0
0.8660	0.4386	1.7097	-2.5
0.8913	0.4591	1.7302	-2.0
0.9173	0.4804	1.7515	-1.5
0.9441	0.5025	1.7736	-1.0
0.9716	0.5255	1.7966	-0.5
1.0000	0.5493	1.8204	0.0
1.0292	0.5741	1.8452	0.5
1.0593.	0.5997	1.8709	1.0
1.0902	0.6264	1.8975	1.5
1.1220	0.6540	1.9251	2.0
1.1548	0.6826	1.9537	2.5
1.1885	0.7122	1.9833	3.0
1.2232	0.7429	2.0141	3.5
1.2589	0.7747	2.0458	4.0
1.2957	0.8076	2.0788	4.5
1.3335	0.8417	2.1128	5.0
1.3725	0.8769	2.1481	5.5
1.4125	0.9134	2.1845	6.0
1.4538	0.9511	2.2222	6.5
1.4962	0.9901	2.2612	7.0
1.5399	1.0304	2.3015	7.5
1.5849	1.0720	2.3431	8.0

(FH-FL)/(FS/QT)=0.9560 S=.6.

Q'T	FL/(FS/QT)	FH/(FS/QT)	PA(dB)
0.5258	0.2326	1.1886	-8.0
0.5412	0.2440	1.2000	-7.5
0.5570	0.2560	1.2119	-7.0
0.5732	0.2684	1.2244	-6.5
0.5900	0.2813	1.2373	-6.0
0.6072	0.2948	1.2508	-5.5
0.6249	0.3088	1.2648	-5.0
0.6432	0.3233	1.2793	-4.5
0.6619	0.3385	1.2945	-4.0
0.6813	0.3542	1.3102	-3.5
0.7012	0.3706	1.3266	-3.0
0.7216	0.3876	1.3436	-2.5
0.7427	0.4052	1.3612	-2.0
0.7644	0.4235	1.3795	-1.5
0.7867	0.4425	1.3986	-1.0
0.8097	0.4623	1.4182	-0.5
0.8333	0.4827	1.4387	0.0
0.8577	0.5039	1.4599	0.5
0.8827	0.5258	1.4818	1.0
0.9085	0.5486	1.5046	1.5
0.9350	0.5721	1.5281	2.0
0.9623	0.5965	1.5525	2.5
0.9904	0.6217	1.5778	3.0
1.0193	0.6478	1.6039	3.5
1.0491	0.6749	1.6309	4.0
1.0797	0.7028	1.6588	4.5
1.1113	0.7317	1.6877	5.0
1.1437	0.7616	1.7176	5.5
1.1771	0.7925	1.7485	6.0
1.2115	0.8244	1.7804	6.5
1.2469	0.8573	1.8134	7.0
1.2833	0.8914	1.8474	7.5
1.3207	0.9266	1.8826	8.0

(FH-FL)/(FS/QT)=0.7206 S=.7

Q'T	FL/(FS/QT)	FH/(FS/QT)	PA(dB)
0.4507	0.2167	0.9373	-8.0
0.4638	0.2270	0.9476	-7.5
0.4774	0.2378	0.9584	-7.0
0.4913	0.2490	0.9696	-6.5
0.5057	0.2606	0.9812	-6.0
0.5204	0.2727	0.9933	-5.5
0.5356	0.2852	1.0058	-5.0
0.5513	0.2983	1.0188	-4.5
0.5674	0.3118	1.0324	-4.0
0.5839	0.3259	1.0464	-3.5
0.6010	0.3404	1.0610	-3.0
0.6185	0.3555	1.0761	-2.5
0.6366	0.3712	1.0918	-2.0
0.6552	0.3874	1.1080	-1.5
0.6743	0.4043	1.1248	-1.0
0.6940	0.4217	1.1423	-0.5
0.7143	0.4397	1.1603	0.0
0.7351	0.4584	1.1790	0.5
0.7566	0.4777	1.1983	1.0
0.7787	0.4977	1.2183	1.5
0.8014	0.5184	1.2390	2.0
0.8248	0.5398	1.2604	2.5
0.8489	0.5619	1.2825	3.0
0.8737	0.5848	1.3054	3.5
0.8772	0.6084	1.3290	4.0
0.9255	0.6329	1.3534	4.5
0.9525	0.6581	1.3787	5.0
0.9803	0.6841	1.4047	5.5
1.0090	0.7111	1.4317	6.0
1.0384	0.7389	1.4595	6.5
1.0687	0.7675	1.4881	7.0
1.0999	0.7972	1.5177	7.5
1.1321	0.8277	1.5483	8.0

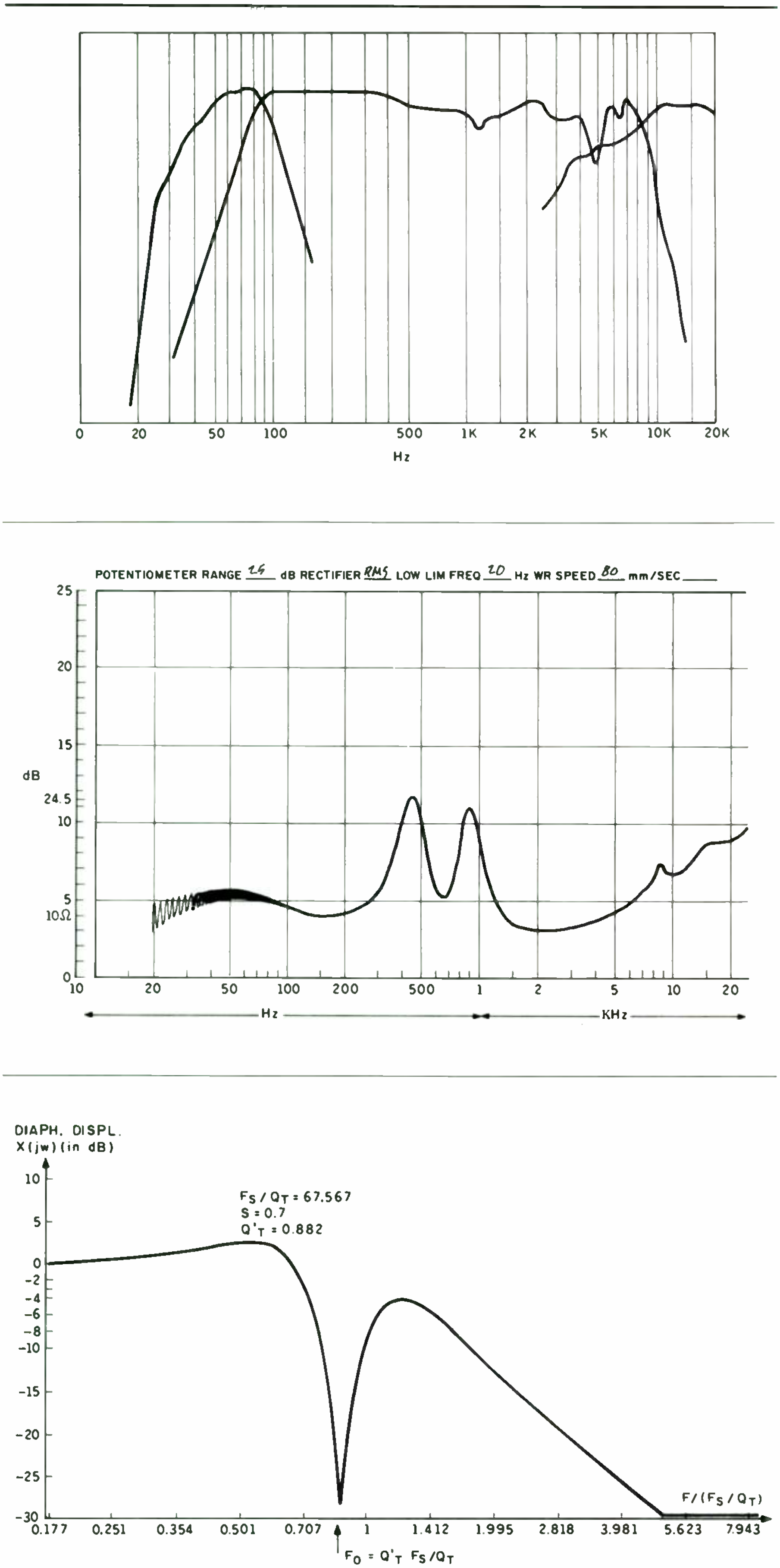


FIGURE 5: Symmetrically-loaded system, the Opera; a) frequency response; b) impedance response—frequency scale 1/10; c) diaphragm displacement curve.

ly calculate the frequency response of your system, I have included *Table 3* (a–d), that I designed with the aid of my computer. Each group corresponds to an S value (0.4–0.7) and is composed of 33 alignments.

One alignment corresponds to the same sensitivity as the classic closed-box or vented-box design, and then I provide 16 alignments with higher sensitivity, and another 16 with lower sensitivity. For the same value of S, the alignments are classified by increasing values of Q'_T (which also increases sensitivity). For each alignment I give $(F_L)_{-3dB}$ and $(F_H)_{-3dB}$. All frequencies are given versus the normalized frequency, F_S/Q_T. The acoustic levels are given in decibels; zero dB corresponds to a classic closed or vented box sensitivity. A positive level shows the system sensitivity is greater than a classic closed or vented enclosure, equipped with the same driver.

You can jump from one *Table* to the other through the acoustic levels scale. To find any specific alignment you can interpolate values within or between the groups.

DESIGN PROCEDURE. To design your own system, I recommend you select a bass driver for its basic characteristics, its price and reputation. As far as I'm concerned, I prefer a low F_S/Q_T ratio, about 60–120, because with an S value of 0.5, you get a low −3dB cut-off frequency of 24–50Hz (roughly $0.4F_S/Q_T$) with an acceptable loss of sensitivity (−3.5dB).

Remember, the lower the F_S/Q_T ratio, the lower the cut-off frequency of the system.

THE OPERA. For an example, I designed one symmetrically-loaded system, the Opera, around the KEF B139 driver. First, let's follow the procedure described by Augris and Santens.

The manufacturer's data for the KEF B139 is as follows:

$$F_S = 25Hz,\ Q_T = .37,\ V_{AS} = 164 \text{ liters}$$

A. You have a choice for the value of S. The design goal is to obtain the flattest frequency response. For the Opera, *Figure 5a* (band C) gives the frequency response, the impedance and the diaphragm displacement curves.

If we fix a modulation, ϵ, equal to zero, we will choose S = 0.7.

B. To calculate the front compartment volume:

$$V_F = (2S\,Q_T)^2 \times V_{as} = (2 \times .7 \times .37)^2 \times 164$$

so $V_F = 44.01$ liters

C. For the −3dB low cutoff frequency F_L we chose

$$(F_L)_{-3dB} = 40\text{Hz}$$

$$F_S/Q_T = 67.57\text{Hz so } F_L / (F_S/Q_T) = .5920$$

D. The value of Q'_T. From *Table 3d* we find:

$$F_L/(F_S/Q_T) = 0.5619,\ 0.5920, \text{ and } 0.6084;$$

$$\text{and } Q'_T = 0.8489,\ x, \text{ and } 0.8992$$

The interpolation gives x = 0.8815. So the frequency and transient response is defined by $Q'_T = 0.8815$ and $S = 0.7$.

E. The calculation of the −3dB high cutoff frequency F_H; $S = 0.7$ as bandwidth:

$$\Delta F/(F_S/Q_T) = 0.7206$$

$$\frac{F_H}{F_S/Q_T} = \frac{F_L}{F_S/Q_T} + \frac{\Delta F}{F_S/F_T} = 0.5920 + 0.7206$$

so,

$$F_H = 1.3126 \times (F_S/Q_T),\ F_H = 88.69\text{Hz}$$

F. To determine the back and total volumes: V_B and V_T. The back volume is given by:

$$V_B = \frac{V_{as}}{\left(\frac{Q'_T}{Q_T}\right)^2 - 1}$$

so $V_B = 35.07$ liters

$$V_T = V_B + V_F, \text{ so } V_T = 79.08 \text{ liters}$$

G. Tuning Frequency of the enclosure.

$$F_0 = Q'_T \times F_S/Q_T, \text{ so } F_0 = 59.56\text{Hz}$$

H. Vent dimensions. The chosen diameter: $D_V = 10$cm;

The tuning frequency: $F_0 = 59.56$Hz;
The front volume: $V_F = 44.01$ liters.

$$L_V = \frac{3 \times 10^4 \times S_V}{V_F \times F_0^2} - 0.9\sqrt{S_V}$$

so $L_V = 7.12$cm

I. Sound pressure level (SPL):

$$\left(\frac{P_A \text{ ch sym}}{P_A \text{ flat baffle}}\right)_{dB} = 3.65\text{dB}$$

TABLE IV

VARIOUS DRIVER ALIGNMENTS

DRIVER	DRIVER CHARACTERISTICS: DIAM. (cm)	SENS. (dB)	F_S (Hz)	Q_T (—)	F_S/Q_T (Hz)	V_{as} (liters)	$\Gamma = B_l/M_{md}$	ALIGNMENT ACOUSTIC PARAMETERS: F_l (Hz)	F_H (Hz)	P_A (dB)	S (—)	Q'_T (/)	CONSTRUCTION PARAMETERS: V_F (liters)	V_B (liters)	V_T (liters)	F_0 (Hz)	D_V (cm)	L_V (cm)
Altec 411-8A	30	92.8	19	.35	54.3	829.2	—	20	59.1	−2.1	.7	.633	199.1	364.5	563.6	34.4	20	24.1
								20	89	−4.4	.5	.777	101.6	210.9	312.5	42.2	15	17.4
								15	84	−7.4	.5	.654	101.6	333.0	434.5	35.5	13	20.7
Dalesford D30/110	—	84	32	.35	91.4	14.9	—	37.5	153.7	−3.2	.5	.8304	1.8	3.2	5.0	75.9	4.0	32.6
Dynaudio 30W54	30	—	22	.357	61.6	257	—	20	98.3	−5.7	.5	.720	32.8	83.9	116.7	44.4	12	43.1
								20	64.41	−3.6	.7	.582	64.2	154.7	218.9	35.9	12	31.5
21W54	21	—	30	.303	99.0	60	—	28	153.9	−7.1	.5	.66	5.5	15.84	21.35	65.6	5	20.8
								30	101.4	−4.3	.7	.56	10.8	22.23	36.03	55.14	7	29.6
Focal 10N501	26	—	22	.231	95.2	213	353	25	146.1	−7.9	.5	.630	11.37	32.6	43.9	60.4	8	29.9
								25	93.6	−5.9	.7	.508	22.3	55.5	77.8	48.4	9	29.4
8N501	20	—	32	.235	136.2	78	670	40	213.1	−6.8	.5	.678	4.3	10.7	15.0	92.3	7	25.9
								40	138.1	−4.7	.7	.546	8.4	17.8	26.2	74.3	8	25.9
7N501	17.9	—	37	.236	156.8	33	732	45	244.3	−7.0	.5	.669	1.8	4.7	6.5	104.9	5	25.2
								45	158.0	−4.9	.7	.538	3.6	7.9	11.5	84.3	6	28.3
7K011	17.9	90	31.6	.321	98.4	65.9	526	25	150.1	−8.2	.5	.622	6.8	23.9	30.7	61.3	6	28.5
								25	95.9	−6.3	.7	.498	13.3	47.0	60.3	49.0	6	21.8
5N312	13.6	89.5	52.4	.323	162.2	11.4	818	40	246.2	−8.5	.5	.612	1.2	4.4	5.6	99.2	3.5	21.8
5N411	13.6	87.0	41.6	.283	147.0	15.3	744	34.3	120.0	—	.775	.436	2.94	11.13	14.1	64.1	3.5	21.1
Fostex FE103	10	89	80	.35	228.6	5.2	—	50	340.6	−9.7	.5	.571	.64	3.1	3.7	130.5	3.5	23.8
								50	214.7	−7.9	.7	.453	1.3	7.7	8.9	103.6	4	24.9
FE103Φ	10	90	70	.32	218.8	6.3	—	50	328.1	−9.3	.5	.586	.65	2.7	3.3	128.1	3.5	24.5
								50	207.1	−7.4	.7	.466	1.3	5.6	6.9	101.9	4	25.5
JBL 2245H	46	96.4	20	.27	74.1	820.7	138	18	71.4	−6.8	.7	.484	117.3	371.0	488.3	35.8	15	23.2
								18	112.2	−8.7	.5	.607	59.8	202.8	262.6	44.9	12	18.5
KEF B139SP1044	21x14	88.6	25	.37	67.57	164	205	40	88.7	+3.7	.7	.882	44.0	35.1	79.1	59.7	10	7.1
								20	105.9	−6.7	.5	.681	22.5	68.7	91.1	46.0	8	25.3
B200SP1063	20	—	25	.45	55.6	132	285	30	100.6	−0.2	.5	.989	26.7	34.5	61.2	54.9	8	12.3
B110SP1003	13	83.1	38	.31	122.6	24	724	35	190.8	−7.0	.5	.667	2.3	6.6	8.9	81.7	4.4	26.1
								35	123.3	−5.0	.7	.536	4.5	12.1	16.6	65.7	5	26.2
McGee Radio TP165F	16.5	89	53	.42	126.2	15.6	—	30	190.4	−8.9	.5	.599	2.8	15.1	17.9	75.6	4.0	20.8
								30	120.9	−7.0	.7	.477	5.4	53.5	58.9	60.2	5.0	26.1
Madisound 81524DVC	20	92	36	.30	120	51	314	28.0	180.5	−9.1	.5	.593	4.6	17.6	22.2	71.1	6	31.8
Peerless TO 125F	13	87	55	.45	122.2	11	—	35	190.4	−7.0	.5	.668	2.2	9.2	11.4	81.6	4.4	27.2
TA 305F	30	89.5	25	.31	80.7	188	—	20	122.5	−8.5	.5	.614	18.1	64.4	82.4	49.5	9	35.9
								20	78.1	−6.5	.7	.490	35.4	125.4	160.8	39.5	10	34.6
Polydax HIF12EB	12	92	56	.43	130.2	15.9	1500	37.5	203.0	−7.0	.5	.670	2.9	11.1	14.1	87.3	5	22.3
								30	195.5	−9.2	.5	.588	2.9	18.3	21.2	76.6	4.4	22.9
HIF13FSM	13	86	50	.37	135.1	18.6	900	40	211.8	−6.7	.5	.681	2.6	7.8	10.3	92.0	5	23.3
								40	137.4	−4.6	.7	.549	5.0	15.5	20.5	74.1	6	26.1
HIF17HS	17	87.6	29	.40	72.5	65.0	417	25	117.2	−5.1	.5	.747	10.4	26.2	36.6	54.1	6	23.1
								25	77.2	−2.9	.7	.606	20.3	50.2	70.5	43.9	7	23.7
HD17B25	17	84	45	.48	93.8	32	470	35	154.2	−4.2	.5	.784	7.4	19.2	26.6	73.5	7	23.4
								35	102.6	−1.9	.7	.639	14.5	41.4	55.9	59.9	8	22.7
HIF20HSM	20	91.4	28	.33	84.9	148.9	506	30	137.9	−4.8	.5	.758	16.2	34.9	51.1	64.3	10	27.2
								30	91.1	−2.6	.7	.616	31.8	59.9	91.6	52.3	12	29.5
HD24S45	24	91	27	.28	96.4	178.0	487	25	147.6	−8.0	.5	.630	14.0	43.8	57.8	60.7	8	22.9
								25	94.5	−6.1	.7	.504	27.4	79.5	106.9	48.6	10	24.9
SEAS P17RCY	17	91	37	.24	154.2	36	636	45	241.0	−6.8	.5	.676	2.1	5.2	7.3	104.1	5	22.2
								45	156.1	−4.7	.7	.544	4.1	8.7	12.8	83.8	6	24.9

The SPL is 3.65dB greater than a conventional closed or vented box.

Figure 5 shows the response curves for the Opera. You may notice that although the alignment we have chosen here allows a good transient response, it does not yield a very wide frequency range.

To gain further perspective, look at another alignment that exhibits a very extended frequency range at the expense of low sensitivity (-6.67dB) and a non-flat frequency response ($\epsilon = 1.25$dB), but in a reasonable volume of 91.11 liters.

$s = 0.5$

$V_F = 22.45$ liters
$(F_l)_{-3dB} = 20$Hz
$Q'_T = 0.6811$
$(F_H)_{-3dB} = 105.89$
$V_B = 68.66$ liters; $V_T = 91.11$ liters
$F_O = 46.02$Hz

Continued on page 64

A CERAMIC SPEAKER ENCLOSURE

BY DAVID WEEMS

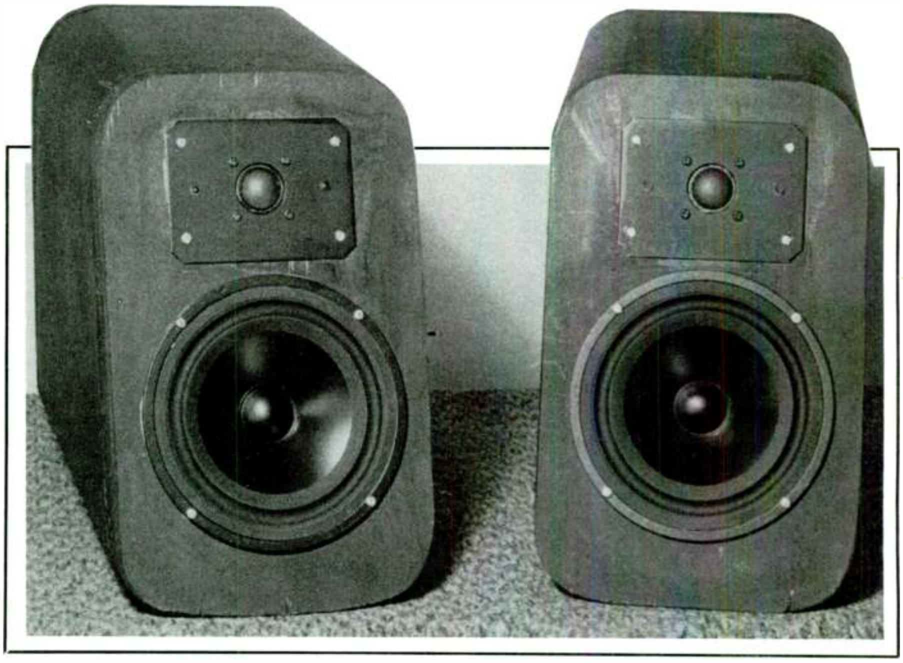


PHOTO 1: Completed speakers without felt and grille. Thin (⅛″) felt was used on the speaker board with double layer around the tweeter dome.

An ideal closed-box speaker system, terminology that I.M. Fried and others might call an oxymoron, would have enclosure walls that are opaque to sound. Such a wall could not move. Practical walls do move and, by moving, absorb energy from the system. Then they radiate it. The transient response of enclosure walls isn't good. Slow to start and slow to stop, they add a mild time distortion at best and severe hangover at worst. The net effect of wall movement is a blurring of tones, especially in the bass.

To see how much importance the average consumer places on this aspect of speaker design, check the wall panels of the speakers in your local department stores. Speaker builders are surely the only people on earth who give any thought at all to wall vibration. I know of one case where an experienced cabinet maker, working on plans for a large speaker system, substituted thin panels in the back where "it doesn't show."

Audiophiles, of course, have known about wall vibration effects since the early days of high fidelity. In efforts to reduce panel resonance, some used everything available, including the kitchen sink, as G.A. Briggs once even resorted to![1,2] "One of the best we ever tried," he stated.[3]

An ideal wall would be both rigid and damped. High density materials are usually rigid but are not always well damped. Steel, for example, is almost ten times denser than particle board, suggesting that you could use steel with a thickness of about one-tenth that of a suitable particle board for a satisfactory enclosure. If properly damped, such an enclosure should perform well. Undamped, it would surely ring.

Brick and concrete block enclosures had their day, but that was before the advent of stereo put a premium on space saving.

During the height of the experiments with unusual enclosures, some builders used ceramic drain tiles. A typical tile enclosure used a three-foot section of ten- or twelve-inch pipe, with a speaker installed at the top of the column, facing upward. Getting clean sound out of a drain pipe proved to be an elusive task. The pipes behaved the way pipes always behave. They resonated. Builders tried various methods of suppression for the air column resonance. Briggs suggested an acoustic filter combined with a degree of stuffing.[4] Those pipes had problems but wall vibration was not a significant one. The stiffness of the material, combined with the curvature, made extremely rigid walls.

Some years ago I used ceramic flue tiles for some speakers. I bonded plywood and particle board to each open end for the speaker board and back, then braced the enclosure with a threaded steel rod through the middle, tying the back to the front. The speakers were a bit awkward and heavy so I gave them to some friends. Recently the friends told me their son had upgraded the enclosures with new drivers and they outperformed every speaker in his collection. With that encouragement I decided to take another look at flue tile.

KIND OF BOX. I must confess that I usually find ported boxes more interesting than sealed enclosures. Designing a reflex system is more challenging and the addition of a port gives me a more complete feeling of accomplishment. But it does add a degree of constriction to the planning, while a second-

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1. Briggs, G.A., *A to Z in Audio*, Gernsback Library, 1961, p. 30.
2. Briggs, G.A., *Cabinet Handbook*, Wharfedale Wireless Works, Ltd., 1962, p. 12.
3. Ibid, p. 12.
4. Briggs, G.A., *A To Z In Audio*, Gernsback Library, 1961, p. 38.

order closed-box (CB) system has the advantage of simplicity. Nothing is inherently evil in doing things the easy way. Audio extremists sometimes forget there is more than one way to get good sound.

The advocates of closed boxes include a number of designers who value "transient purity" over other virtues in box choice. Ported box fans say that reflex designs offer superior damping in the critical bass region. The CB people, on the other hand, claim their sealed systems excel in the upper bass band, and that the CB woofers have "faster" bass there.

A less controversial advantage of CB systems is the second-order aspect. The sharp cut-off rate of the common reflex was what A.R. Bailey criticized when he suggested the transmission line as an alternative.[5] A slow rolloff at each end of the spectrum is both theoretically desirable and practically useful. Such a system can produce considerable bass power below the theoretical cut-off point, extending the audible bass range.

Critics mention several problems of the closed box, not all of which are acknowledged by its champions. Some call it a "pressure box" and talk about the "oil can effect." Fried considers them unworthy for high fidelity use. He says the BBC rejected them long ago for nonlinearity at low levels and that Danish researchers found they produce "nonlinear time distortion."[6]

Fried gives the impression that these reviewers reject all closed-box systems. In fact, the performance of speakers in sealed enclosures varies widely. You can hear bad examples of closed-box design in almost any discount or department store and not always from unknown brand names. The typical problem can be heard several aisles away, in the thump, thump, thump of heavy bass. Such speakers may be designed with a high Q to attract both juvenille rock fans and those older listeners who remember old juke boxes with nostalgia.

For many years the manufacturers of most better quality "acoustic suspension" speakers aimed for a system Q_{TC} of about 1. This gives the most extended bass response but with a degree of peaking at resonance. Speakers designed for rock music are likely to have a higher Q_{TC}, sometimes much higher. For optimum damping, a Q_{TC} of 0.7 is usually suggested. Martin Colloms says that even this degree of damping may not be adequate in conventional listening rooms. He recommends a Q_{TC} of 0.5 to 0.6.[7] That corresponds to the values found in some of today's highly esteemed audiophile speakers.

Considering all this, I decided to develop a closed-box system for my experiment with a new ceramic enclosure and aim for a Q_{TC} below 0.7. There seems little point in taking the trouble to reduce panel resonance and yet allow the system to peak at resonance. Next, I needed to find a flue tile of suitable size and to choose a driver.

WHICH TILE AND WOOFER? To keep the weight of a tile enclosure within reasonable limits, I set the limit for woofer size at no greater than 8", maybe smaller, depending on which kind of flue tile was stocked by the local dealers.

I visited several places and found some carried only a single size, an 8 by 8 by 24" tile. The tiles are labeled by their outside dimensions and only roughly at that, and may vary from brand to brand.

5. Bailey, A.R., "A Non-Resonance Loudspeaker Design," *Wireless World*, October 1965.

6. Fried, I.M., "What About Loudspeakers?" publication of Fried Products Co.

7. Colloms, Martin, *High Performance Loudspeakers*, second ed., Pentech Press, 1980.

The 8 by 8″ tiles I found were really about 8½ by 8½″ on the outside and about 6½″ square on the inside. A tile of square cross section was not very promising for my purposes, so I went on to other yards. Another dealer had large flue tiles of about 13 by 18 by 24″, with a total cubic volume of almost 2 ft.[3] Such a tile might be appropriate to make a pair of enclosures for suitable 8″ woofers.

The next size I found, which I chose, was 9 by 13 by 24″. The total cubic volume of one is about 1700 in.[3], or 27.5 liters. It should be sufficient to make a couple of enclosures with a cubic volume just under 14 liters or, after subtracting 10 percent for the space occupied by the drivers, about 12.5 liters.

Flue tiles are not made to precise dimensions so if you get a similar size you may find a slight variation in volume. Some flue tile dealers are evidently careless in handling their stock; many specimens I saw had hairline cracks in them. Each tile should be carefully inspected before acceptance. And don't drop one of these onto a hard surface.

The available cubic volume seemed too small to use an 8″ woofer at low Q_{TC} so I narrowed the search for a smaller driver. A quick look through some speaker specification lists showed two likely candidates, the Focal 7N401, which has a recommended volume of 12 liters and the SEAS P17RC. SEAS lists a volume range for this unit at 7–18 liters. I didn't have an up-to-date catalog listing for the Focal woofer so I ordered a couple of the P17RCs from Audio Concepts. These little speakers are nicely made, but at their current price they should be. For the tweeters I finally went to the well-tested Audax 12X9D25-8, available from McGee.

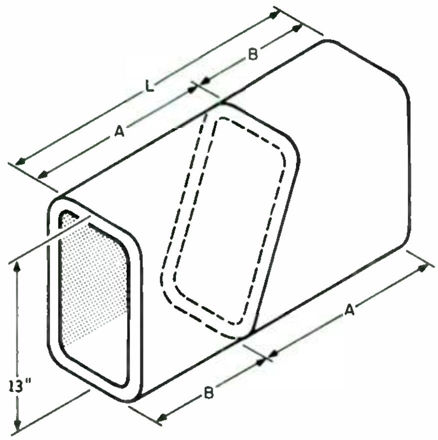


FIGURE 1: Dimensions of cut tile for various slopes. (Length of tile can vary.)

TABLE I

ENCLOSURE DIMENSIONS (inches)

Slope	A	B
For L = 24:		
20°	14⅜	9⅝
15°	13¾	10¼
For L = 23½:		
20°	14	9½
15°	13½	10

BOX CONSTRUCTION. One advantage of using a tile enclosure is that the basic box is ready-made. After playing around with several possible designs, I decided to make a box with a sloping speaker board. But how much slope? My first idea was to place the enclosures rather low in the room because of their weight, so I planned for a 20° slope. A 15° slope might be better in some situations, for instance, if the speakers are placed with their bases about 24″ above the floor. *Figure 1* is a cutting guide for the different slopes (see *Table 1* for dimensions).

To cut the tile you can use a masonry

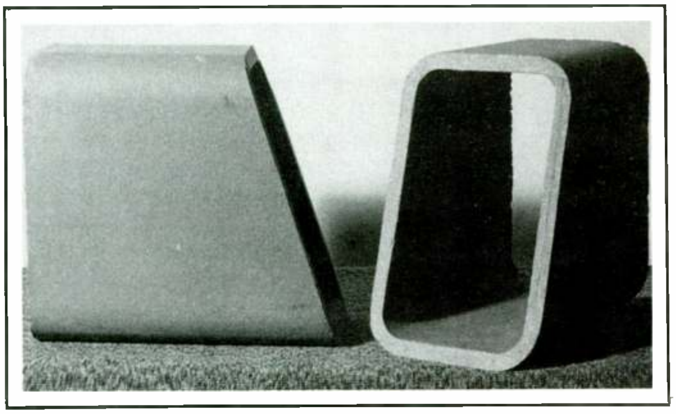


PHOTO 2: A single piece of 9 by 13 by 24" flue tile, cut in half to make two enclosures.

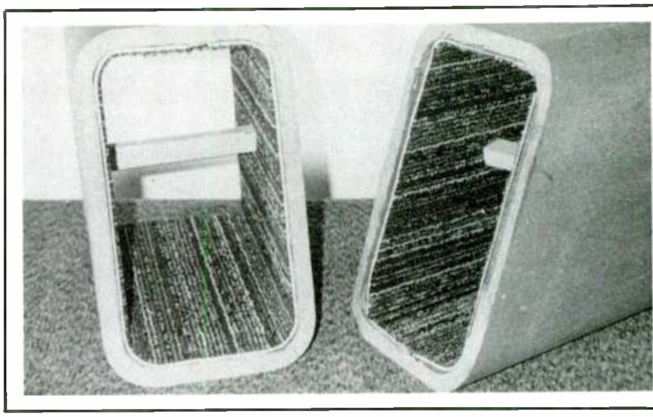


PHOTO 3: Tile sections lined with foam-backed carpeting and braced in the middle of each side. Ends of braces are angled slightly to make solid contact with walls when driven almost perpendicular.

blade in a portable saw. This job requires great care, and goggles and a dust mask are essential. The cutting should be done in a well-ventilated area because the saw blade throws off a great fog of tile dust.

The instructions with my masonry blade stated that only a 90° cut should be attempted; other angles can cause side pressure on the blade with the danger of shattering. For the end cuts, at the top front and bottom front of the enclosures, you must set the saw at the angle desired for the speaker board. To make the cuts, set the angle, such as 20° for the speakers shown here, and then set the saw adjustment so it will make no more than a very shallow cut at each pass over the tile. The maximum safe cut is under ⅛" per pass. These end cuts should be made first, with the tile supported by a solid work table or placed on a floor. After you have cut through the end walls, lay the tile on its side to finish the cutting. Set the saw at 90° but don't try to cut too deeply. Limit each cut to ⅛" and you should have no trouble.

After cutting out my box sections, I tested the tile by rapping it. Sure enough, there was some ringing. By tapping the tile on the sides I found that one section rang at about 233Hz, the other at about 273Hz. The difference was a surprise because the two sections were almost identical. Although the walls appeared to be of equal thickness, probably some variation accounted for the difference in frequency. Only the large flat sides showed any significant susceptibility to ringing. My first impulse was to bond ½" asphalted insulation board to the inner walls, but that would reduce the internal width by a full inch.

After shopping around for thinner material that might damp the walls, I found a likely prospect in K-Mart's home department. They stock rolls of thin indoor-outdoor carpeting, about ¼" thick, of which ⅛" is a polyurethane foam backing. At the store where I bought mine, a running foot, 6 feet wide, sold for $3.69. One foot was all I needed for two enclosures. I glued the carpeting inside the tile by bonding the foam backing to the walls with silicone rubber sealant.

After installing the carpeting, no ringing occurred when the tile was hit on the inner wall, but, if struck on the outside wall, some residual ringing remained. I found that a brace, cut just long enough to be wedged between the two side walls greatly subdued that.

I chose particle board for the back and top quality plywood for the speaker board of my enclosures. I found some Finnish ¾" birch plywood with 14 plys and absolutely no flaws, in the discard pile of a furniture factory. Such plywood, if you can find it, is far superior to the stuff sold by most lumberyards. I glued and screwed a single ¾ by ¾ by 6" brace to the inside surfaces of the front and back, placing the brace on the speaker board in the space between the woofer and tweeter cutouts.

Mount the speaker terminals on the particle board back before attaching it. You may also want to install an L-pad on the back panel. I mounted the crossover network and L-pad on a separate

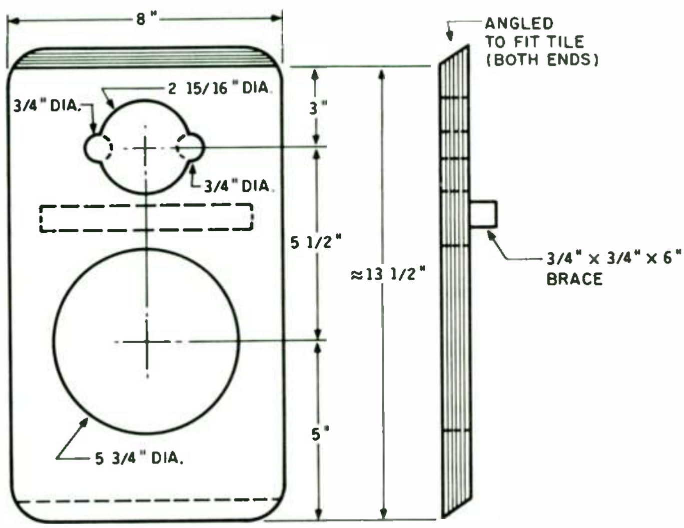


FIGURE 2: Speaker board dimensions.

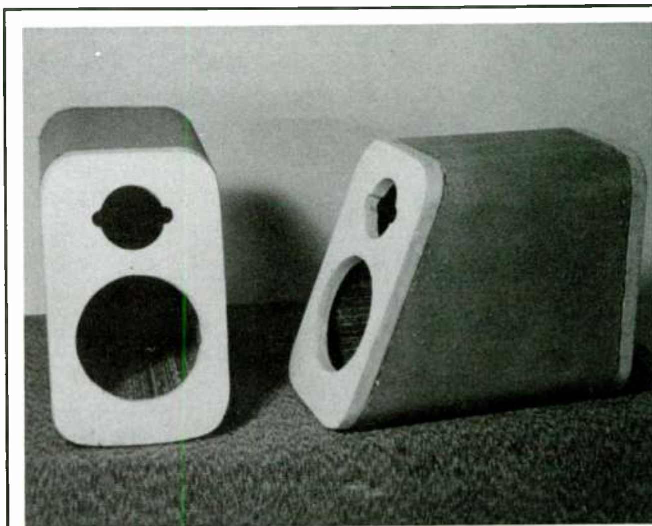


PHOTO 4: Enclosures after speaker boards and backs are glued on, before painting.

PARTS LIST

Audax HD12X9D25
SEAS P17RC

C1	3.6μF
C2	8.4μF
C3	12μF
R1	4Ω
R2	7.5Ω
L1	0.3mH
L2	0.4mH
L3	0.32mH

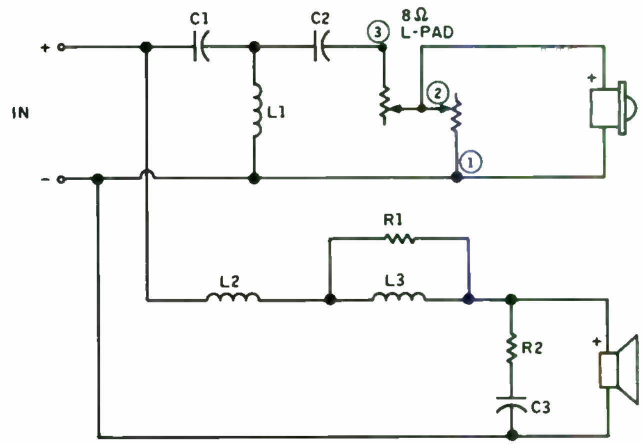


FIGURE 3: Schematic diagram of crossover network used in completed system.

small board, attached to the outside of the back panel. Another possibility is to install these components in a separate compartment below the enclosure.

Set the back on a flat surface, supported by rails to protect the terminals. Run a bead of good quality silicone rubber cement around the matching surfaces of the back panel and the rear edge of the tile. Set the tile in place on the back; its weight ensures a good seal. After the glue has set, in a day or two, prop the enclosure so that the front edges of the tile are horizontal. In this position you can glue down the speaker board with enough weight on it to make a seal.

After the glue has set, you can rasp or sand off the edges of the speaker board and back to fit the tile. The tile can be covered with any suitable material, such as plastic veneer, or it can be painted. I painted my enclosures flat black. I installed about 4 ounces of polyester batting in each enclosure, enough to loosely fill the interior without compression.

CROSSOVER NETWORK. I first planned to use a bargain tweeter, the Audax HD100D25-BA-HR. I built up a "book" crossover for that tweeter, aimed at a crossover frequency of 2,500Hz. The tweeter was too prominent unless an excessive amount of cut was used; excessive by comparison with expectations. I ran some frequency tests and found the tweeter was peaking below the theoretical crossover frequency. I tinkered with the network until I got the performance I wanted. Then, I substituted the old stand-by tweeter, the HD12X9D25. This model seemed smoother and performed more to my expectations, so I used it instead.

It should be possible to do away with

the L-pad by adding a simple resistance to the tweeter circuit. The resistance might be added before the crossover elements where it will not upset the frequency response of the crossover.

I worked out the crossover values, but in the specifications published by SEAS, the value for the voice coil inductance of the P17RC is 1.1mH. That seemed rather large for such a small woofer, so I tested mine and determined the value is below 0.5mH. In any case, the Zobel network of 7.5Ω and 12μF produces an impedance curve that is virtually flat at 6Ω above 500Hz.

After wiring the system, the sound, while having a great feeling of presence, seemed a bit forward. I added the second coil to the woofer circuit, bypassed by a 4Ω resistor.

LISTENING TESTS. This is not a rock speaker. Neither is it one for high power full-range service. For example, the digital cannon on the Telarc CD of Tchaikovsky's *1812 Overture* bottoms the voice coils of the SEAS woofers at a lower volume than occurred with Peerless TP-165F woofers in tapered pipes ("Experiments with Tapered Pipes," *SB* 2/87). The dynamic range of the SEAS drivers seems more limited than that of some other brands of 6½" woofers.

So much for the bad news. On the other hand, these speakers are natural and uncolored. Their forte is in reproduction of small groups, voice, and piano music. They do a good job on most symphonic programs, with a solid bass response that is surprising for their size. But they reveal the lack of bass, if such exists. In fact, these speakers show just how much difference exists in various program sources. The bass appears light on many records or CDs and because of that I find that I use the loudness control much of the time.

The Q_{TC} of these speakers is below 0.7, as planned. The Q isn't quite as low as Colloms suggested, but I hear no undamped bass.

The virtue of this kind of speaker is subtle. They don't have the power to excite you by sheer sound intensity levels. And the bass isn't the kind that bowls you over. But clarity and definition are evident.

Regardless of how much importance one places on tonal purity, these speakers have one advantage over my last experiment in speaker building that cannot be denied. Living in an old house, we have our share of rodents and these speakers have no ports to worry about. One of these will never become a mouseoleum.

A TECHNIQUE TO MEASURE INDUCTANCE

BY G.R. KOONCE
Contributing Editor

When it appeared that the inductance bridge I built some ten years ago was measuring low, I decided to look for some way to verify the error. The technique I developed has proved to work well and is documented here. It allows measurement of the inductance (and approximate Q) of chokes used in passive crossovers, and you require only a sinusoid source (where you know the output frequency), an AC voltmeter, an amplifier, an ohmmeter and some selected power resistors.

While not as quick and convenient as a bridge or LCR meter, my inductance measuring technique shows good accuracy, measures inductance and Q at a frequency typical of the application range and will allow establishing the choke linearity over the current range to be used. You can compare results for a variety of chokes with other measurement techniques in the *Tables* I provide, and, in addition, I present a simple BASIC program to perform the needed computations.

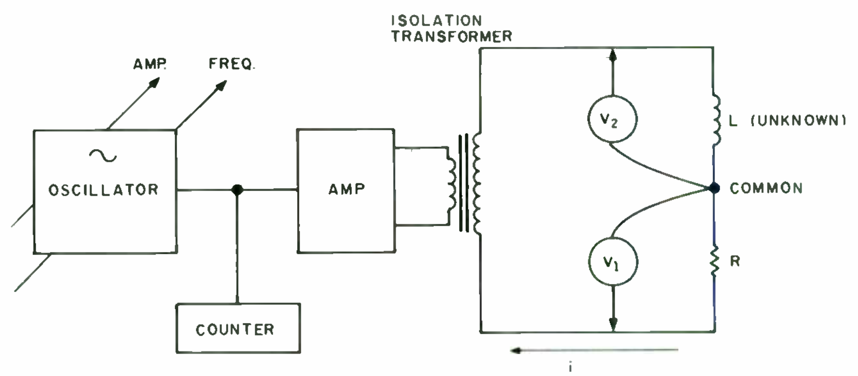


FIGURE 1: Basic inductance measurement approach.

Figure 1 shows the basic concept. The sinusoidal output of an amplifier is transformer isolated (to remove ground reference, later found not to be necessary), then drives the unknown choke and a resistor in series. Voltage magnitude is measured from the common point, at the choke-resistor junction. V1 is the voltage across the resistor, and V2, across the choke.

Adjust the sine wave frequency until V1 equals V2. Assuming for a moment a high choke Q, then:

$$i \times (2 \times \pi \times f \times L) = i \times R$$

where:

L is choke inductance in henries,
R is the resistor value in ohms
i is the RMS current.

Now, if we select R to have a value of $2\pi\Omega$ (6.28Ω), we have:

$$i \times (2 \times \pi \times f \times L) = i \times (2\pi), \text{ or } L = 1/f$$

so the inductance of L (henries) is the reciprocal of the frequency (hertz) where V1 = V2.

The result is a simple way to measure inductance. If the real choke can be modeled as a series ideal inductor and an AC resistance, this approach can be accurate to ±0.5% for choke Q greater than 10; down to ±2% at a Q of 5 (I'll show a method to correct for Q later in this article).

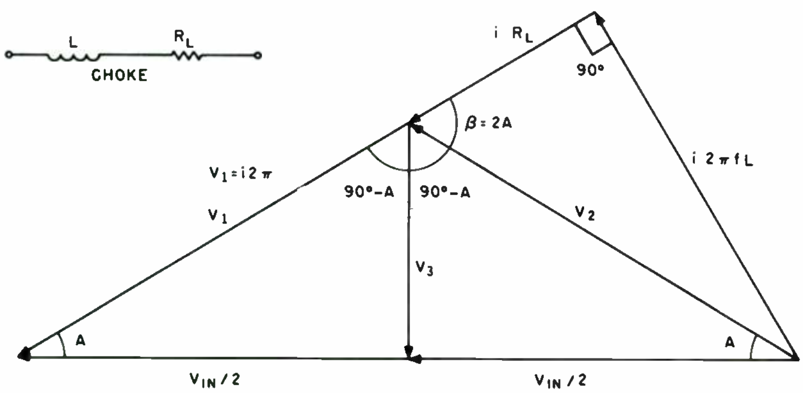
The approach offers the following advantages for audio choke measurement:

- You only need equipment generally available to speaker builders.
- You can measure the inductance and Q at a frequency that is representative of the frequency band where the choke will be used, i.e., 2mH at 500Hz; 100μH at 10kHz.
- The inductance can be measured at whatever RMS current level you decide to use. Be mindful of choke and amplifier limitations and use resistors of sufficient power rating.
- You can verify choke linearity with power. At low power set V1 = V2. Increase the current until V1 and V2 are no longer equal. Assuming the amplifier stays linear and the resistors do not overheat, you have found the choke linearity limit. Be very careful. Using this steady-state test you could easily cook the test setup or the coil before meeting the linearity limit. For a better technique for measuring choke linearity limit, see my previous article ("Crossover Component Capabilities and Requirements," *SB* 3/86).

Initially I thought the transformer was needed to isolate the amplifier ground from the "common" measurement point. I tried some coils with the setup of *Fig. 1*, then removed the transformer (*Fig. 2*) and obtained identical results. For these tests all test equipment was run floating from power ground. The impedances involved are low enough that the capacities associated with the floating "common" do not bother the measurements.

As a further test I added 300pF across a 0.53mH choke, then across the 6.28Ω resistor. The capacitor changed the results about 1%. Reasonable test equip-

Vector Diagram and Equations for Choke with Series Resistance



GIVEN: MAGNITUDE OF V_1 = MAGNITUDE OF V_2

BY INSPECTION: $\sin A = V_3 / V_1 \qquad \beta = 2 \times A$

$$\sin\beta = \frac{i2\pi fL}{V_2} = \frac{i2\pi fL}{V_1} = \frac{i2\pi fL}{i2\pi} = fL$$

SO: $L = \sin\beta / f = 1/f * \sin\beta$

FOR CHOKE WITH SERIES RESISTANCE $Q = \frac{2\pi fL}{R_L}$ AT f

SO: $Q = \frac{i2\pi fL}{iR_L} = \tan\beta$

ment run floating from ground should produce accurate results. Observe the necessary safety precautions when operating AC powered test equipment isolated from ground.

I happen to own a dual channel AC voltmeter, which is ideal for this test. You simply adjust the frequency until the two needles line up. You can then move the voltage up and down to observe linearity. The technique will work with a single voltmeter, but adjustment will be a little more time-consuming. Two independent voltmeters would help speed the testing.

With a little math work the technique will give an indication of coil Q and correct the inductance value when Q is low. *Figure 2* is the test setup. The two selected 4Ω resistors establish a center tap across the source (amplifier output). The sidebar shows the vector diagram of how Q is derived and all equations. Thus the technique allows two approaches (*Fig. 2*).

SHORT TEST. Approximate inductance, no Q.

1. At a convenient voltage level, adjust frequency until V1 equals V2. Record V1 (volts) and f (hertz).
2. Calculate:

approximate $L = 1/f$ (henries).

RMS choke current for test = V1/6.28 (amps).

LONG TEST. More accurate on low Q coils, gives indication of Q.

1. At a convenient voltage level, adjust frequency until V1 equals V2. Record V1 (volts) and f (hertz).
2. Move either meter to indicate V3 without changing drive level or frequency. Record V3 (volts); it should be less than 0.707 times V1.
3. Calculate:

Angle $B = 2 \times \arcsin(V3/V1)$;

$L = 1/f \times \sin B$ (henries)

$Q = \tan B$

RMS choke current for test = V1/6.28 (amps)

Figure 3 shows a simple program, LXCOMP.BAS, which is included on both my program disks ("New Software," *SB* 5/88; available from Old Colony, SBK-F2A and B), to perform these calculations which should run in most BASICs compatible with GW BASIC. If no value is entered for V3, the program uses the short test computation, otherwise it performs the long test.

During my work on this project, the hand held LCR meters came on the scene and I purchased one to compare its results with my technique. The LCR meters generally have 2mH full-scale on the lowest inductance scale, so you are measuring most crossover inductors on the bottom two scales, where rated accuracy is less acceptable. Also, Jim Far-

TABLE I

DESCRIPTION OF CHOKES

Sample	Type	Nominal Inductance	Wire Size	Resistance Ohms	Dia. mm	Length mm
A	Ferrite Bobbin	2.5mH	18	0.26	45	36
B		1.0mH	20	0.31	22	18.3
C		0.5mH	18	0.102	28	24.4
D		0.35mH	18	0.082	28	24.4
E		0.15mH	18	0.052	28	24.4
F	Air Bobbin	1.25mH	20	0.80	48	23.6
G		0.23mH	18	0.021	40	27.4
H	Air Solenoid	0.14mH	18	0.205	24	25.5
I	Iron Core	4mH	20	0.31	/	/

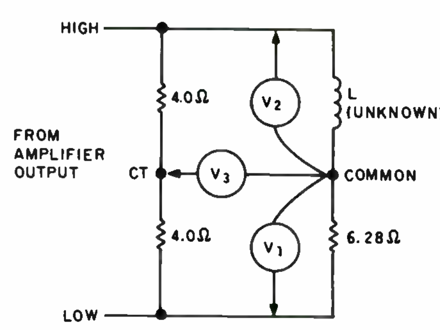


FIGURE 2: Practical circuit allowing inductance and Q measurement—6.28Ω power resistor selected; two 4Ω power resistors matched.

rand pointed out one caution, that some LCR meters use AC to measure resistance. Do not try to measure resistance of a speaker or inductor with these meters or a major error can result.

Table I shows information on a group of chokes used in this study. Ferrite core, air core and one iron core are included, with inductances from 0.15mH to over 4mH. *Table II* shows results for all but the iron core choke. Note that I corrected the inductance value given for the bridge measurement by +5%, as this work and other tests show the bridge reads 5% low.

The inductance results using the three methods are in agreement. Clearly, the Q given by the described technique is just an indication. While the math yields an accurate Q, it is impossible to get values of V1 and V3 accurate enough to yield accurate computations. The accuracy is good for low coil Q, but once we move above a Q of 10, for example, the result should be taken to mean the Q is "above 10." Keep in mind, a bridge normally measures Q at a fixed frequency (1kHz for mine) while the described technique measures the Q at the test frequency, which is usually more in line with your interest in the Q for crossover work. The LCR meter does not measure Q and its inductance value is only accurate if coil Q is high.

Table III shows test results for the iron core choke. I included this small transformer-shaped choke (49mm wide by 41mm high by 13mm thick) to show the problem with iron core coils and the advantage of the measurement technique developed here. With an LCR you would read this choke to be 3.5mH; with my bridge I get 4.4mH.

My technique shows you can have almost any value you wish, by whatever current level at which you choose to measure; ranging from 5.4mH to 7.3mH for 16mA to 800mA test currents. While the LCR meter and my bridge would not indicate this lack of linearity, the described technique shows it.

```
10 REM Program to do inductor test - LXCOMP.BAS
20 WIDTH 80: REM  5/88 - GR Koonce - Version 1.00
30 STRQ$ = " >50": STRL1$ = "Approximate ": STRL2$ = "Inductance = "
40 CLS: LOCATE 3,23
50 PRINT "Choke Test Computation Program": GOTO 120

100 CLS: LOCATE 3,15
110 PRINT "Last data entry was invalid, reenter data:"
120 PRINT
130 LINE INPUT "You may enter a coil type or name (optional) - ";N$
140 PRINT: F = 0
150 INPUT "Enter frequency in Hertz  ",F
160 IF (F < 1.1 OR F >=100000!) THEN GOTO 100 ELSE PRINT: V1 = 0
170 INPUT "Enter V1 in volts  ",V1
180 IF V1 <= 0 OR V1 > 100 THEN GOTO 100 ELSE PRINT: V3 = 0
190 INPUT "Enter V3 in volts (ENTER for short test)  ",V3
200 IF V3 = 0 THEN K1 = 1: GOTO 330
210 IF (V3 <= 0 OR V3 > .78 * V1) THEN GOTO 100

300 IF V3 > .7 * V1 THEN K1 = 1: GOTO 330
310 VX = SQR( V1^2 - V3^2 ): B = 2 * ATN( V3/VX )
320 K1 = SIN(B): Q = TAN(B)
330 L = K1 * 1000 / F: I = V1 / 6.2832: PRINT
340 IF V3 = 0 THEN PRINT STRL1$; STRL2$; ELSE PRINT STRL2$;
350 PRINT USING "###.### mHy";L
360 IF V3 = 0 THEN GOTO 390
370 PRINT: PRINT "Coil Q at ";:PRINT USING "##### Hz is";F;
380 IF K1 < 1 THEN PRINT USING " about ##.#";Q ELSE PRINT STRQ$
390 PRINT: PRINT "RMS coil current during test was = ";
400 PRINT USING "##.## Amps";I: PRINT: PRINT

500 INPUT "Do another computation? (Y/N)  ",M$
510 IF (M$ = "Y" OR M$ = "y") THEN GOTO 40
520 END :  REM change to SYSTEM for exit to DOS
```

FIGURE 3: BASIC Program to compute results for test approach.

TABLE II

TEST RESULTS—FERRITE AND AIR CORE

Sample	LCR Meter mH	Bridge* mH	Bridge* Q @ 1k	Test Approach mH	Test Approach Q @	Test Approach f (Hz)	Test Approach I (amps)
A	2.47	2.47	30.0	2.46	26.0	406	0.48
B	1.013	0.999	8.8	1.00	12.0	992	
C	0.513	0.505	14.5	0.504	26.0	1,983	
D	0.361	0.352	17.5	0.348	26.0	2,872	
E	0.157	0.149	22.0	0.149	26.0	6,687	
F	1.313	1.32	7.5	1.31	6.4	757	
G	0.242	0.243	5.7	0.235	9.7	4,229	
H	0.150	0.142	4.2	0.143	8.9	6,955	

*Inductance values for bridge corrected by +5%

TABLE III

IRON CORE COIL TEST RESULTS (SAMPLE I)

LCR Meter mH	Bridge* mH	Bridge* Q @ 1K	Test Approach mH	Test Approach Q @	Test Approach f (Hz)	Test Approach I (amps)	Test Approach V1 (volts)
3.53	4.37	4.4	5.40	5.4	182	0.016	0.1
			5.81	5.4	169	0.032	0.2
			6.33	5.4	155	0.080	0.5
			6.74	5.9	146	0.160	1.0
			7.15	5.9	138	0.320	2.0
			7.32	5.4	134	0.800	5.0

*Inductance values for bridge corrected by +5%

Pox Humana

Chrome-Dome & Loudspeakers as Weapons

By Dick Pierce

LET'S CONTINUE OUR SERIES on hi-fi foibles by taking a look at some of the problems dealers and manufacturers face when dealing with customers. The main character in our story here is a young (high school) kid, whom we subsequently dubbed "Chrome-dome."

Way back when, shortly after I and several others had started a high-end hi-fi store in Boston, we were selling, among other items, the completely new Yamaha electronics line and Celestion speakers. For those who may not be familiar with them, Celestion at that time made a line of speakers that combined reasonable accuracy with high efficiency and remarkable power handling. The top-of-the-line speaker, the Ditton 66, consisted of a 12-inch woofer/passive radiator bass system, a phenomenal 2-inch midrange that would handle untold power and a wideband, uncolored 1-inch dome tweeter.

One day this kid walks in, carrying a copy of Pink Floyd's *Dark Side of the Moon.* It seems he's looking for a killer hi-fi system. He had been sold on JBL speakers, Phase Linear amps and Crown preamps at another store, but his budget was unfortunately about 40 percent shy. We suggested he listen to the Celestion 66s and a Yamaha CA-1000 integrated amp. The system (including turntable, and so on) was within his budget, the only remaining task was to convince him that the system met his needs.

He handed us the record and we said we had a new copy. "No," he said, "I don't trust store's records, they're usually awful."

"Okay," we said, "we'll play yours."

To this person, record care was something you did with stuff you scrape off your sneakers and apply with 40-grit sandpaper. He agreed his copy was "a bit worn," but that was done by another store. We put our copy on and he was happy.

While it was playing (fairly loud) he asked that the volume be turned up. Up it went. Not loud enough, he wanted it louder. At this point the level was approaching pain, but the Yamaha had plenty left in reserve, so we didn't care much. It was far louder than we had ever listened to anything, but we were amazed at how the speakers were managing to handle it, and they showed no signs of distress. Louder and louder, still, he wanted it.

He was not convinced until the strangest thing happened. The Celestions were sitting on a couple of short pieces of 2 by 4s, because, when placed directly on the floor, the bass could sometimes be overwhelming. The 2 by 4s seemed to cure the problem. Well, whoever had set up the speakers hadn't seated them properly and during a particularly loud passage the left speaker slipped and, rocking on the verge of total imbalance, proceeded to walk forward about a foot or so.

Well, the kid was impressed beyond recovery. At that time, he took out his wallet and paid us for the system, cash, despite our attempts to convince him that the speaker had merely slipped, and was not walking under its own bass prowess. He took the system home and immediately called us up to tell us how wonderful it was.

For several weeks we heard nothing. In the interim, it seems his father had gotten sick and tired of his hippy kid playing that rock music at deafening levels. So, he had grabbed our hero, dragged him to a barber shop and had his hair cut. He ended up with a crew cut, of which there has never been any cut crewer, before or since. I doubt there was a single follicle exceeding a sixteenth of an inch.

At his next visit, we didn't even recognize him. He said of his fate, "Well, I guess you guys can call me Chrome-dome." The name of course, immediately stuck.

This visit was not all that humorous. He had brought back the speakers, which he thought didn't sound as clear as they once did. A quick listen revealed the problem: the tweeters were very thoroughly fried. We replaced them no charge and advised him that he should exercise some caution. He went away happy, while we tried to figure out how he had managed to fry two tweeters.

A week later, our friend was back; same problem. We installed new tweeters. This time he was required to pay for them, since the problem was not with the equipment. At $36 a crack, this was beginning to become an expensive lesson to learn.

But, the next day, he was back again. This time he brought his amp along. I checked it out, just to make sure it was not unstable or oscillating or any such thing. Well, we had warned him, and told him to get his $72 ready. I retired to the lab to replace the tweeters.

When the new units were installed, I tested the speakers. There was no high-frequency output still, and the midrange sounded strange. I quickly checked the feed to the tweeters and found nothing. So out came the woofer and all the damping, and there was the crossover before me, or what remained of the crossover. Both high-pass capacitors were gone. Simply not there anymore; totally *in absentia.* One of the midrange caps looked more like a sausage than an electronic component. There was the unmistakable odor of burnt plastic and the rear wall of the enclosure looked like it had been hit with shrapnel. And a lot of aluminum and paper confetti was just floating about.

Well, we put Chrome-dome on the hot seat. "What happened?" After a few minutes of questioning, he revealed he liked to play his harmonica along with the music, and conveniently, the Yamaha had a microphone jack on the front panel. Well, visions of infinite feedback squeal danced in our heads. Sure, I thought, that might be enough abuse to send an amp so far into clipping that it might destroy some tweeters.

He even agreed to demonstrate. We, instead, suggested that if he must play his harmonica, we would show him how it should be done. He hauled out his microphone and we plugged it in, and carefully turned the volume control to the level he wanted it, about two-thirds full.

He said, "No, that's not how I do it." Then, he removed his microphone, turned the volume all the way up and proceeded to slam the mic plug back into the jack. There was, of course, a deafening "ka-boom," accompanied by a sharp but muffled "crack." Simultaneously, there was the unmistakable ring-like flash of light around the periphery of the tweeter domes, signaling that the tweeters in our floor models had gone to join their compatriots in some direct-radiator nirvana. A few seconds later, we detected the strong odor of burning plastic, tempered by a slight acid smell. Our floor models were destroyed. The capacitors had been blown apart, and one of the crossover circuit boards had fractured.

And there, next to the Yamaha, which had shut down in protest, was Chrome-dome, smiling a toothy grin, saying, "Wasn't that awesome?"

THE BAXANDALL AMPLIFIER TESTER

BY REG WILLIAMSON

Peter Baxandall's unique testing procedure (*SB* 5/88) is characteristic; an impressively elegant example of careful thought about the parameters that determine how well an amplifier will perform with a variety of load conditions. It has long been accepted that a high proportion of the subjective aberrations degrading the performance of an ostensibly well designed amplifier could be categorized as overload *in one form or another.*

This includes the misbehavior of over-

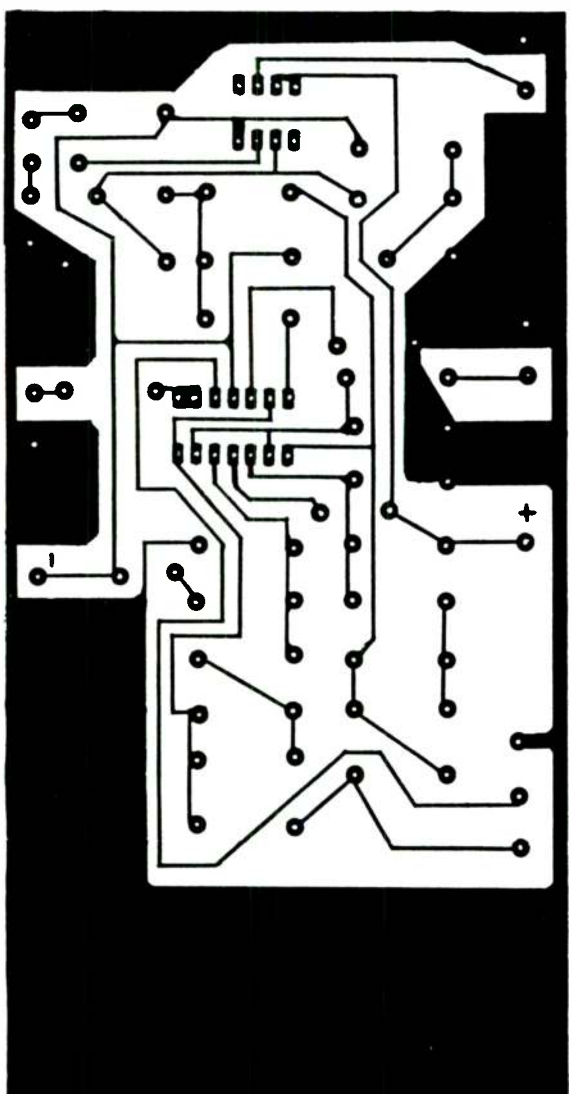


FIGURE 1a: Circuit board pattern, waveform generator.

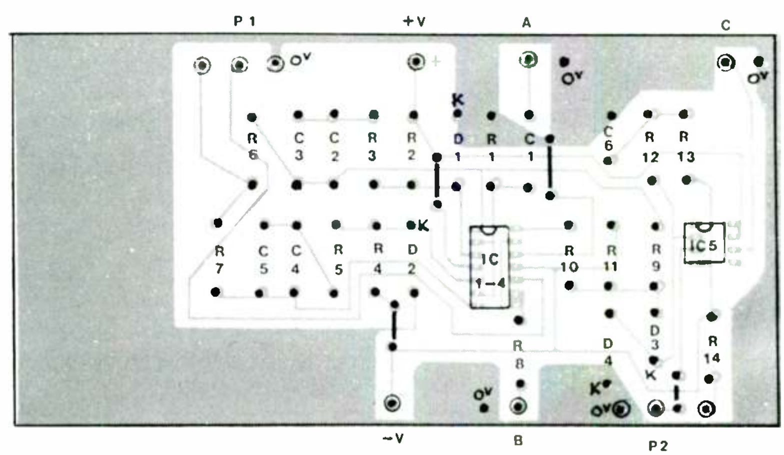


FIGURE 1b: Parts placement.

PARTS LIST

WAVEFORM GENERATOR

R1	4.7k	C1	220p
R2	100k	C2	33p
R3	56k	C3	470p
R4	100k	C4	470p
R5	56k	C5	33p
R6	10k	C6	1μF
R7	10k		
R8	2.2k	D1	1N4148
R9	10k	D2	1N4148
R10	10k	D3	1N4148
R11	8.2k	D4	1N4148
R12	10k		
R13	120k	P1	10k lin
R14	15k	P2	10k lin

4 links

IC 1-4 TL 084
IC 5 TL 081

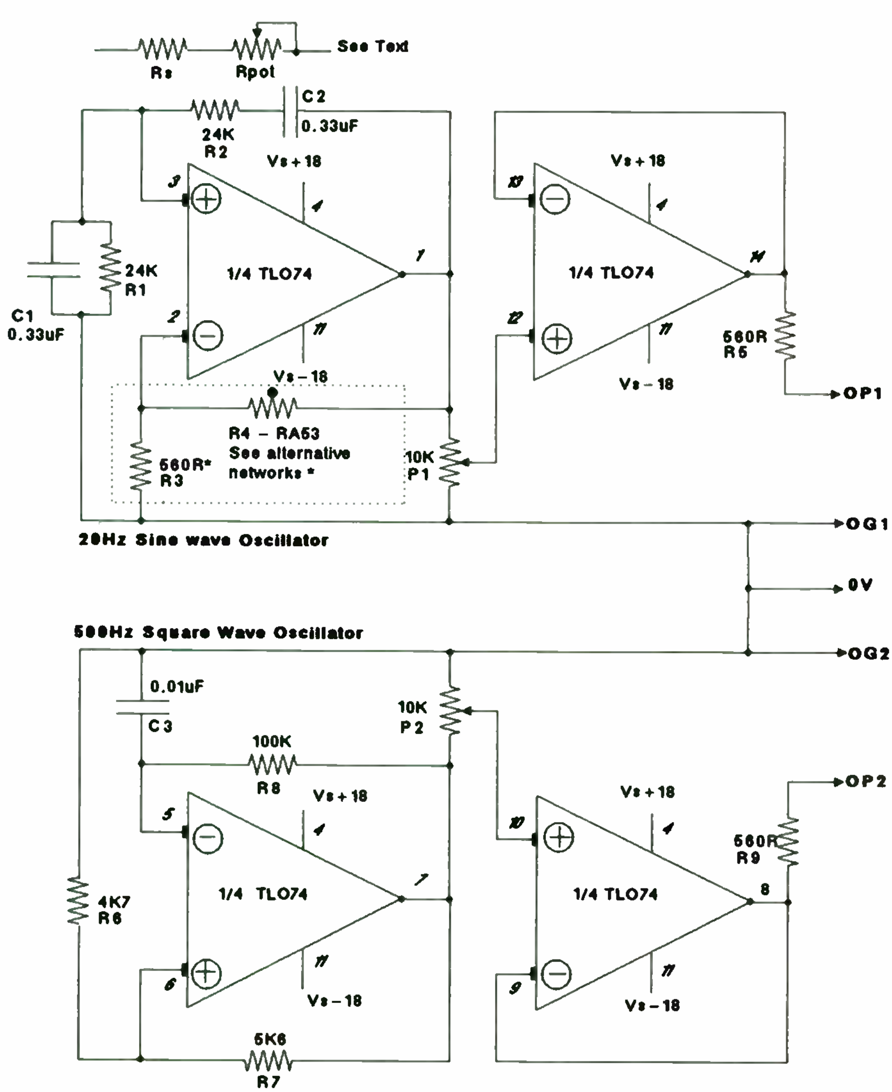


FIGURE 2: Schematic.

load protection circuitry. Any amplifier that survives Peter Baxandall's test procedure unscathed can begin to lay claim to being a top quality design and understandably, our Editor sought to have it published in *SB*.

For building Baxandall's tester, I now present this design, made easier for the home constructor using a circuit board layout (*Fig. 1*) devised by my colleague, Alan Watling. However, not many enthusiasts will possess the signal sources required. One perhaps, but not both. So, I have designed a little outrigger unit (*Figs. 2* and *3*) that incorporates the two, and which can be run from the tester's power supply.

It consists of separate square wave and sine wave generators, each with a gain adjustable buffer amplifier. Both use conventional design techniques, but the sine wave generator provides for some variation in construction and facilities.

Like most Wien bridge oscillators, it requires some form of amplitude stabilization. The ideal is a thermistor in the degenerative side of the bridge (as shown). The penalty one might have to pay is that, dependent upon the tolerance of the thermistor, the output under load may be only 800mV. However, I know of no transistor amplifier that has a sensitivity as low as this. Additionally, a suitable device is costly and since ultra low distortion is not essential, we have the option of two alternative networks (*Fig. 4*), one exploiting the varia-

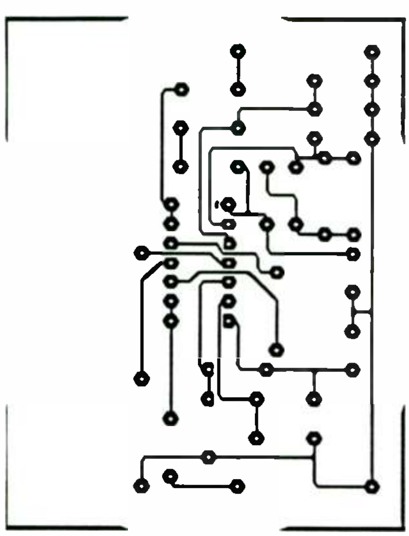


FIGURE 3a: Supplementary circuit board pattern.

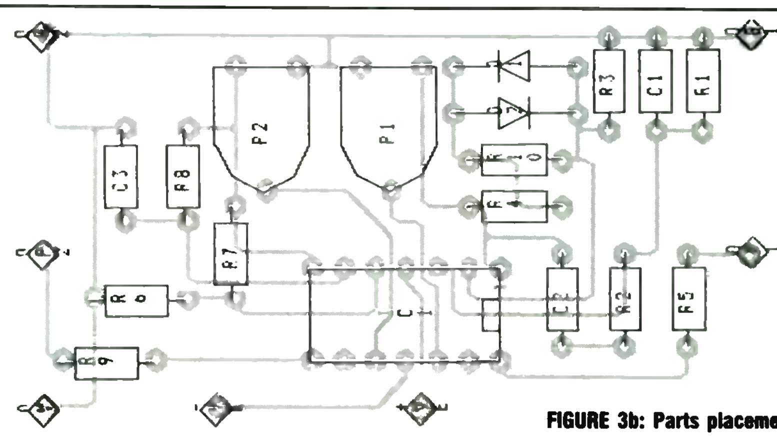


FIGURE 3b: Parts placement.

PARTS LIST

OSCILLATOR

R1, 2	24k 2%	P1, 2	10k trimmer type
R3	560R, 4.7k or p.t.c. device*	C1, 2	0.33μF min. polyester layer 10%
R4	RA53, 8.2k or 680R*	C3	0.01μF min. polyester layer 10%
R5, 9	560R	IC1	TL074
R6	4.7k	Si diodes	(2) 1N4148 or equiv.*
R7	5.6k	PC board, sundry solder pins, DIL8 socket	
R8	100k	*See text	
R10	1.5k*		

tion in resistance of a parallel diode/resistor combination as a function of the voltage across it. The other uses an inexpensive PTC device instead of the thermistor.

The diode combination has one virtue over thermistor stabilization: there is no bounce. However, it does require close tolerance components and even if all the resistors in the bridge are 1%, there is the remote possibility that R3 may require some small adjustment.

The other alternative has much to commend it, yielding a much higher output from the sine oscillator and is also very cheap. The PTC devices I selected were two subminiature panel lamps (wire-ended) 28V at 24m/A. With this type of stabilization, the output was 2.8V RMS at only 0.08% THD.

It is also likely that you may wish to trim the frequency away from its nominal 20Hz. Peter Baxandall suggests this might be useful, in his article, so this feature can easily be incorporated by substituting, for both R1 and R2, a variable resistor in series with a fixed one. The latter is required to set the maximum frequency with the variable resisitor at its minimum value. Obviously, you will insert solder pins in the holes

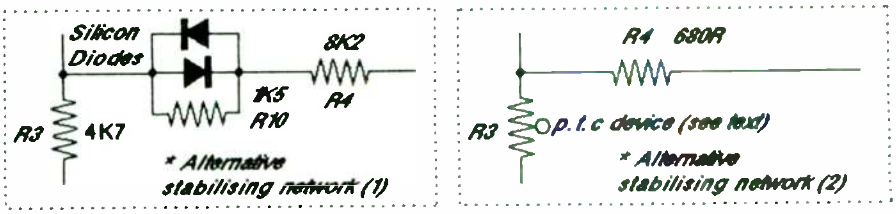


FIGURE 4: Two alternative stabilizing networks.

provided for R1 and R2.

The calculation of these two resistor values R_s and R_{pot} is quite simple. Let's assume we want to cover a range from 20–100Hz. It should go no higher than 100Hz, by the way. The value of R_s can be derived from this simple calculation, where F_{low} and F_{high} are the upper and lower frequencies of the selected range:

$$482{,}288/F_{high}$$

which, if we choose 100Hz, comes to a value for R_s of 4.829kΩ. Then, to find the maximum value of the potentiometer we are going to use for the variable element, the following applies:

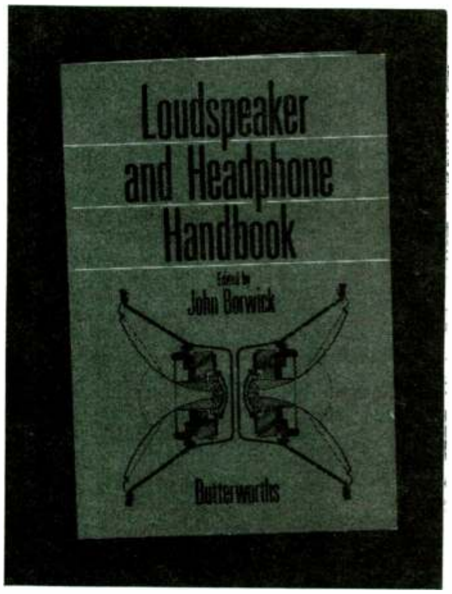
$$(482{,}288/F_{low}) - R_s$$

which works out to 19.285kΩ. Respectively, the nearest preferred value is 4.7kΩ and 20kΩ, the latter being the value of each section of a ganged pot. You will find that a log (audio taper) law is desirable. In common with good practice, it may be advisable to decouple both positive and negative VE power supplies to ground with 0.1µF ceramic disk capacitors, wired on the rear of the boards as close as convenient to the ICs. I did not provide for these components in any of the layouts.

The output of the square wave generator is well in excess of that required by the tester. The preset should be set initially to its minimum value, then slowly raised until the astable input of the tester starts to function with the appearance of the 50µsec pulses. Thereafter, it should not require further adjustment.

Book Report

An Audio Treat



Reviewed by Peter Muxlow

Loudspeaker and Headphone Handbook, *edited by John Borwick;* Butterworths, 1988; 573 pages. Available from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467; price: $97.50.

The publication of this new book on loudspeakers is cause for celebration. John Borwick, editor of *Loudspeaker and Headphone Handbook*, is the audio director of the English periodical, *Gramophone*. He asked a number of leading audio authorities to write a chapter each on a specific subject. Every chapter is self-contained, complete with its own references and bibliography.

The first chapter, by R. Ford, "Principles of Sound Radiation," is a theoretical overview of sound waves, radiation and electromechano-acoustical analogs.

The next chapter, "Transducer Drive Mechanisms," by Stanley Kelly, starts with a historical overview but deals mainly with the most common transducer, the moving coil driver. It looks at the diaphragm, the suspension, the motor unit and the magnet, including three good pages on practical magnet measurements. Also covered are horn drive units and ribbon loudspeakers.

A chapter on "Electrostatic Loudspeakers" is written by Peter Baxandall; he should be familiar to *SB* readers from his recent article, and he was also interviewed for an *Audio Amateur* article (4/79, 1/80). This is an in-depth study of electrostatics, and highly mathematical. It analyzes areas I have never seen published before, including effects of damping, radiation impedance of baffled strip radiators, and so on. Baxandall also does a specific analysis of both Quad loudspeakers, the early model and the EL63—great stuff, but I believe it's only accessible to advanced ESL buffs.

Laurie Fincham of KEF writes on "Multiple Drive Loudspeakers." He discusses crossover network theory and gives a good summary of the pros and cons of the different types of networks. Practical design procedures are also included.

Martin Colloms, the author of that fine book, *High Performance Loudspeakers*, contributes with a look at the "Amplifier-Loudspeaker Interface." He discusses the up-to-date advancements of oxygen-free copper cable, conjugate impedance compensation and active loudspeakers.

"Loudspeaker Enclosures" is covered by Desmond Thackeray. I believe this chapter could have been extended. It omits any coverage of the transmission line, the Isobarik, and Fincham's and Bose's band-pass enclosures.

An easy-to-read account of the "Room Environment" is provided by Glyn Adams. He covers basic room acoustics and measurements, loudspeaker placement, and living room design.

"Sound Reinforcement and Public Address" is Peter Mapp's subject. A contributor to English hi-fi periodicals, he examines types of coverage, the inside and outside environment, equalization, speech intelligibility, time delay and the Haas effect.

The only American contribution comes from Mark Gander of JBL, who covers pro sound in "Loudspeakers for Studio Monitoring and Musical Instruments." In comparison with some of the others, I found this chapter, like Thackeray's, too short to be comprehensive.

John Borwick contributes a chapter on the current state of the art in "Loudspeaker Measurement."

"Subjective Evaluation" is a discussion of program material for evaluation of loudspeakers, listening rooms, stereo/mono tests and experimental procedures for listening tests. The author, Floyd Toole, of the acoustics section of Canada's National Research Council, is a world authority in this field.

C.A. Poldy, from Austria's AKG, writes on "Headphones." To my knowledge, this chapter is the only extensive account of recent headphone technology. He covers isodynamic, moving coil, electrostatic, electret and electromagnetic types, and explains circumaural, super aural, closed and open types of design. Poldy lucidly explains the hearing mechanism which relates to headphone design—sound localization, out-of-head localization, and so on. A very good practical section includes testing methods, and 178 references are listed at the end of the chapter.

The last two chapters are a comprehensive survey of "International Standards" and "Terminology," by J.M. Woodgate.

SUMMARY. The thirteen chapters, with extensive references and bibliography, offer a good variety of subjects with well-designed text and diagrams. It is apparently aimed toward the level of *JAES* readers, with some parts highly mathematical. With so many contributors to this book, it is probably inevitable that some chapters are more comprehensive than others. However, I believe it is hard to justify 90 pages on the ESL, while only 16 pages are devoted to loudspeaker enclosures. That aside, I believe it will become a classic reference, probably most valuable to more advanced *SB* readers. It is a must for anyone seriously interested in audio.

Software Report
Full-Featured CAD

by Gary A. Galo
Contributing Editor

PRODESIGN II, V. 2.5; Computer Aided Design Program for IBM Compatible Computers; American Small Business Computers, 118 South Mill St., Pryor, OK 74361; (918) 825-4844. List Price: $299. Source: Review copy submitted by manufacturer.

System Requirements: IBM PC/XT/AT or true compatible. 512K RAM minimum; 640K recommended. Graphics Display Card and Monitor. Dot matrix printer or plotter.

Optional Equipment: Math Coprocessor; Mouse or digitizing tablet.

ProDesign II is billed as "The Easy To Use CAD Package" and it certainly measures up to that claim. The self-explanatory menu system allows you to begin drawing almost immediately, with minimal use of the manual. Of course, there is much more here than first meets the eye, and the manual is a necessity if all of the features of this program are to be used to full advantage.

During the past year, we have seen several software manufacturers offer full-featured CAD (Computer Aided Design) programs at a reasonable price. Industry standards, such as AutoCad, cost thousands of dollars, making their use unlikely in home or small business situations. Now excellent programs are available for only a few hundred dollars. ProDesign II is an excellent example of such a package.

Since I have some experience with Generic CADD 2.0, I will make a few comparative comments along the way, where appropriate. I will have a complete report in the near future on Generic CADD 3.0, which is the current version.

ProDesign II comes with four disks and a 402-page manual. Also supplied are a quick reference card and a sheet illustrating the various hatching patterns supported by the program. A third card is supplied, but its function is a mystery at first. More on this later.

The printed manual is soft bound and the text appears to have been prepared on a laser printer. Although the "no-frills" manual is not as slick in appearance as some other software documentation, it is well written and complete, with five basic parts. The first introduces the program, which contains information on installation and setup. Second, there is a tutorial which, in seven lessons, gives a solid overview of the workings of ProDesign II.

The major portion of the manual is an alphabetical reference covering all the commands supported by the program. This third section covers both the keystroke commands and the menu commands. A fourth section, called "Using Prodesign II," gives special attention to some of the more difficult concepts, including Zooming, Rotating, Vectors, Grids, Overlays and Macros. Area and length calculations and scaling are also discussed. Finally, nine "Appendices" contain technical information on the program which, as the manual states, is not normally needed by the ProDesign II user. The manual is well indexed.

Installation

Getting started with ProDesign II is quite easy. Instructions are given for installation on both hard and floppy disk systems. I am admittedly highly biased when it comes to hard disks, but in 1988 I do not see how any serious computer user can survive without one. High quality drives, such as those from Seagate packaged with Western Digital controllers, have become so inexpensive (under $300 for 20 Meg; under $350 for 30 Meg) that they are affordable by all but the most casual computer users.

Complex programs requiring several floppy disks are a nuisance on floppy systems. A hard disk not only spares you frequent disk changes, it also dramatically reduces the time required to read and write large files.

Installation on a hard disk system involves making a subdirectory called PD2 and copying the entire contents of three disks to it. ProDesign II supplies four disks. Disks #1, #2 and #4 are for systems without math coprocessors (8087, 80287 or 80387). Disks #3, #2 and #4 are for systems with a coprocessor. Although the Intel coprocessors are still extremely expensive, especially those for use in 80286 and 80386 based machines, I deem the speed increase to be worth the expense.

The PDSETUP program allows you to select American or Metric units of measurement. The manual refers to the choice as "English or Metric," but the English were much smarter than we Americans; they converted to the Metric system years ago, as we should have.

You are also asked which type of printer you intend to use. Using the three pages of printer choices, you should have no difficulty finding support for most printers in use today. Since the IBM ProPrinters are among the choices, any printer which is compatible with it (the Epsons and others can be set, via dip switches, to emulate the ProPrinter) should work properly if you select the ProPrinter type.

The printer output device must also be selected. ProDesign supports LPT1, LPT2, COM1 and COM2, thus covering both parallel and serial printers in any conceivable configuration. Similar choices exist for plotters.

Next, the mouse or digitizer tablet must be selected. ProDesign does not supply mouse drivers. You must use the driver supplied with your mouse. Both the Microsoft Mouse and the Mouse Systems PC Mouse (also sold as the IMSI Mouse) are supported.

All currently popular graphics standards are supported, including Hercules, EGA and CGA. Once the display standard is selected, you are asked to choose an aspect ratio. A default is given and normally you do not need to change it.

The final step involves aligning the command template if you have a digitizer tablet. The command template, as it turns out, is the mystery card mentioned earlier. If you do not have a digitizer, and ignore the instructions for supporting such a device, the function of this card will probably remain elusive.

After the setup is complete, load the program by typing PD. ProDesign requires a great deal of memory, You should have at least 512K, and preferably 640K. If the amount of free memory after boot-up is substantially less than 512K, a message will appear on your screen telling you that certain features of the program will not be supported. If you use RAM resident programs, it is essential that they conform to the MS-DOS standard for memory management. If not, unpredictable conflicts can result. In this case, I recommend using a program such as PopDrop to remove the RAM resident programs *before* ProDesign is loaded.

Using the Features

ProDesign II has a well-written, seven-lesson tutorial which will familiarize you with the most important and frequently used program features. The first four lessons involve drawing an automobile, and take you from the beginning through to printing a scale drawing. The lessons cover setting the drawing units; basic shapes, such as lines, arcs and circles; and proper methods for moving the cursor. The tutorial does not cover the use of a mouse; it assumes you will use the keyboard for cursor movement. The basics of zooming, stretching drawings, layering and dimensioning are also covered. You will undoubtedly find the dimensioning feature is one of the most powerful components of this program.

Only one paragraph in the tutorial contained either a misleading or incorrect instruction. On page 33, paragraph 5, you are told the first point in this step will be the same location as the third point in the previous step. In fact, it is at the same location as the *second* point in the previous step. The illustrations which accompany the tutorial are correct, however. The distance given to aid in the placement of the first point is also incorrect. Otherwise, I found the tutorial to be free of errors and easy to follow.

Lessons 1–4 of the tutorial concentrate on the keystroke commands. Lesson 5 introduces the menu commands and also introduces the various ProDesign II hatching patterns. Lesson 6 involves drawing a star, and introduces the INTERSECT, CUTOFF and ERASE commands.

Lesson 7 introduces the System and Drawing Parameters menus. This lesson, by itself, is not especially helpful. For a complete explanation of each menu selection, refer to the System Parameter and Drawing Parameter sections of the Command Reference.

The Drawing Parameter menu allows the selection of line colors, cursor step size, text size and angle, line type and width, and display grid size and width. The grid feature is an essential part of this program and is the only way you can determine exactly where the cursor is, or should be. I prefer that the grids be displayed as dots, but you can choose solid lines or border lines as well. There is a bug in the Generic CADD 2.0 grid feature. When you retrieve a previously made drawing, the grids do not always line up exactly as before. The ProDesign II grids have no such problem.

The System Parameter allows you to select the text character font, the units-per-inch of drawing output, and the type of angles to be displayed. The mathematical angle format puts zero degrees at the three o'clock position; the geographical format puts zero degrees at twelve o'clock.

Another important system parameter selection, the dimension format, allows six choices, including decimal, feet/inches and fractions. You can choose to save the system and drawing parameters with your drawing. This allows each drawing to contain a unique set of system parameters, which I find extremely useful.

ProDesign II uses two methods of selecting commands, keystroke commands and menu commands. Although many commands can be executed both ways, certain commands are unique to each format.

The menu appears at the upper right hand corner of the screen, to the right of the drawing window. Menu selections can be made with the mouse or with the keyboard's cursor keys. Using the mouse, you must move the cursor to the extreme right. Then, moving the mouse from top to bottom will allow you to scroll through the various menu selections. This differs slightly from the approach used by Generic CADD 2.0, which does not require moving the cursor to the right; the menu selections are scrolled whenever the mouse is moved from top to bottom. This may seem less cumbersome, but it also invites the possibility of a menu selection by accident.

When selecting a menu command, you begin at the root and your selection takes you down a path to secondary menus. The bottom choice on each secondary menu is "root menu," which makes returning to it extremely easy.

At first, the mouse seems to be the easiest way to make menu selections, but this also moves the cursor from its location in the drawing. To use the keyboard, you must first hit "enter" and then make your menu selection using the cursor keys. The two methods are both useful and, depending on your particular drawing and situation, you will find it is handy to have a choice.

The twenty-six keystroke commands are easy to use, by pressing a corresponding single letter of the alphabet. There are also sixteen ALT combination, ten function key commands and full use of the punctuation characters and all of the characters normally found above the numeric keys at the top of the keyboard. In addition, ten ALT-function key combinations are used for user-definable macros. The macro feature is very powerful, and can save many extra keystrokes.

A status line always appears at the top of the drawing window, with three choices. Status line #1, the default, shows the number of points that have been set, the current layer and the current zoom factor, among other items. Choice #2 also shows the number of points set, along with the distance from the cursor to the last point, and the angle of the cursor from the last point. If the cursor is moved diagonally, the horizontal and vertical distances from the last point are displayed. A third choice also displays the number of points set, and gives the cursor's X and Y coordinates.

I find myself using Status Line #2 consistently, because the distance displays make this the most useful. Status Line #1 automatically appears when the drawing window first appears on the screen. Unfortunately, you cannot change the default or save a status line with a drawing. If you prefer another status line, you must change it every time you use the program. A simple keystroke, the letter "J," facilitates the change.

ProDesign II supports a complete array of drawing features, including circles, lines, boxes of various types, curves, arcs and ellipses. Advanced features, such as filleting and chamfering, are also supported. Thirty-nine hatching patterns are included. Up to twenty-one layers may be used with each drawing. When the print program is run, you can select specific layers for printing. I find the menu structure is very good, and many of the commands are intuitive in nature. The advanced commands are not quite so obvious, and a few command paths through the root menu are not readily apparent, but these are exceptions. I believe the beginner can begin useful work in a very short time.

Setting points is extremely easy. The "0" (zero) key or the first button on the mouse can be used. Normally, I find it very difficult to press the button without moving the mouse, which results in a different point location from what you intended. I use the mouse for large cursor movements, the cursor keys for "fine tuning" and the "0" key for setting points.

Two of the most important functions use the period (".") and comma (",") keys. These are the gravity point commands, and they return the cursor to the nearest previously set point, which is essential for exact positioning. The period sets a new point in the same position as the old one, which is most useful for starting a new line or shape.

The ERASE command will remove any line or entity from your drawing. If the wrong object is erased, or you change your mind, the OOPS command will restore what you just removed. ProDesign II also has excellent zooming and rotating features. The Zoom feature is essential if you are making large drawings with many smaller internal details.

To automatically put a dimension into a drawing, you simply mark three points. Two points are placed on either end of the line to be dimensioned and the third point is placed where the dimension will actually appear. Hit the enter key and the dimension, along with arrows pointing to the two extremes, rapidly appears. If you have carefully watched status line #2 while preparing your drawing, the calculated dimensions will be exactly as you had envisioned. You can even calculate the area of any space.

Output

Printing your drawing is a simple process. If you are already in the ProDesign program you can simply select PRINT from the menu (or hit F5) to activate the print program. Otherwise, type PDPRINT at the DOS prompt to activate the print program. The program begins by listing six options, one which lists all of the PD2 drawings on your disk (or in a given subdirectory). After you select from the first menu, a second menu appears which allows you to select the drawing resolution, drawing size, rotation, and so on. Three modes of resolutions are supported; normal, high and very high. The output quality on a 9-pin Epson FX-286 is excellent. Certainly a laser printer would be better, but a 9-pin dot matrix drawing can still be considered "camera ready" as far as publication quality is concerned.

The lower resolution modes are useful to see how a drawing will appear on the page and print much faster than the high resolution mode. The print program supports the math coprocessor, if you have one. Since the CPU must periodically stop and process new data, the coprocessor decreases the printing time by over 50 percent. The IBM and Epson printers are actually capable of four modes of resolution, but since ProDesign II supports the highest mode, I do not believe the omission of one of the middle modes is any problem.

Generic CADD's Dot Plot option supports all four modes, but I did find a bug in Generic's program. Long diagonal lines are printed with one or two breaks in the line. The breaks are very small, but visible. The ProDesign print program has no such bug. Generic CADD may very well have corrected this problem, but I don't have the new version 3.0, to check.

ProDesign II comes with several impressive sample drawings to show off the capabilities of the program; my favorite is the Space Shuttle. Strangely, I saw an inexpensive CAD program, CCS Designer, in *Computer Shopper*, which uses what appears to be the identical shuttle drawing in their ad. Perhaps it was originally done with AutoCad, and converted to these other formats with a DXF translator program. This is unsupported speculation, of course, but the situation is rather curious. Both ProDesign and CCS Designer offer a DXF format translator as an option.

One minor difference exists in the way ProDesign II and Generic CADD 2.0 begin. ProDesign II puts the drawing window on the screen immediately. If you begin making a new drawing, you are prompted for a filename when you invoke the SAVE (or F10) command. Generic CADD prompts you for a filename *before* the drawing window appears. If the filename already exists, the previous file is displayed on the screen. If the filename does not exist, you are asked if it is a new filename. If you answer "yes," the drawing window appears and you are ready to begin a new drawing. I prefer the ProDesign II approach.

Both ProDesign and Generic CADD offer electronics symbols libraries. Generic's version offers over 50 symbols, while ProDesign II contains 38. New symbols can easily be created, however. Generic CADD retrieves symbols as components. ProDesign treats each symbol as a drawing.

ProDesign II is an excellent program. It is easy to use and comes equipped with the features most CAD users are likely to need. At the beginner level, many drawing commands are intuitive. The more advanced ones are not, however, and you will probably refer to the command reference card frequently, at first. With a program loaded with so many features, not everything can be immediately obvious. ProDesign II's help menu, which simply duplicates all of the commands on the reference card, is not particulary useful. I'd rather leave the drawing on the screen and refer to the card for help.

ProDesign is certain to be of enormous value to anyone with a need to create two-dimensional drawings. Authors and publishers alike will appreciate the ability to create drawings ready for printing, eliminating the need to have sloppy, hand-drawn diagrams re-done by a third party. I prefer ProDesign II to Generic CADD 2.0. How Generic 3.0 stacks up remains to be seen.

ProDesign II Update

I have recently received the update to ProDesign II, which is now called Design CAD 3.0. The new version offers a substantial increase in features, including libraries featuring over 500 symbols—ten libraries of architectural symbols, two libraries of cabinet making symbols and one each of electrical, electronics and piping symbols. 100 new features have been added to the program.

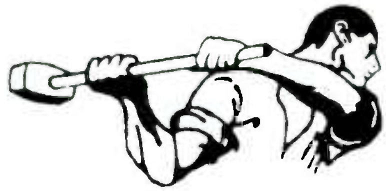
The improved documentation now includes separate installation and tutorial manuals. The quick reference card is also easier to use, with each command identified by its menu and keystroke forms. Design CAD 3.0 supports the Lotus-Intel-Microsoft Expanded Memory Specification. It automatically allocates 640K of expanded memory on my PMI FastCard IV when the program is executed. The setup program has been greatly improved. It is now menu driven and you no longer have to go through the entire setup to change one item.

Ten disks are supplied with Design CAD 3.0, although only nine are needed, since there are two versions of the Program Disk, one with coprocessor support, and the other without. Three disks contain the symbols libraries. This program will take up a sizeable portion of your hard disk. The only symbols I installed were those for electronics, yet the program takes up about 1.8 megabytes of disk space. You'd better have a hard disk to run this program. Floppy disks will make a mockery of a sophisticated program such as this one.

Design CAD 3.0 offers support for the new IBM VGA standard. I'm using it in the VGA 640 by 480 mode on a Zenith ZCM-1490 Flat Tension Mask monitor supported by a Zenith Z-449 display card. The program is obviously well behaved, and makes no VGA hardware calls. My Z-449 display card is only BIOS compatible with VGA, but Design CAD runs without problems.

The update to ProDesign 2.0 is offered for only $30, which includes a new set of manuals. American Small Business Computers is a firm that obviously cares about their customers. *SB* readers will find Design CAD 3.0 to be an ideal solution to their drawing needs. This is a superb program for loudspeaker diagrams.

This program can be obtained from many mail order firms for less than $150; try Telemart, 8804 North 23rd Ave., Phoenix, AZ 85021, 1-800-426-6659. At the list price of $299 it is an excellent buy. At the "street price," it is simply one of the greatest bargains in computer software.



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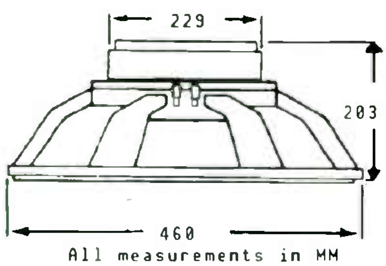

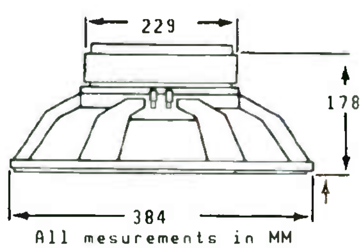

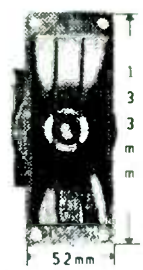

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VB: 200 Liters
FB: 45 Hz.
F3: 48 Hz.
Vent Diameter 6" Vent Length 4.5" Use two vents.

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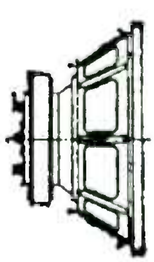

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Fast Reply #KC114

Tools, Tips & Techniques

POSTS REPLACE SPRINGS

I built my VMPS subwoofers from a kit, and I have been quite happy with these speakers with one exception. The kit included spring-loaded connections for the speaker wires coming from the amplifier. This type of connector has a hole for the wire and a spring-loaded button, which when depressed, moves a piece of metal aside to open the hole.

You depress the button, insert the wire, and release the button, allowing the spring to press the metal against the wire. The metal is connected to the terminal inside the enclosure, to which the driver and crossover wiring is soldered.

The movable piece of metal makes a line contact with the wire, and you cannot tighten this connection beyond the pressure exerted by the spring. I have always thought this arrangement conducts less than the maximum amount of signal from the amplifier to the speaker.

I contacted VMPS about my worry, and they obligingly sent me two pairs of gold-plated, color-coded, five-way binding posts to replace the spring-loaded terminals. The VMPS crossover and speaker terminals are mounted on a preformed, flanged, plastic plate that fits in a cutout on the back of the enclosure. The spring-loaded terminals appear to have been heated and pushed through the plastic plate.

I couldn't see any way to remove them to make space for the new binding posts, so I elected to install them in another location. The top of the plastic plate, between the binding posts for the satellite speaker connections and the satellite attenuator knob, has adequate space—both internally for the wiring connections, and externally to connect the wires from the amplifier.

First I removed the eight screws holding the plastic plate to the enclosure. I inserted a dual banana plug in the two new binding posts to fix the spacing of the posts. I positioned the binding posts on the plastic plate and marked the location

using a marker with silver metallic ink (to show on the black plastic plate). I then selected a drill bit as close as possible to the diameter of the section of the posts to be inserted through the plastic plate, to minimize air leakage. I carefully drilled the holes at the locations I marked, without damaging any wiring on the back of the plate. After cleaning away the burrs on the plastic, I installed the new binding posts, securing each with a nut and a toothed lockwasher.

I believe in making strong mechanical connections before soldering, so I did not merely unsolder the connections from the spring-loaded terminals. I cut the wires from them and removed a section of insulation from each wire. I worked with one terminal at a time to avoid errors in making the new connections. I resoldered the wires to the new terminals, after adding short sections of wire to reach the new terminal location. I then reassembled the first speaker, taking care with the seal under the plastic plate to keep it airtight, and tested the speaker to be sure it still functioned properly.

I then made the same modification to the second subwoofer, which took half the time.

Was the modification worth the effort? Certainly, from the standpoint of peace of mind. I believe a connection that can be tightened has a better chance of working consistently well than a spring-loaded one.

Sonic memory is very short for absolute comparisons, but an extended listening session using selections I had been listening to previously, indicated an improvement in imaging and also in bass definition and tightness. All of the electrons to both subwoofers and satellites must pass through these terminals, so I wasn't surprised that the satellite performance was also improved by the modification. I think any speakers with spring-loaded terminals will benefit from substituting good quality binding posts.

James T. Frane
Orinda, CA 94563

NEW LIFE FOR OLD SPEAKERS

I would like to relate my recent experience with a pair of speakers that seemed to be going bad. This anecdote may help others thinking the time has come to buy new speakers, to conclude that less costly alternatives are worth exploring first.

I own a pair of five-way electrodynamic speakers that were considered to be among the best audiophile-quality speakers available about ten years ago. Some time ago I made modifications, with new wiring, crossover changes, and replacement of three of the drivers with better units, including the popular Audax dome tweeter. Meanwhile, the speakers were sometimes played at loud volume, but never at abusive levels.

Upon moving from our old house where the speakers were used in a rather large living room in competition with furniture, to our new house where I installed them in a smaller but dedicated listen-

Continued on page 58

Tools, Tips & Techniques

Continued from page 56

ing room, I noticed how harsh they sounded. Thinking this was caused by the acoustic nature of the room, I borrowed some acoustical foam and liberally treated the walls behind and to the sides of the speakers. While this ameliorated the problem, it did not cure it. Upon closer listening, the Audax tweeters proved to be the source of the harshness. At Bruce Edgar's suggestion, I tried placing foam "rings" of various geometries around the tweeters, which also helped but did not cure the problem.

Years ago in yet another, smaller location, the speakers had sounded very good. Thinking that my listening ability had evolved beyond the actual ability of these speakers in the intervening years, I started to consider purchasing new drivers.

Instead, Bruce Edgar suggested I replace the Audax drivers with another brand, which would involve mechanical and electrical changes as well, or try brand new versions of the same Audax drivers. Then, at the 1988 Santa Monica Stereophile Show, Larry Hitch of Madisound suggested replacing the voice coils and integral domes with new units, told me how easy it was, and later supplied new ones for a very reasonable price. The installation is quite simple.

With this minor surgery my speakers were reborn. The harshness vanished, replaced by a seamless smoothness. But I now noticed that they sounded thick and muddy. Removing the foam from the rear wall cured this problem. The foam was absorbing the treble to the point of drastic attenuation. With the new voice coils, wall foam only on the side walls, and the foam "rings" removed, the system sounds excellent; open, airy, spectrally balanced, detailed, and smooth.

Upon examination, the old domes showed discoloration relative to the new ones; they also had some tiny pinholes in parts of the surround. The old domes had been exposed to the ozone of the Pasadena area for years, which could not have helped them. In addition, they were occasionally asked to perform at high levels with some very demanding material. So, if you think you need a new pair of speakers because your old ones don't sound as good as they used to, or they make a poor showing of themselves compared to what you just heard at the CES or your local dealer, it may be because your old driver elements have undergone a slow decay, the effects of which you have suddenly clearly noticed for the first time. It may be that only minor surgery rather than a major transplant is needed.

Dave L. Glackin
Altadena, CA 91001

Technology Watch

Subwoofers, Part I

By Peter Muxlow

With the introduction of digital recording and the availability of CDs, the importance of low-frequency sound and subwoofers is being re-examined. In a recent report, Fielder and Benjamin auditioned a number of CDs and concluded that a response down to 16Hz was needed for accurate reproduction[1]

Celestion, an English company, has an unusual solution to low-frequency reproduction[2] Their subwoofer is a double dipole—two dipoles face to face. A dipole is a loudspeaker that is unbaffled and radiates sound equally from the front and back. It has a figure-eight radiation pattern, the same as a cardioid microphone. The Celestion speaker physically couples two 12-inch drivers closely together, but pointing in opposite directions. The drivers are wired so they move in phase with each other. The two outputs *add* below 100Hz and the even-order harmonic distortion created by the drivers is cancelled because they point in opposite directions.

Now, with no baffle the low-frequency response starts to roll off, as you would expect. However, the trick with dipoles is the roll-off rate is only a gentle 6dB/octave; with a closed box it's 12dB/octave, and with a reflex enclosure it's 24dB/octave.

The 6dB/octave rolloff is compensated for by electronic equalization, which is supplied by a black box inserted between the preamp and the power amp. The black box has 20dB of bass boost and variable bass extension from 70Hz to 20Hz. An advantage of dipoles is the figure-eight radiation pattern does not excite as many room resonances.

Another method of pumping a lot of air is—wait for this—synthesis of loudspeaker mechanical parameters by electrical means[3] If we take a reasonably-sized cabinet, use reflex loading and tune to 20Hz, the driver would have impossible specifications. Stahl's ingenious answer is to use electronic circuitry to modify the apparent mechanical properties of the driver.

In any driver the electrical and mechanical sides of the speaker interact; for example, the bump in electrical impedance created by the mechanical resonance. Stahl uses this interaction to control the three mechanical parameters of the driver—the moving mass, the compliance, and the damping. He quotes an example of an amplifier driver combination where:

- moving mass is increased from 38g to 260g;
- damping is increased from 26 to 58kg/s;
- compliance is decreased from 0.45 to 0.25mm/n.

In essence, the complex output impedance of the driving amplifier, which involves inductance, capacitance, and negative resistance, controls the apparent mechanicai properties of the driver.

The only catch: you need a specially designed and dedicated amplifier.

1. Fielder, L.D. and E.M. Benjamin, "Subwoofer Performance for Accurate Reproduction of Music," AES 83rd Convention, Preprint #2537.
2. British Patent #2191065.
3. Stahl, K.E., "Synthesis of LS Mechanical Parameters by Electrical Means: New Methods for Controlling Low Frequency Behavior," *JAES*, Sept., 1981.

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

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Fast Reply #KC563

ALTERNATIVES TO THE POLK 10 MODIFICATIONS

BY JAMES T. FRANE

Since my article on modifying the Polk 10 speakers (*SB* 4/87), I have received correspondence from readers, which included suggestions and rationale for possible wiring configurations. I no longer own the Polk 10s, but I decided I could experiment with and compare other configurations just as well by using the Kaufman speaker configuration[1] with other systems.

In my original article, references to Polk's SDA speakers may have been somewhat misleading by diverting attention from the actual goal of my modification—enhancement of stereo reproduction. My circuit diagrams, rather than illustrating what Matthew Polk described in his *Audio* article,[2] showed that the −R component of the L−R signal from the outboard left speaker will cancel the +R signal arriving at the left ear from the right inboard speaker and, of course, the obverse situation for the right side of the system. It may be that interaural crosstalk, *per se* is not cancelled by my circuit, but imaging and ambience reproduction are noticeably improved over the stock Polk 10s.

Separate speaker enclosures placed side by side will perhaps work even better, because using this setup removes the variable of mid/woofer interaction with each other and with the passive radiator.

The main speakers I use are Siefert Research Maxim IIIs, placed atop a pair of VMPS Smaller Subwoofers. In normal configuration, each Siefert/VMPS pair reproduces a stereo channel. I run the Sieferts full-range through a level control on the subwoofers, interfaced by a passive crossover that rolls off each VMPS above 100Hz. I connected the 8Ω Sieferts and 8Ω VMPS's in parallel to present a nominal 4Ω load to the amplifier.

For the ambience enhancement (or interaural crosstalk cancellation speakers), I use Radio Shack Minimus 7s, placed beside and outboard to the Sieferts.

Figure 1 shows the same Kaufman configuration in which I wired the modified Polk 10s. This puts the 8Ω Minimus 7s in series, creating a nominal 16Ω load. When this is connected in parallel with the Siefert/VMPS combination, the resultant impedance is:

$$(Z1 \times Z2) / (Z1 + Z2) = Z$$

$$(4 \times 16) / (4 + 16) = 3.2\Omega$$

For future reference to this configuration, I'll call the Siefert/VMPS system without the Minimus 7s, System A; and the Kaufman wiring configuration, System B.

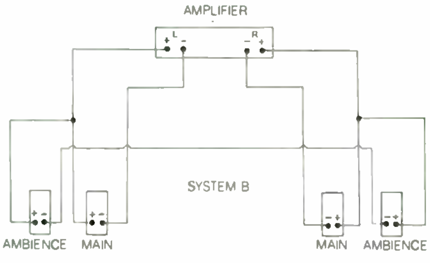

FIGURE 1: System B—Kaufman configuration using Minimus 7s for ambience speakers.

With a stereo source, the ambience speakers will produce only the difference signal between the right and left stereo channels, because in each configuration they are wired across the left and right channel terminals of the amplifier. The ambience speakers will reproduce no sound with a monaural source, since no channel difference is present.

Two cautions are in order before we proceed. First, not all amplifiers will tolerate the ambience speakers being connected across the positive terminals of the left and right channels. Second, not all amplifiers will tolerate the low impedance of the combined load of the ambience and main speakers, particularly at high listening levels.

Figure 2 shows a configuration that is the same as System B, except the ambience speakers are wired out-of-phase with the main speakers. This configuration will be referred to as System C. The calculated impedance would be the same as for System B.

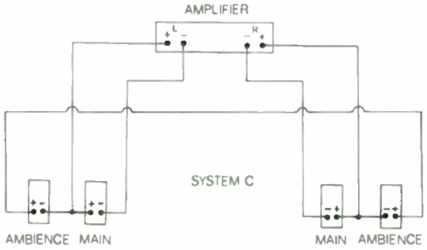

FIGURE 2: System C—the ambience speakers are wired out-of-phase with the main speakers.

Figure 3 shows another configuration, wired the same as System C, but the left ambience speaker is located next to the right Siefert speaker and vice-versa. This will be called System D.

TEST SETUP. I generated some frequency response curves for each of the configurations using Hi-Fi/Stereo Review's *Test Record*, Model 211 and an ADC sound level meter (SLM), Model SLM-2. I centered the SLM on the axis

1. Kaufman, R., "Build a Passive Image Enhancer," *Audio*, Nov. 1986.

2. Polk, M., "Polk's SDA Speakers, Designed in Stereo," *Audio*, June 1984.

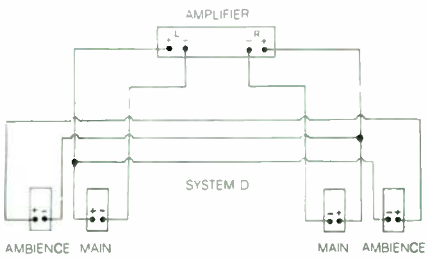

FIGURE 3: System D—the left ambience speaker is located next to the right Siefert speaker and vice-versa.

TABLE I

Freq. Hz	Mid-Freq.	SYSTEM A	B	C	D
14.7-20k	17,350	SLM range is 32Hz-8kHz, therefore this data is omitted			
10.3-14.7k	12,500				
7.36-10.3k	8,830	67	65	66	66
5.2-7.36k	6,280	69	68	69	68
3.7-5.2k	4,450	69	69	67	64
2.6-3.7k	3,150	66	64	67	62
1,840-2,600	2,220	66	65	67	61
1,300-1,840	1,570	64	65	66	64
920-1,300	1,110	66	65	66	63
650-920	785	72	71	70	71
460-850	555	69.5	71	68.5	70
325-460	392.5	72	70	72	73
230-325	277.5	69	68	67	70
160-230	195	62	62	60	65
115-160	137.5	68	67	69	69
80-115	97.5	72	71	73	73
40-80	60	70	70	70.5	70
20-40	30	68	66	67	67

of the right speaker system, one meter in front of the speaker, to minimize room boundary reflection effects (compared to placing the SLM at the listening position). I mounted the SLM on a tripod at a height halfway between the centerline of the VMPS front woofer and the Siefert tweeter. I set the meter in the "Fast" response position and the "C" weighted mode.

Table 1 shows the digitally tabulated data from these tests. This record uses warble tones. The first column in the *Table* shows the frequency range of each test band on the record, numbered 1–18. The second column shows the mid-point of each frequency range. The following four columns show the sound level meter readings for each of the system configurations.

Figures 4 and *5* show this data translated into response curves for easier comparison; drawn on a scale larger than is customary so that differences are more readily discernible. Both the digital data and the curves show the differences in frequency response are slight, but of course, this is only part of the picture. Frequency response alone cannot completely describe the sound you will hear; we must resort to listening.

I evaluated each system by listening to four different recordings.

CDs:
Simon and Garfunkel's Greatest Hits, Columbia CK31350;

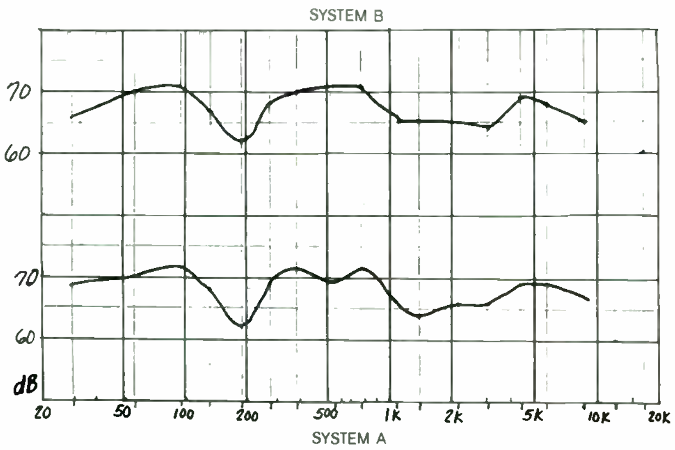

FIGURE 4: Measured frequency response for Systems A and B.

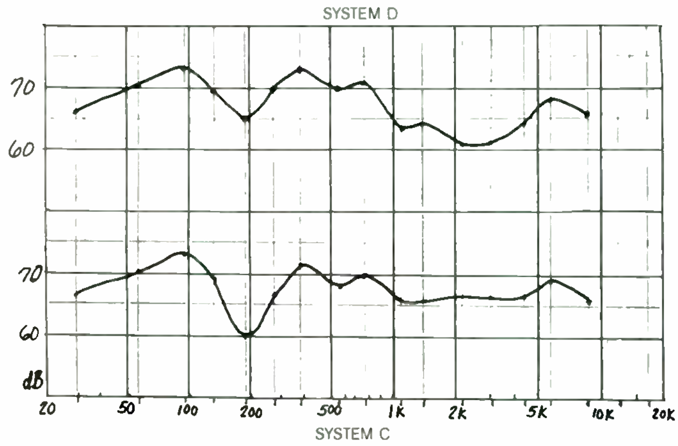

FIGURE 5: Measured frequency response for Systems C and D.

Holst, *The Planets*, Leonard Bernstein and the New York Philharmonic, CBS MYK37226.

LPs:
Paul Desmond and the Modern Jazz Quartet, Finesse Records FW37487;
Julie London Sings Cole Porter, Liberty LST-7434.

RESULTS. I listened to each source through each of the system configurations. My listening position was midway between the speakers at approximately 1½ times the distance between the pair.

Simon and Garfunkel

System A: The guitar is located to the right of the stage and cymbals to the left. Vocals are centered. Localization remains stable with moderate stage depth. No sound extends beyond the outside boundaries of the speakers.

System B: The sound from the guitar extends beyond the outside of the right speaker, otherwise the sound is the same as System A.

System C: The vocals now extend the full width between the speakers, is unnaturally wide. A sense of "phasiness" (out-of-phase sound), which although it adds ambience, gives poor localization. No sound extends beyond the speaker boundaries. The sense of stage depth is good. Cymbals move toward center stage.

System D: The sound from the guitar extends slightly beyond both speakers, which is far too wide. Vocals are also too wide. The phasiness is still present. I notice some tendency for the images to wander. Cymbals are on the left. The stage depth is decreased.

The Planets

System A: Good stereo spread with some sounds extending beyond the speakers. Stage depth is good and localization is stable. The relative widths of orchestra sections are realistic.

System B: Essentially the same as System A; no improvement or degradation.

System C: Orchestra sections become too wide and the images lack localization. The ambience is slightly increased, proportional to a sense of phasiness.

System D: Phasiness is enhanced and becomes annoying. The orchestra sections remain too wide and localization is unstable.

Paul Desmond

System A: There is a good spread of the instruments across the soundstage. Stage depth is good. Some sounds extend slightly beyond the speakers. Localization is good.

System B: Sound is essentially the same as System A, with added ambience, increased depth and more sounds beyond the outside edges of the speakers. Images remain stable.

System C: The saxophones become too wide. Ambience becomes unnatural with the out-of-phase sound. Soundstage depth is markedly decreased. Images tend to wander.

System D: The soundstage depth is partially restored. Images are too wide (even the cymbals assume a width that extends across the soundstage) and tend to wander. Phasiness adds to the ambience, but detracts from the realism.

Julie London

System A: Good ambience; good depth. Localization is stable and the vocal remains a proper relative width.

System B: Ambience is good, but slightly decreased from System A, as is the depth. Not an improvement over System A.

System C: The vocal moves from stage center to the right. Depth collapses into almost a two-dimensional image. The phasiness detracts fom the ambience.

System D: Phasiness detracts from the ambience. Both the vocal and instruments become too wide. Soundstage depth increases beyond System C, but not to the same depth as with Systems A or B.

SUMMARY. This experiment convinces me that most stereo recordings will be enhanced to some degree by the ambience speaker configuration described by System B. The ambience might also be increased using Systems C and D, but almost always at the expense of some of the realism. I attribute this to the ambience speakers being out of phase with the main speakers. System B is easy to set up and it will reward you with added ambience on some recordings.

LETTER WRITERS AHOY . . .

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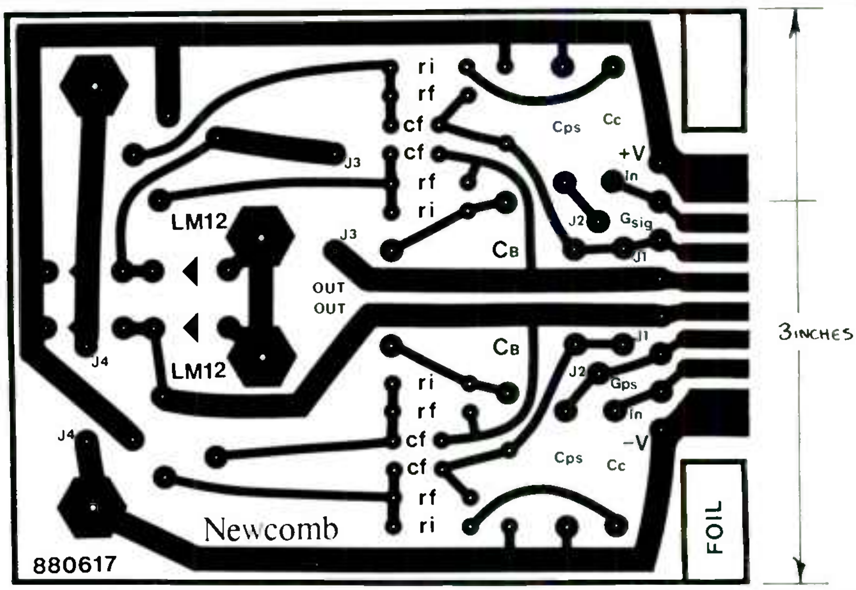
Please leave room in your letter for replies. Your questions should relate to the article, be framed clearly, and written legibly. Please do not ask for design advice or for equipment evaluations.

Letters to authors or other readers cannot be acknowledged, unfortunately. Any letter which does not comply with the requests above will not be answered.

LM12 MODIFICATIONS

I am enclosing corrected art for the large circuit board discussed in my article, "Sub-Bass Power Boosting and Other Applications Using The LM12" (*SB* 4/88). Also enclosed are corrected and/or clarified layouts for both boards and the schematic of *Fig. 6*, which showed the positive supply bypass capacitor improperly connected to the amplifier output. Sorry about the mixup.

Art Newcomb
Yorktown, VA 23690



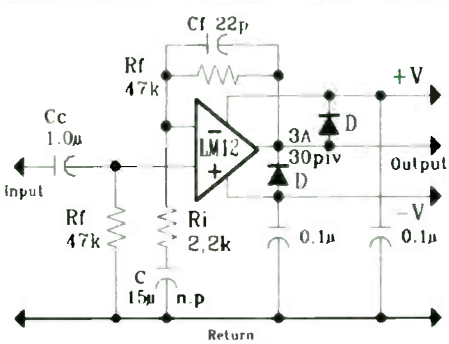


FIGURE 6

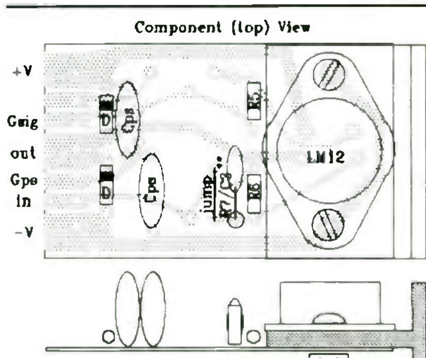


Figure 5

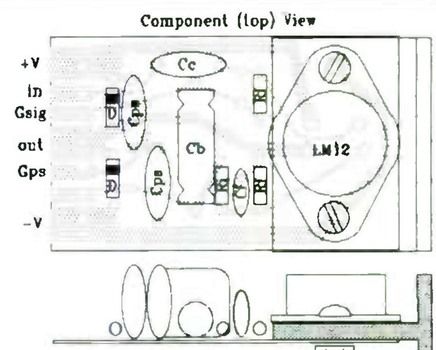


Figure 6a

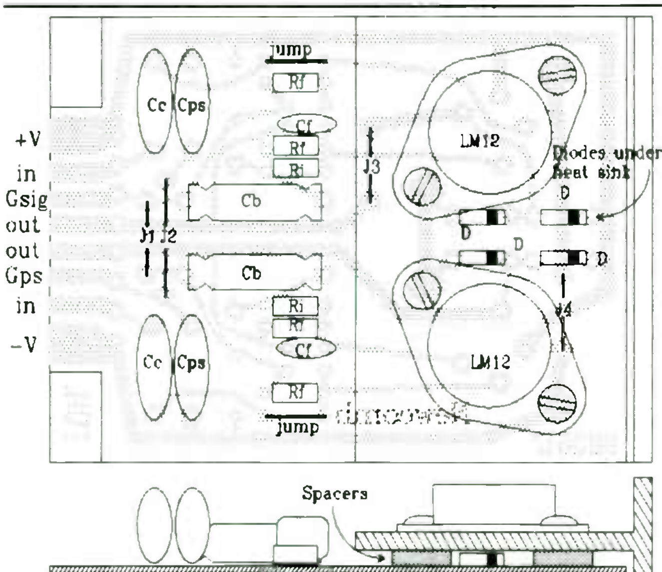


Figure 7a

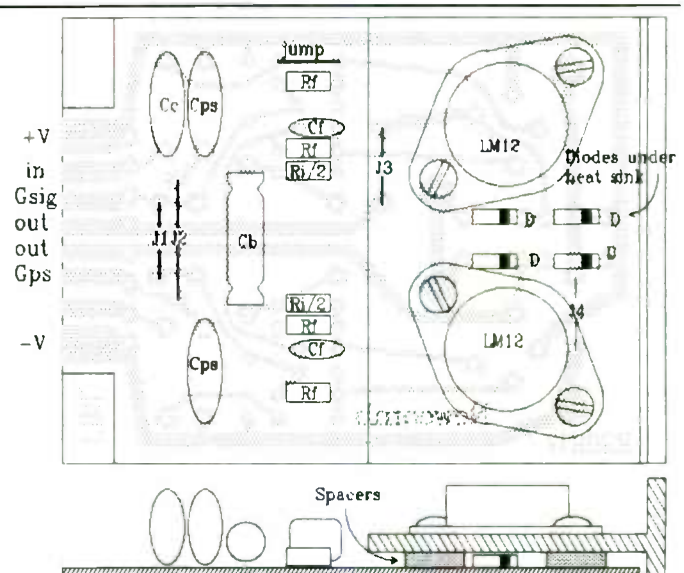


Figure 7b

MORE MINIMUS-7 MODS

Because of continuing interest in my article, "Modifying Radio Shack's Minimus-7" (*SB* 1/88) as evidenced by *SB* readers' letters, I would like to add the following to help others who might be thinking about writing to inquire about possible modifications to the little systems, other than those I mentioned.

Several readers asked about other crossover schemes, including the use of the Zobel network, as improvements to the speakers, and requests for a preference of one particular set of transfer functions over another (such as Gaussian, Butterworth, L/R, and so on). The answer is that any of these can be used, if two things are kept in mind.

1. Any crossover used on the little systems *must include a frequency response equalization function*. Failure to do this will result in a system amplitude response *worse* than that provided by the simple crossover I used.

2. Consider the following graph of frequency response and distortion of the Minimus-7 woofer (either the original system woofer, or the driver sold separately by Radio Shack, they both measure the same):

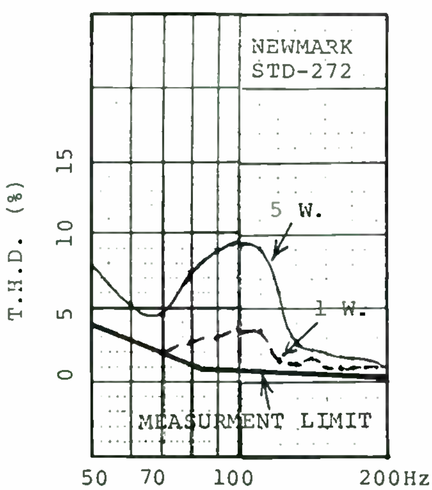


FIGURE 1: Measured THD of the Minimus-7 system's woofer at 1W and 5W input.

Shocked? Although the system sounds quite good, both the woofer and tweeter exhibit this high distortion, as well as poor phase and frequency response.

So what's the bottom line? It would be ridiculous to spend \$50–\$75 on passive crossover components, or \$300–\$500 on a biamp system, including amplifiers, for these little speakers that can be bought for as little as \$30 each. As modified in my article, the simple changes made are a carefully considered balance between cost and performance on one hand, and the drivers' built-in limitations on the other. Enough said?

But is this all? Some additional changes can be made to the systems, such as using them in multiples, or using an auxiliary baffle around the system's face. Both these types of additions can make some real improvements to the system's performance and are worth exploring.

William R. Hoffman
Reno, NV 89502

SWAN IV

I have the following questions about the Swan IV system detailed in the 4/88 issue:

The 32 by 16 by 16.5-inch cabinet dimensions suggest internal box ratios of approximately 2:1:1, while a ratio of 2.3:1.6:1 is generally thought to minimize standing wave problems. Does the limited bandwidth of this box obviate the problem? If not, how have the authors dealt with it?

I read in a supplier's catalog that Precision, maker of the TA305F woofer, may have gone out of business, jeopardizing the supply of these drivers. Based on incomplete published specifications, the Eton 11-580/50 Hex looks close in parameters, but is terribly expensive. What else might work with minimum design alterations, if the TA305F is not readily available?

It would be interesting and enlightening to know what other drivers were considered, and why they were rejected.

Stephen Katz, M.D.
Topeka, KS 66614

Contributing Editor D'Appolito replies:

Concerning your first question, in well-designed concert halls the distribution of room modes which produce standing waves is dense and uniform. This ensures that all frequencies are reinforced equally and that hall response is flat. This condition is much more difficult to attain in rooms typical of the home. For these much smaller rooms, modes are far fewer and more widely separated, producing an irregular room response. The ratios you quoted produce the most uniform (i.e. regularly spaced) low-frequency standing waves and have the added benefit of being non-harmonically related.

The 2.3:1.6:1 ratios are well-known in architectural acoustics, but their utility in loudspeaker enclosure design is limited. Loudspeaker enclosures are obviously substantially smaller than the rooms in which they are placed. The resulting standing-wave modes are of much higher frequency and all but the lowest of these are easily damped by proper box treatment. The production of standing waves in a small box depends not only on box shape, but also on driver placement in the box.

In the Swan IV bass module, the first standing wave, the vertical mode, is at 250Hz. To excite this mode, however, the drivers must be mounted in the top or bottom of the box. The dual drivers and port are distributed along the vertical face of the box in such a manner that the first mode cannot be excited. The second mode at 550Hz is well above the crossover frequency and is well damped by the box lining, as are all higher modes.

The problem of standing waves in loudspeaker enclosures is greatly exaggerated. By far the most important problem is enclosure panel vibration caused by the mechanical coupling of driver motor reaction forces into the enclosure walls.

As we noted in *SB* 5/88, we have found no suitable replacement 12-inch driver so far and it appears that the TA305s will not be available again for some time, if ever. We are currently recommending the 10-inch Eclipse 10W38R, available from Meniscus or Madisound.

Concerning your last question, our goal of 110dB SPL at 25Hz in a 100-liter box places extreme requirements on woofer Q, F_{sa}, displacement volume (V), and V_{as}. The numbers required are respectively 0.3, 22-25Hz, 275-300cm^2 and no more than 180 liters, with something closer to 160 liters preferred. These requirements rule out all 15-inch drivers and most 12-inch drivers.

In addition, we found that manufacturers may quote specs optimistically, or in nontraditional and possibly misleading ways. For example, Dynaudio quotes the peak-to-peak volume displacement, rather than the one-sided value used in the Thiele/Small equations. Using their number predicts 6dB SPL higher than you actually attain. The linear volume displacement of the 30W54 or 30W100 is actually less than that of many good 10-inch drivers like the Eclipse. The 30Ws are excellent wideband units, but remember that we limit the upper end to 200Hz. In this range there are less expensive drivers with substantially more low frequency output.

We did not mention the woofers we tried by name, because to do so would imply that they were not good drivers, when in fact most of them, like the Dynaudio, simply were not useful in our design.

Third Dimension

Continued from page 36

vent: $D_V = 8cm$; $L_V = 25.3cm$
P_A symmetrically/P_A conventional = $-6.67dB$.

I have built three different systems using symmetrical loading—the previously mentioned Opera; a small, 15-liter internal volume enclosure; and the 56-liter Delta. All have strong, clear and tight bass output. Among these, I prefer the Delta, which I shall describe in detail in my next article.

If you do not wish to make any calculations, I have included in *Table 4*, 27 drivers, with two possible alignments for each. I hope your favorite driver is included.

SWAN IV QUESTIONS

I read with much interest D'Appolito and Bock's Swan IV Speaker System article in *SB* 4/88. I had previously built Falcon's Rogers LS3/5a kit. I find myself in need of another small speaker system. It appears that the Symmetrical Satellite with the Treble Coupler crossover as specified in the article would fill the bill. Also I could upgrade later by building the woofers and active crossover.

Is this reasonable? Is the cabinet optimized for Thiele/Small volume to extend bass for the 5N412-DBs as much as possible? And if not, what should the volume be? I could build the cabinets deeper to increase volume and then use filler when upgraded.

Richard Bechler
Long Beach, CA 90813

Contributing Editor D'Appolito replies:

The Swan IV satellite is optimized for use with the Swan IV bass module. Its low-end response is deliberately limited to extend dynamic range. The satellites are down 3dB around 180Hz and fall off at 6dB/octave below that point, down to about 60Hz, where they transition to a 12dB/octave rolloff.

To extend the bass response of the Focal mid-bass drivers, satellite box volume would have to be at least doubled. Because of their very low Q_{ts}, a bass-reflex design would also be required. I'm certain a full-range system using the Swan IV satellite drivers alone could be made to work well, but it would not be compatible with the Pedal Coupler as presently designed. Any enclosure associated with such a design would have to be discarded when going to the full Swan IV system.

A STICKY DEBATE

In the Swan IV article, the authors argue that regular wood glues do not lend themselves to use with particle board, which I agree with. They further state that epoxy adhesives are therefore "virtually required." I strongly disagree with this contention.

Considering the possibility of "permanent dermatitis," the relative expense and the extreme care required, I would suggest an alternative, Clear Siliconized Acrylic Caulk (CSAC). I have used this adhesive for five years with superlative results. Swift assembly is not required since CSAC remains workable for a half-hour or so. The caulk has good adhesive properties, even when wet, and will hold a small cabinet (1 ft.^{3}) together with no clamps.

When I first used CSAC for assembling loudspeaker cabinets, my friends expressed doubts. They suggested using pure silicone adhesive at about $8 a tube. I reasoned that since Elmer's glue is pure acrylic, then the addition of some silicone should provide better performance. To test my theory, I assembled with CSAC, a 2.7 ft.^{3} rectangular cabinet, of ¾-inch particle board, and dropped it approximately six feet onto a concrete floor. It bounced.

I use mitered joints and when clamping, let the caulk squeeze out to ensure that all gaps are filled. Since I know my box will be air-tight, I fasten my crossover and all batting material to the interior sides of the box before caulking,. That way, I never have to work through the speaker holes.

When the caulk is firm (24 hours) I scrape away the excess with a knife, corner round the edges with a router, and paint. If you prefer veneer, just clamp a little tighter, shave off the excess and proceed as usual.

Dan Greene
Smithtown, NY 11787

James Bock replies:

Mr. Greene's idea of using Clear Siliconized Acrylic Caulk as an adhesive may be a good one. I have not tried it but I will. I am inclined to reason that it would not likely be the best form of adhesion for the Swan IV enclosures.

I learned to appreciate the extraordinary values of epoxy adhesives when I was an engineer in a plastics lab, where I evaluated adhesives and coatings. Most woodworking adhesives require a thin glue line for strength because the hardened glue itself is not strong, only its ability to adhere to the surfaces to be bonded. Epoxies themselves are strong (high tensile and shear strengths) and have remarkable adhesion to a wide variety of materials. The benefit is they do not require perfect fit and high clamping pressures, which is critical for a thin glue line. One source for accurate technical information on epoxy properties is Gougeon Brothers (see sources in the Swan IV article, Part I).

My experiences with acrylic and silicone *caulks*, not adhesives, is that they are elastic and not particularly adherent. Elasticity in a joint caulk is desirable for sealing and for that purpose extraordinary adhesion is not required. Elasticity does not appear to be desirable for structural joints, since usually tensile and shear strength is wanted. These qualities are not consonant with elastic yielding.

SATELLITE SMORGASBORD

I have built D'Appolito's original (*SB* 4/84) speakers with Dalesfords and more recently with Focal 5N401s. I used the specified tweeter (Dynaudio D-28) and passive crossover in conjunction with an active (Marchand) 24dB/octave division at 115Hz to a 12-inch Dynaudio 30W54 in a modified Sanders transmission line. The results have been most impressive.

Lately, I have added another active crossover at 2kHz and have put aside the passive crossover except for a paralleled 5µF polypropylene and 5Ω resistor in series with the tweeter. I have yet to equalize the D-28 with active circuitry. I am driving the tweeters with Borbely 60W MOSFET amplifiers (*TAA* 2/82) which are biased at 450mA—essentially Class A output. The Focal mid-bass units are driven by 20W Class A Lang units (*TAA* 2/86), which do not seem overly stressed (I rarely exceed 90dB, C-weighted, at three meters on peaks). The bass transmission lines are driven by an ADCOM 555.

I gather from the *Mailbox* section and from D'Appolito's articles that the D-28s' selection as tweeters remains an active topic. I hope you can bear another go around. Have you auditioned the Siare TWZV (not to be confused with the similar TWYV)? Its specifications are unusual—resonance in the mid 500s, high power, flat response from 600Hz to 20kHz—on paper it looks like a winner. It costs, however, approximately $65. I'll probably need a phase-adjustable crossover to fully integrate it with the Focals. Any comment?

I'd also like to comment on Mr. D'Appolito's correspondence with Judson Barber (*SB* 4/88, pp. 65-66) regarding the passive versus active crossover approaches. I've tried both and am convinced to a high degree that active crossovers are superior, albeit at higher cost and construction complexity.

This is not to say that his system, as published, is somehow defective. But in my New York City apartment living room, 12 by 30′ with two connected 10 by 10′ spaces and 8-foot ceiling, the active approach has yielded cleaner transients, better depth imaging and a smoother overall response to *my* ears.

I would recommend first building the published system to discover the excellent "stock" performance, before going off the deep end with extra amplifiers and electronics. I've lost count of the hundreds of hours I've spent trying to make this modification work. In my case, it has been worth it. Nonetheless, I intend to build the Swan IV and will report back after its completion. It looks superb.

Les Winter
New York, NY 10003

Contributing Editor D'Appolito responds:

I agree completely with your advice to *SB* readers that they proceed first with the stock version of the Swan IV before going on to triamping. I have heard from many *SB* readers who have tried triamping with earlier versions of my satellite design. They all indicate that the active crossovers produce cleaner transients, better depth and imaging, less glare, grundge, tizz, sizzle, and so on. I believe their results, but I question the oft-stated source of the improvement.

As speaker builders, we are perhaps too preoccupied with the purported benefits of direct amplifier connection to the individual drivers and ignore what I believe to be the more important benefit of reduced intermodulation distortion from amplifiers which have to produce only part of the spectrum. The cited sound characteristics of the passive crossovers relative to the active ones are more representative of amplifier IM and TIM than of driver/passive crossover interactions.

Many power amplifiers within our budget simply do not handle upper midrange and high frequencies very well. One may be very happy with amps in this price range, but they often produce a steeliness, grain and lack of "air" which becomes apparent when tested against the likes of Krells, Spectrals and Cochran Delta Modes, to name a few. At Swan's Speaker Systems we have had the opportunity to hear our satellites connected to these amps and to compare them directly with representative medium-priced amps, such as Haflers and ADCOMs. Based on these listening tests, I have come to believe that a system using the latter amps would benefit from triamping, while the former would show very little, if any, improvement.

Regarding your question on the Siare tweeters, I have no direct experience with them. I considered them for use a few years back, but dropped the idea after examining the manufacturer's data sheet. It showed them to be too directional at high frequencies.

PARAMETER PUZZLE

I gave the formula from my article, "A Primer on Driver Designs" (*SB* 2/88), as it would be entered into a calculator. The correct algebraic form is:

$$R = 20 \log_{10} Q_t$$

where R is the output of the speaker at resonance in decibels, relative to the midband response.

I wish to point out that the Q of a speaker is not as Mr. Honeycutt defined it in his response to my article (*SB* 4/88, p. 53); it is the ratio of output at resonance to the output in the pass band of the speaker (or filter). In *SB* 3/80, Linkwitz gives a formula (after Small) which gives the response of a driver whose cone behaves as a rigid piston and is mounted in a closed box, as follows:

$$F_W = \frac{(f/f_0)^2}{\sqrt{[(f/f_0)^2 - 1]^2 + (1/Q_0 \times f/f_0)^2}}$$

Converting to decibels, $R = 20 \log_{10} F_w$, and at resonance, $f = f_0$, so $f/f_0 = 1$. Thus:

$$R = 20 \log_{10} \times 1/\sqrt{(1/Q_0)^2}$$

which is equivalent to the expression $R = 20 \log_{10} Q_0$.

Although this formula is given for a closed box, it applies to a well-built woofer in free air, which also behaves as a second-order high-pass filter. Therefore, Mr. Honeycutt is right: what I'm doing *is* very much like trying to convert the shape of a mountain top into the altitude partway up the slope. It works, too, because the mountain in question follows the curve of a second-order filter.

Finally, my syntax was incorrect when I referred to a driver as having a Q at a particular frequency. However, the behavior of a driver is more easily predicted at resonance than at any other frequency, and calculations involving Q are most useful at frequencies near resonance. This is, of course, because a speaker's Q is measured by examining the impedance curve in a band of frequencies centered at resonance.

Thank you for the opportunity to clarify these points.

Perry Sink
Lincoln, NE 68505

Richard A. Honeycutt replies:

As so often happens when we humans try to communicate, we seem to have a breakdown here. Mr. Sink's original article contained the following statement relative to Audax's test methods:

"Woofers function as if mounted in an infinitely large enclosure, and the response curves correspond closely to the Thiele/Small parameters. For example, if a driver has a Q_t of 0.31 at 25Hz, it will be 10.2dB down at that frequency, because 0.31 log × 20 is −10.2."

Nowhere is it mentioned that 25Hz is the resonant frequency of the hypothetical speaker. Thus I interpreted the passage to imply that Q could be used to determine the response at any arbitrary frequency. This is clearly false. As it turns out from my recent telephone call to Mr. Sink, this is also not what he meant. Apparently, he intended to say that if a speaker has a Q of 0.31 *and a resonant frequency of 25Hz*, then the acoustical output at 25Hz will be 0.31 times the midband output. Of course, $20 \log_{10} (0.31) = -10.2$dB is correct.

In regard to the nature of Q itself, it is most fundamentally defined as 2-3 times the ratio of the energy stored to the energy lost, in any period. It was first applied to band-pass filters, for which it does indeed represent the ratio of fr to bandwidth. More recently, the concept of Q has been applied by extension to low-pass and high-pass filters. As Mr. Sink correctly states, when so used, Q does not equal fr/BW, although it does approximate this ratio at high values of Q.

I hope this series of interactions will help to clear up what is sometimes a point of confusion concerning speaker parameters.

THE FALCON BOX

I'm a Brazilian engineer and like to build my own amplifiers and loudspeakers. I read "Building the Falcon Monitor" (*SB* 4/81), but in Brazil it is impossible to purchase high quality units. However, I'm taking a trip to Europe in the near future and I hope to purchase some new drivers.

What I have in mind is an enclosure with an 85-liter internal volume akin to the AR-93, but with the bass/mid crossover around 140Hz (first ripple boundary of Allison's published curves in *Audio*). I would feed the two 10-inch bass units from a separate power amp, possibly using motional feedback, −3dB at 30Hz. Now, I'm looking for units to cover the band from 140Hz on, and the mid enclosure does not need to be big, to avoid losing enclosure volume. My first obvious choice was the LS3/5A class of small boxes, but these are far too insensitive for CD and to my fairly large room (180m³). The problem is that I will not be able to listen to anything I purchase before getting back home, and that's why I decided to ask for some advice.

I've considered, in small boxes:

- Falcon Tempo, Focal 5N402-DB, plus Scanspeak's 2008 (or 2010) tweeter;
- Falcon DB-13, Focal 5N402-DB, plus Audax' 9-8 D25BAHR tweeter (these are 4dB more efficient, however Focal is completely unknown to me).

In not so small boxes, but amenable to "squeezing" to 10-15-liter volume, 89-90dB:

- Falcon 2800B, Focal 7N401-DBE, plus T-120 tweeter (this could be changed);
- Wilmslow/Peerless KP 65W FX-8P, plus Audax titanium tweeter;
- Wilmslow/Scanspeak 18WB (or 18WPP), plus 2010 tweeter;
- KEF CS-3 in a 15-liter enclosure (I think the B-200G is only a compromise for the mids).

I have some prejudice against paper cones. That's why I like Scanspeak 18WPP in place of the 18WB Wilmslow. Am I right? Actual speakers: KEF B200-A, plus Audax 9-12D25W with KEF DN-13 crossovers.

Jorge O.F. Oliveira
13083 Campinas Sp.
Brazil

Ralph Gonzalez replies:

First let me suggest some additional sources. *Speaker Builder* magazine sells back issues containing loads of information on drivers, crossovers, enclosures, and so on, and Vance Dickason's *Loudspeaker Design Cookbook* should be very helpful, even to an engineer. You may be able to mail-order drivers from US distributors such as A & S Speakers and Madisound Speaker Components. These distributors can also supply specifications and graphs on many of their speakers, and may give design suggestions.

The drivers you suggest are excellent choices. I have heard very good things about the Scanspeak ¾-inch tweeters. I have had good luck using the less expensive (Scandinavian) SEAS H225 ¾-inch tweeter. MB Electronics makes a ¾-inch titanium dome tweeter (MBH19E), but I suspect the Scanspeak ones can handle more power. Morel makes excellent high-power 1-inch tweeters: MDT28, MDT30 and (high efficiency) MDT33. I have also heard good things about the Focal tweeters, but have never tried them.

As for midranges, I too have been looking for a unit with a flat response from around 140Hz to around 6kHz. Your "prejudice" against paper cones is probably a good thing, since they may act strangely in Brazil's humid climate.

I would also suggest you consider using *two* midrange drivers per speaker, to increase power handling and sensitivity. In *SB* 4/85, Joe D'Appolito discusses using the high-quality Focal 5N402-DB and 5N401 drivers, arranged with the tweeter between the two bass-midrange units. (That is, a 5N401 or 402 at the top of the baffle, a tweeter beneath it, and another 401 or 402 beneath the tweeter.) This is said to improve vertical lobing behavior (dispersion). The tweeter he uses is a Dynaudio D-28 which is similar to the Morel MDT28, but has higher midrange sensitivity due to horn loading.

The driver I have experience with is the SEAS 11FGX 4½-inch bass-midrange. This unit has a coated paper cone which should be resistant to humidity. It has an unusually wide and smooth response, from 55Hz to 7kHz, a very large magnet, and good power handling and distortion. I have seen it used in some very nice, very expensive commercial speakers. You could use two wired in parallel, in a 2.5 liter subenclosure (both drivers in *one* 2.5

liter box) for a natural second-order cutoff at 140Hz (that is, no electrical high-pass filter). This would give you a sensitivity around 92 or 94dB/W/M (2.834V), depending on the width of the main baffle and the accompanying "diffraction loss."

Other drivers to consider are the Scanspeak 13M (8640) and the expensive Eton 4-200/25. Peerless makes good, inexpensive drivers as well. I would avoid drivers larger than 5 inches, since these will have worse high-frequency dispersion and extension, and will require large enclosures. Good luck and please let me know how it turns out.

IMPEDANCE SCALING

I've read Robert Bullock's two-part article about crossover networks (*SB* 1, 2/85) and I took advantage of some information such as the impedance equalizer circuit and the important difference between a two-way and a three-way crossover.

I wonder whether I could impedance-scale the impedance equalizer network so that I could end up with some practical values for L and C. What possible effects do you think would result from doing so? Also, when designing a crossover can you calculate the source impedance as being the ratio of the load resistance to the damping factor of the amplifier to be used? For example: damping factor = 60 and $R_L = 7\Omega$; $R_S = 7/60$. Is it safer to do so or just assume that the source impedance is 8Ω or 0Ω?

Jude Duverger
Brooklyn, NY 11212

Contributing Editor Bullock replies:

The impedance equalizer cannot be scaled to obtain more practical element values because it would no longer equalize the loudspeaker to the same resistance, thus defeating its purpose.

Your procedure for calculating source resistance is correct.

ARE ZOBELS PASSE?

In reaction to Paul Graham's letter in *SB* 2/87 (Bravo Ballard, p. 58), I cannot help wondering whether Zobels/passive impedance equalization still have a place, even in active systems.

I am a staunch supporter of active crossovers; I believe they make a significant improvement in all areas.

However, I still wonder about the load that the individual driver presents to its dedicated amplifier; wouldn't an impedance matching Zobel make the load appear even more resistive and thus give the amp an easier time? I imagine this would contribute to a more "transparent, accurate" system.

Angel Rivera
Brooklyn, NY 11204

Paul Graham replies:

It seems to me that if a single amplifier has sufficient current delivery capability for a single speaker load, at any impedance presented by the load throughout its usable range, then I do not need to alter the impedance. Clearly, a load with considerable capacitance may be more discriminating about amplifier performance relative to subjective sound quality, and perhaps offering the amp a more resistive load might improve things. However, dynamic drivers with large capacitive elements are generally high frequency types such as leafs and ribbons, and these have exceptionally flat impedance curves and in any event, place small power demands on a suitably sized amp, owing to their frequency range of operation; thus avoiding the problem of amplifier stress in the first instance.

Please understand this is my opinion, based on certain, mostly theoretical, information. We don't often find published reference to experience with demonic loads except in the case of electrostatic speakers, which is an entirely different realm. I would say, as a rule, that circumstances of individual amplifier and driver selection, and noticeable improvement by use of a Zobel or other device to increase the resistivity of the load, would be rare enough to qualify as an exception.

Apparently, you are multiamping at present. Have you built resistivity networks and tried them? Do they improve sonic performance? My bet is you could probably concentrate your resources in other areas, but by all means, experiment with this if the idea will not leave you, and let us know what you find.

My idea of a Zobel is that through its intelligent use, it will help flatten the impedance rise of a driver in the region where it crosses over to another driver, whose impedance rise should be similarly flattened. The point of this is to present the amp with a constant impedance through the crossover region, and therefore is valid for correct performance of a passive crossover. For actives, the Zobel is not practical, because the amp and driver are one-on-one, and no amp is being asked to look into a complex impedance. Thus, nothing is accomplished and the presence of "more stuff" can actually detract.

Passive equalization is yet something else, though again, more stuff. Nothing is to be gained from a passive frequency-altering element, or network at power level, that cannot be gained by an electronic device at line drive level. One of the great advantages of multiamping is that you can use low-noise high-slew op amps with precision, close-tolerance components for the crossover. Why defeat this philosophy by hanging large caps and inductors on the amplifier outputs? If you are adverse to graphic equalizers and wish to obtain a specific equalization curve, this could be done by a minimal op amp circuit ahead of the crossover.

Jung's and Lancaster's books (see references from my article, "Fast and Easy Active Filter Calculations," *SB* 4/87, p. 63), should give you a start on how to design active equalization. Note I said *ahead of* the crossover: I try to keep all active devices in front of the component whose outputs are intended to have specific phase relationships, to benefit from this important characteristic of a good crossover. Nothing follows the crossover outputs in my system except the power amp inputs.

My final thought here is that the whole idea of multiamping removes from consideration all of this passive garbage, wherein lies its inherent advantage.

QUAD OCTALINE

The more I looked at John Cockcroft's Octaline design (*SB* 3/87) with my bifocals, the more it grew. My Octalines (*Photo 1*) weigh 40 lbs. each and are 12½ by 25 by 12 inches, with grilles. The quads now speak with authority. The bass is quite impressive, midrange is smooth, and highs, silky; thank you, John Cockroft.

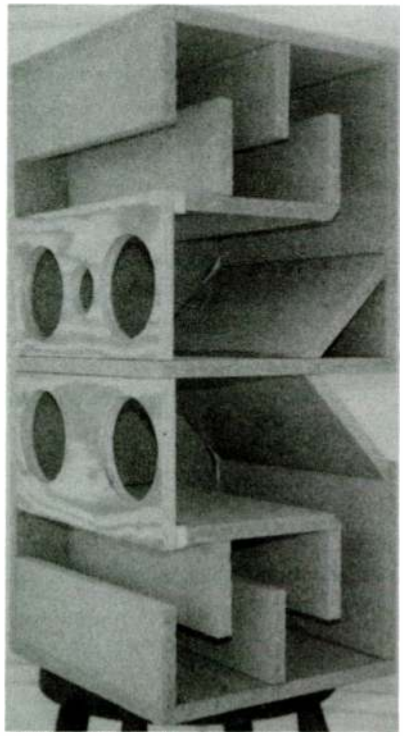


PHOTO 1: Cross section of the Quad Octaline.

For each enclosure, I wired four Radio Shack 40-1022 woofers in series parallel and used Speaker Coat on the cones (a copolymer sold by Landes Audio, Rt. 24, Chester Mall, Chester, NJ 07930).

I chose Audax's 1-inch dome model (HD100D25HR) after listening to several tweeters, including the Audax TW60 titanium dome and the Radio Shack ribbon, 40-1390. These are fine tweeters but seem to work better at higher crossovers, 7kHz and up. My combination seems to blend beautifully, with a seamless quality. I wired a 4μF Mylar capacitor to the tweeter, per Mr. Cockcroft's modification.

All the drivers are mounted on the front panel, which is covered with felt. I omitted a section of the back panel to allow installation of the crossover and tweeter level control. I had to move the tweeter to the top panel—it was too large to fit between the woofers as in my original design.

I took all the measurements from Cockroft's Octaline—stuffing, padding and so on. To make a weight for the woofer cone, wrap lead or solder around a D battery, overlapping both ends and cutting this with a knife. This makes a perfect ring in one piece.

By the way, when I purchased the Radio Shack woofers, two were from Japan, six from Korea. I noticed the Korean models had heavier surrounds. I recently purchased four more, and the surround seemed thinner. They might be getting their specs down.

Manuel Favorito
Albuquerque, NM 87114

John Cockroft replies:

I must say I'm extremely impressed with the product of your imagination. I certainly appreciate your enclosure (no pun intended) of a photo of the cross section of the Octaline Quad. I am quite proud to have been the source of your inspiration. (It should go over well with the bee keeping set; it looks like a honey of a speaker.) Humor aside, it appears a well-thought-out variaton and I would expect the bass to be as you state.

I've used the Audax HD100D25 with the 40-1022s with quite pleasing results. As I recall, the impedance was similar to the tweeter I used in the Octaline, so the 4μF capacitor should do well. My only problem with them occurred when I blew one in a Shortline. In fairness, I believe my amplifier was underpowered and the sound level was a bit high. Clipping distortion was audible prior to the blow out. With more suitable equipment, I have played these tweeters at fairly decent levels.

It would be interesting to roll off the two outer woofers at about 200–300Hz, so the two inner ones could function as the line source, to give better sound shaping in the midrange. You would not need to change the crossover, unless a bit of attenuation is called for. The midrange impedance would change to 4Ω, or 16Ω, depending whether they are used in the series or parallel leg (see *Fig. 1*). Since two woofers will be radiating the whole frequency range, up to about 5700Hz, the phase situation should be excellent.

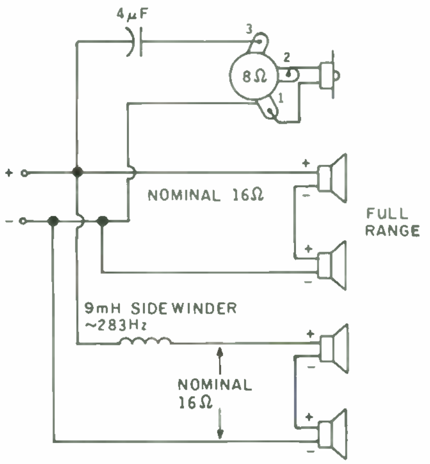


FIGURE 1: Proposed crossover. Placing the coil in the parallel leg (to get 4Ω), both pairs will be rolled off. Try both polarities and use the one that sounds best.

I had made drawings for a quad version of another system, as yet unbuilt (a type of smaller Shortline), configured differently, in a sort of four-barrel shotgun that reduces to two barrels in the midrange.

Incidentally, out of curiosity, I switched from a 4μF Mylar capacitor to a 4μF Solen polypropylene type. I didn't expect a change, but now I think I hear a subtle change for the better (or else I've succumbed to mass hypnotism).

SANDWICH POSSIBILITIES

Since I am the importer of the Focal and Cabasse drivers, I must respond to the Speaker City ad (*SB* 5/88, p. 2) which states, "The true formed shaped cones are a decided improvement over the sandwich types which do not allow for any other cone shape other than a straight side. The possibility of curvilinear cones does not exist with sandwich honeycombed types."

The second sentence is simply not correct. Focal currently has in production and available to its customers a 5¼-inch cone midrange (5K413) and mid-bass (5K013 L) driver, both featuring the Focal K2 diaphragm, a true sandwich woven Kevlar/micro balls of foam-synthetic resin compound/woven Kevlar shaped with exponential profile.

Cabasse has been producing for a few years a variety of drivers featuring true concave dome shaped, honeycomb diaphragms; including flocculated damping/Kevlar/Nomex honeycomb/Kevlar. I admit it is somewhat rewarding to handle such exclusive products which others in the trade think impossible to manufacture.

I believe the first sentence I quoted is misleading. It confuses profile and structure, that is, "true formed shaped cones" and "sandwich types." If the point is that curvilinear cone drivers outperform straight profile cone drivers, I have two observations to make.

Straight cones are understandably more rigid, therefore show harsher break-up modes, and are best suited as bass or bass-mid drivers. Exponential "cones," somewhat less rigid, allow quite smoother top rolloff, lending themselves to mid and mid-bass drivers. Concave domes, interestingly, take advantage of both types. Such different types all have their intrinsic qualities.

If one type was clearly superior in all cases, the whole industry would probably be busy making only that type.

Regarding strength, it seems that sandwich structures offer superior rigidity compared to single layer structures, along with the possibility of better internal damping (within the sandwich).

The industry in general, and *Speaker Builder* and associates in particular, are making a tremendous effort to grow and increase their audience. This goal can only be attained by giving users accurate information, along with true competition among the manufacturers. Therefore, we see any potentially misleading or erroneous statements as counterproductive. Let's all keep up the best work.

Kimon Bellas
President, Focal America, Inc.

DIMINUTIVE WOOFERS

It cannot be coincidental that many highly regarded audiophile speakers use a 4–5-inch woofer/midrange to cover the critical 200–3,000Hz frequencies. Examples are KEF 107, 104 1/2, AR MGS-II, Thiele 3.5 and Snell Type 1/IV. From test reports, these 4–5-inch drivers appear to be equally excellent in distortion and dynamic ranges, compared to more typical and frequently used 6–8-inch midranges, and offer much less coloration than their bigger brothers.

In a review of KEF-107, *Audio* magazine shows tremendous short-term peak SPL capability down to the 200Hz range (handled by the 4-inch midrange). What kind of PP excursion limit must it have not to bottom the voice coil at 120dB peaks?

I am extremely impressed by the quality and lack of coloration of sound from AR 4-inch speakers in a 0.1 ft.^{3} enclosure.

Would *SB* consider testing a bunch of 4-5-inch drivers for a Gary Galo feature article? He seems to prefer dome midranges, but with the exception of Dynaudio D-76, none can operate down to 200Hz. Cone midranges of 4-5 inches made of polypropylene (such as KEF and AR types) appear to be more practical and less expensive solutions for the 200Hz and up applications.

George S. Li
Independence, OH 44131

Contributing Editor Galo replies:

You are correct that many fine loudspeakers do use 4 to 5-inch drivers in the midrange. The 5-inch drivers which are receiving a great deal of praise these days are those from Focal. I do like what dome drivers can do at the upper end of their operating range, but, as I pointed out in my review, most have some limitations at their lower end.

Even the Dynaudio D-76, excellent in this regard, can't be operated as low as 200Hz; I still recommend 400Hz as the low crossover frequency for this driver. If you wish to go as low as 200Hz, you must use a cone driver.

I'm pursuing a change in my already modified TL-10 loudspeaker (*SB* 2/82) which will use the Focal 5-inch drivers crossed over at 200Hz and I hope to report on these drivers.

I cannot comment on the AR driver you mentioned, since I have not heard it. I also do not know the excursion limit of the KEF driver used in the 107. I heard the KEF 107 in two different demonstrations at the Stereophile show last year. I must say that I was not as enthusiastic about this loudspeaker as many others have been. I found the bass to be extremely powerful, and the spatial characteristics were exceptional. However, I found the midrange to be lacking in detail and clarity. Don't accept this as gospel, however; it's simply an opinion based on listening under the circumstances available to me at the time.

VERTICAL vs. HORIZONTAL

In Mr. D'Appolito's article in *SB* 4/84, he states "interdriver spacing is $4\frac{5}{8}$ inches, which corresponds to two-thirds the wavelength at 2kHz." My question is: why did he choose 2/3λ as his interdriver spacing distance?

In Vance Dickason's *Loudspeaker Design Cookbook*, (p. 55), for third-order networks, a null condition can be achieved when "the high-pass driver is forward of the low-pass driver by .25λ of the crossover frequency."

Mr. D'Appolito did not discuss this group delay null condition in his article. Should d2, horizontal driver separation, be calculated as: $0.25 \times 6\frac{25}{32}'' = 1\frac{11}{16}''$ (using λ at 2kHz) *or* $0.25 \times 4\frac{5}{8}'' = 1\frac{5}{32}''$, which is the spacing in the D'Appolito design? This is assuming the drivers are driven in-phase. What would be the best crossover network to use in an active system with the D'Appolito speaker? I'm using Focal 5N411s and the Dynaudio D-28.

Brendan Biever
Encinitas, CA 92024

Contributing Editor D'Appolito replies:

I believe you are confusing vertical and horizontal separation of the drivers. In my 3/2 geometry the $4\frac{5}{8}$-inch figure corresponds to the vertical driver separation. This is the dimension d1 (*Fig. 7.8*, p. 56) of Mr. Dickason's excellent book. In a conventional 18dB/octave crossover with reverse polarity, a null in the on-axis response (along the 0° line of *Fig. 7.8*) is obtained when the acoustic center of the tweeter is forward of the woofer acoustic center by 0.25λ. This is the distance d2 in *Fig. 7.8*. Lambda (λ) is the wavelength at the crossover frequency and is unrelated to the vertical separation of the drivers. Thus the correct figure is that of your first calculation, $1\frac{11}{16}$-inches.

The 18dB/octave crossover used in my satellites is not conventional. The unique frequency response of the D-28 and the phase characteristics of its horn loading, combined with an in-phase connection of the tweeter, are used to produce an "in-phase" 18dB/octave acoustic crossover. A detailed explanation of how this crossover works is contained in my recent Swan IV article (*SB* 4/88).

Driver vertical separation affects the off-axis polar response of my satellites. The vertical spacing of 4⅝-inches was selected to produce off-axis nulls at 35° above and below horizontal. These nulls combine with the natural high-frequency directivity of the drivers to limit polar response to ±15° in the vertical plane (−3dB points). This greatly reduces ceiling and floor reflections. Increasing the vertical driver separation will further narrow the vertical polar response pattern, but may also introduce undesirable off-axis lobes.

PUSH-PULL DESIGN

I saw Mr. Kang's letter in *SB* (3/88) asking for advice concerning push-pull designs. A few years back I built a large (200-liter) enclosure with a push-pull drive ("The Curvilinear Vertical Array," *SB* 2/85) and I would like to share my limited experience.

As far as formulas go, the only change to the T/S system is that for two woofers (whether push-pull or not) in the same box, V_{as} must be doubled. This assumes two identical woofers, which is always desirable.

The two drivers should be mounted as close together as possible; after all, we are creating the equivalent of one driver with a push-pull system. If the enclosure is vented, the vent(s) should be placed an equal distance from both drivers.

Mounting the inward-facing driver satisfactorily has been a perpetual problem for all push-pull enthusiasts. In the case of my CVA, I made a "well" of boards for the inward driver, so its magnet would not stick out from the front panel. As is always the case, compromise is required and the well I made is a cylinder for the first 4.5 inches and then flares out for the remaining 1.5 inches. Avoiding horn-loading or Helmholz resonator effects is important here.

Of course, with a subwoofer, for example below 100Hz, even a well structure, like mine, would be too small to be significant.

Finally, I recommend making an inner screen of grille cloth to cover the back of the inward-facing driver. This protects its motor assembly from dust, which it would not encounter if mounted "normally."

Scott Ellis
Biloxi, MS 39530

UNBAFFLED THINKING

I have heard a large number of subwoofers, but have yet to hear even one system using a single, matrixed box, which is natural sounding on LPs. With CDs, the mastering engineers are no longer using mono sub-50Hz signals, so I would expect that single subwoofers systems would give a poorer showing.

Speaking of subwoofers, I think that possibly the single most exciting topic in audio today is baffle-less subwoofers. Celestion and Enigma are simply turning people's thinking around, although I think both are nowhere near as close to perfection as their prices would suggest. The concept, however, is extremely promising. I have not read enough theory, but offhand it seems that the driver must be near Q = 0.7, with an f_c as low as possible. Looking through driver catalogs, I do not see many stock candidates.

Another relevant area where I am hoping to see some action is polymer concrete enclosures. A number of European and Canadian companies have already realized this, but we seem to be backward here in the US. D'Appolito's satellite design would be a natural to be put up this way. Some company would be required to supply the cabinets, since home construction of a cast polymer concrete box is beyond most amateurs.

My ideal satellite would use D'Appolito's driver layout, but using Focals for both tweeter and mid-bass. The cabinet would be a cylinder of polymer concrete with the front face sawed off, that is, a D-shaped cross section. This would allow easy driver mounting, and would also break up the Bessel function nodes which would otherwise occur inside a perfect circle. For the same resonance reasons, the top of the enclosure would not be horizontal, but would slope backward, at a 30° angle, for instance. If you could combine this with a free-air subwoofer, which should be less expensive to build than a well-damped, braced type, I think you could be coming very close to an ideal loudspeaker.

Vytenis Babrauskas
Bethesda, MD 20814

"HOME-BUILT" SYSTEM UPGRADES

We are building our new home from the ground up. I'm toying with the idea of building a stereo system in the wall of the family room. Is the idea even possible? Could anyone advise me on its feasibility—positive and negative facets of such a project? Perhaps someone has such a system, and could supply construction details.

Anton ElAmma
241 Beacon Dr.
Phoenixville, PA 19460

DUO-CONE DRIVER

I have a pair of loudspeakers that might be of interest to *SB* readers. They are RCA LC1-Cs, built in the mid-fifties, if my information is correct. The speakers I have were used as studio monitors by a small Toronto FM station, now defunct. I purchased them in 1976 and have used them every day since then. Harry Olson describes these speakers in his book, *Acoustical Engineering* (see *Fig. 1*).

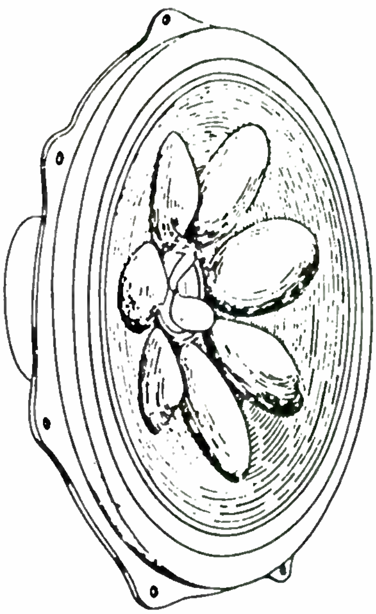


FIGURE 1: A duo-cone loudspeaker with domes attached to the low-frequency cone.

In my opinion, few speakers exist that reproduce sound as well as my LC1-Cs, as long as you listen at "normal" SPLs of 120dB or less. The voice coils in the tweeters disintegrate at inputs of 7W or more.

My speakers use a floor-standing, six-sided enclosure, as specified by Olson and his associates (Olson, Preston and May, *JAES*, Vol. 2, No. 4, 1954, p. 219).

Robert Bennett
Bailieboro, Ontario
Canada K0L 1B0

DELAY DERIVED CROSSOVERS

I share Mr. Rumreich's concern about "the acoustic centers of my drivers" (Mailbox, *SB* 5/88, p. 57), for his "Electronic Time Delay" (*SB* 3/88).

One could talk about piston action, centroids, and so forth, but the dominant specification relevant to his measurements is the inductance of the voice coil. It will introduce a delay of L/R = 100μsec = 1.2 inches. Unfortunately, the L is not constant with frequency, due to eddy current losses, mostly in the center pole.

I've built delay derived crossovers, and they perform well (exceptionally, actually), but they are very difficult for low frequencies. I believe that for better delay derived crossovers we will have to wait for suitable digital crossovers using digital signal processors. I know how to do this, but the effort is considerable.

R.D. Crawford
Los Altos, CA 94022

Mark Rumreich replies:

I appreciate your comments. I suppose it would be possible to compensate for the effects of frequency-dependent driver delay. This improvement would probably be even more subtle than that provided by delay compensation.

I assume the difficulties you refer to with low-frequency delay derived crossovers are due to the long time delays required. You are not alone in your anticipation of DSP in crossovers.

FLAT FREQUENCY RESPONSE

I thank Edwin Gianelli (Mailbox, *SB* 5/88) for referring to himself as a "home hobbyist" as opposed to a home cryogenic engineer, a home mathematics Ph.D., or home computer scientist, toward whom *SB* has become more and more directed. To recap Edwin's dilemma, he has a pair of ADS speakers with what he perceives to be crossover-related sonic flaws. He has no specs on the individual drivers, and wants to determine their sensitivity and frequency response before designing a new filter network.

The answer to Edwin's prayers (and mine) is the 10-band Octave Spectrum Analyzer with onboard pink noise generator and calibrated microphone. Sound expensive? I've been using a BSR SA-3X for five years with great results. It only cost $100, and for another $50 or so, many brands include a stereo 10-band octave equalizer. It won't tell you the absolute sensitivity of each driver, but it will give you the relative sensitivity between your drivers, which is more to the point in crossover design.

A total lack of data on your drivers is not necessary to appreciate an analyzer. As we all know, manufacturers' specs can be pitifully off-target. One maker's sensitivity rating of 91dB can be another's 94dB due to differing measurement methods. Further, the "cookbook" speaker builder, using formulas to determine crossover component values, is virtually assured of a non-flat frequency response in the finished system.

Dynamic drivers, the impedance of which varies with frequency, do not present a purely resistive load to the crossover, an assumption all of the formulas are based on. Zobel (impedance compensation) networks, when calculated mathematically, are virtually certain to be imprecise, a fact that Robert Bullock acknowledges in his article on passive crossover design (*SB* 1/85).

The surest and least expensive method the home hobbyist can use for obtaining flat frequency response (the most fundamental and overriding aspect of good sound) from a loudspeaker project is to build a test box and vary crossover topologies and component values until the not-so-elusive ruler-straight curve develops on your analyzer readout.

Dan Greene
Smithtown, NY 11787

THE WOOD EFFECT

Please welcome a new voice to these pages. A friend who subscribed just this year has handed me his first four issues. I always judge a magazine by its readers, and in the *Mailbox* department I located a widespread intelligence. Thanks also to the editors for not suppressing it.

I also realized that all loudspeaker builders require my recently published book, *The Wood Effect: Unaccounted Contributor to Error and Confusion in Acoustics and Audio.* Although not about speaker building *per se*, it does pertain, most emphatically, to speaker evaluating.

I quote designer Robin Marshall of Epos Acoustics (UK), in a letter to *Hi-Fi News & Record Review*: "The designed-in anti-defect will remain, and the loudspeaker will be flawed." He refers to the wood effect, pitiably unknown to most people, but demonstrable on any decent speaker system. In fact, if you can't hear it, I say, "Blame your speakers!"

My purpose here, besides blatantly obtaining further royalties, is to spread the good news. Nothing I see in *Speaker Builder* leads me to think that the phenomenon is clearly recognized.

The publisher (Modern Audio Association, 23 Stillings Street, Boston, MA 02210) offers a 1988 introductory price of $7, postpaid. Please disregard the self-promotional aspect of this communication. I write because every speaker and listener will benefit.

Clark Johnsen
Boston, MA 02130

KITS • COMPONENTS

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 *Audio Amateur* and *Speaker Builder* are helpful guides. Article reprints are included with the kits. Our aim is to get you started with the basic parts—some of which are often difficult to find—and let you have the satisfaction and pride of finishing your unit in your own way.

PREAMPS

DAVE VORHIS

KP-8: VORHIS LAST PAS MOD. [4,5:82] Precision parts, 1% polystyrene capacitors and metal film resistors, gold-plated phono jacks and B+ regulator board included. Does not include balance, volume control, or 71k capacitors. Each **$210**

FRED GLOECKLER

KP-6: MIKE PREAMP. [3:82] Gloeckler's adaptation of Advent's MPR-1. All parts, input transformer, gain switch and PC board for two channels. Case, connectors and batteries not included. Each **$75**

JOSEPH CURCIO

KS-6: CURCIO VACUUM TUBE PRE-PREAMP. [5:84] Contains board, all parts, jacks, tubes & sockets for amplifier/regulator. With master power supply, including board and transformers. No enclosures supplied. Each **$130**

KT-1: CURCIO TUBE PREAMP. [2:85] Contains board with all parts, bridge, relay & socket, tubes & sockets, heatsinks, and switches. Power supply with transformers included. Each **$365**

KV-3: AUTO MUTE. [1:86] Turn-on delay for power amp and speaker protection. Each **$16**

ERNO BORBELY

KT-2: BORBELY PREAMP. [4:85, 1:86] A highly sophisticated preamp using the best technology and design ideas available. KT-2 includes five sections for each channel of the stereo preamplifier. Each section is built on its own board. In addition to the two boards of the amplifier proper, there is the tape buffer, line amplifier and power supply. The kit comes complete with two function switches and volume control. Included are two toroidal power transformers, gold-plated connectors and all precision passive and active components. An article reprint is included. The kit does not contain hook-up wire or cable, solder, housing or knobs. Each **$625**

KT-2 SECTIONS. (1-5) Available separately. Inquire for pricing.

KT-2/KW-3: With supply upgrade to KW-3. Each **$775**

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

POWER AMPLIFIERS

WILLIAM Z. JOHNSON

KH-6: AUDIO RESEARCH ST-70-C3 MODIFICATION KIT. [4:77] Kit supplies ALL parts: wire, tubes, controls, assembled circuit board and hardware to modify Dynaco's Stereo-70 stereo power amplifier to *Audio Research Corporation* levels of performance. The transformers, power switch and cord, fuse post, chassis and a few of the screws and nuts are the only parts of the Dynaco ST-70 unit used. Fully detailed construction booklet with step-by-step instructions. Each **$400**

KH-6B: AUDIO RESEARCH ST-70-C3. Modification kit construction booklet. Each **$5**

ERNO BORBELY

KP-3A: BORBELY 60W MOSFET AMPLIFIER. [2:82] All parts for one channel (except "L" choke) including board, driver and output heatsinks. Each **$85**

KP-3P: BORBELY 60W MOSFET AMP POWER SUPPLY. [2:82] Two channels. Transformer and all parts except chassis and large filter caps. Each **$75**

KP-3PC: BORBELY 60W POWER SUPPLY. [2:82] Power supply with two 10kuF @ 75V output filter caps. Each **$88**

KS-1: BORBELY SERVO 100 MOSFET POWER AMP. [1:84] One channel includes all parts, board and output heatsinks. Each **$140**

KS-3: BORBELY DC 100 MOSFET POWER AMP. [2:84] One channel includes board, all parts, and output heatsinks. Each **$150**

KS-3PA: SERVO 100, DC 100 POWER SUPPLY. [2:84] Two channels, includes two 10kuF @ 75V caps, one bridge, fuses and one 500VA toroidal transformer. Each **$160**

KS-3PB: SERVO 100 or DC 100 MONO POWER SUPPLY. [2:84] One channel with all parts and 225VA toroidal transformer. Each **$115**

KS-3TA: SERVO 100, DC 100 TOROIDAL TRANSFORMER, 500VA ONLY. Each **$115**

KS-3TB: SERVO 100, DC 100 TOROIDAL TRANSFORMER, 225VA (mono). Each **$72**

KS-1M: COMPLETE BORBELY SERVO 100 MONO AMP with KS-3PB POWER SUPPLY. [1:84] Each **$255** Two or more, Each **$240**

KS-1S: BORBELY SERVO 100 STEREO AMP with KS-3PA POWER SUPPLY. Each **$425**

KS-3M: BORBELY DC 100 MONO AMP with KS-3PB POWER SUPPLY. [2:84] Each **$265** Two or more, Each **$250**

KS-3S: BORBELY DC 100 STEREO AMP with KS-3PA POWER SUPPLY. Each **$445**

REG WILLIAMSON

KK-13CH: WILLIAMSON 40/40 [4:79] Complete power amp, two channel without power supply. Toshiba outputs. Each **$170**

KK-13P WILLIAMSON 40/40 [4:79] Power supply for two channels. No heatsinks or transformer. Each **$33**

KA-2T: AVEL-LINDBERG 40/3018 TRANSFORMER for the 20/20 or 40/40. Toroidal with flying leads, 50V@1.8A. Each **$68**

POWER AMPLIFIERS

JAMES BOAK

KJ-8A: DYNACO MARK III POWER SUPPLY MOD. [1:78] All parts needed to add a solid state regulated power supply to the Dynaco Mark III, including heatsink. The kit also includes all parts necessary to add a balance and adjustable, independent bias control for the Mark III. Each **$36**

KL-1P: BOAK REGULATED POWER AMP POWER SUPPLY. Pass Class A version. [1:80] All parts except large, flat heatsink for either positive or negative half of Pass power supply. Two required for one two-channel Pass amp (4:78). Board included. May be built as either (+) or (−) supply. Requires Pass DC supply components. Each **$82**

KL-1W: BOAK REGULATED POWER AMP POWER SUPPLY. [1:80] Williamson 40/40 or 20/20 amplifiers. All parts except large, flat heatsink. Positive supply. Use Pass power supply (KJ-5-4) for best results and increased power output. Each **$70**

KL-1D: BOAK REGULATOR KIT [2:81] Revised regulator for use with ST-150 MOD KM-8. All parts for regulator including new board. May be built as either positive or negative. One channel only. Includes heatsink. Each **$100** Four for **$320**

NELSON PASS

KJ-5-7: CLASS A 40W AMP A40. [4:78] This bipolar design from Nelson Pass of Threshold Corporation is as rugged as it is clean. Kit contains two boards (3" x 3") and all parts for two channels, including eight heatsinks and all parts for one ± 44V @ 8A (KJ-5-4) stereo power supply.
Each **$440**
West of Rockies, Each **$450**

PASS-CITATION MOD

KM-7: MOSFET CITATION 12 MOSFET MODIFICATION. [2:81] Complete kit, two channel, all parts to modify Harman-Kardon's Citation 12. Each **$170**

BOAK-JUNG-AMER

KM-8: ST-150-BJ-1 DYNA 150 MOD. [2:81] All parts except power supply kit, (see KL-1D), two channels, all resistors and capacitors for the amplifier circuit mods. Each **$90**

K. LANG

KV-2S: LANG CLASS A MOSFET POWER AMP. [2:86] A compact 20W per channel design from Germany which utilizes 4 push/pull power MOSFETs. Especially well-suited for multi-amp systems. Kit includes two circuit boards, all parts, two power supplies (± 24V @ 100mA; ± 15V @ 8A) with toroidal transformers and article reprint. Stereo pair. Each **$255**

KV-2DM: LANG CLASS A MOSFET POWER AMP. [2:86] Includes two amplifier channels and two each of the input and output power supplies for dual-mono operation (each channel independently powered). Each **$340**

FILTERS & SPEAKER SAVER

KF-6: 30Hz RUMBLE FILTER. [4:75] Two channel universal filter card supplied with WJ-3 (F-6) circuit board and all basic parts, 1% metal film resistors and 5% MKM capacitors for operation as an 18dB/octave 30Hz rumble filter. Kit may be adapted as two- or three-way single channel crossover with added capacitors and resistors. Each **$28**

KH-2: SPEAKER SAVER AND OUTPUT FAULT DETECTOR. [3:77] Two-channel kit provides turn-on and off protection. Fast optocoupler circuitry prevents damage from transients. Additional parts provided offer speaker protection in the event of amplifier failure. Kit includes circuit board, all board-mounted components, power supply parts and article reprint by Walt Jung. Each **$62**

CROSSOVERS

For KC-4A choose ***ONE*** *frequency from those listed below. For KC-4B choose* ***TWO*** *frequencies from the list. NO OTHER FREQUENCIES ARE AVAILABLE FOR STOCK KITS.*
60, 120, 240, 480, 960, 1920, 5k or 10kHz.

KC-4A: ELECTRONIC CROSSOVER, KIT A. [2:72] Single channel, *two-way*. All parts and C-4 circuit board. Includes new LF351 ICs. Suitable supplies include KE-5 and KF-3. Each **$12**

KC-4B: ELECTRONIC CROSSOVER, KIT B. [2:72] Single channel, three-way. All parts and C-4 circuit board. Includes new LF351 ICs. Each **$15**

KK-6L: WALDRON TUBE CROSSOVER: Low pass. Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot, level control, National tubes, board and three frequency range determining capacitors. *Specify ONE frequency range per kit please.* (Hz.): 19-210; 43-465; 88-960; 190-2100; 430-4650; 880-9600; 1900-21,000. Single channel, Each **$58**

KK-6H: WALDRON TUBE CROSSOVER: High pass. Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot, level control, National tubes and three frequency determining capacitors. Please specify **one** of the frequencies above. No other can be supplied. Each **$60**

KK-6S SWITCH OPTION. 6-pole, 5-position rotary switch, shorting, for up to five frequency choices per single channel. Each **$9**

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

SBK-A1: LINKWITZ CROSSOVER/FILTER. *Speaker Builder's* [4:80] first kit, including all parts and board for one channel of the three-way crossover/filter/delay. 24dB/octave at 100Hz and 1.5kHz and 12dB/octave below 30Hz, with delayed woofer turn-on. Board is 5½ × 8½". Requires ± 15V supply, not included. Use the Sulzer supply KL-4A with KL-4B or KW-3. Per channel **$72** Two channels **$132**
SB-A1 Board only, Each **$14**

SBK-C1A: JUNG ELECTRONIC CROSSOVER. [SB 3:82] 30Hz filter adapted as a single channel, two way crossover. Can be 6, 12 or 18dB per octave. Includes WJ-3 (F-6) PC board, 4136 IC, quality parts. Choose frequency of 60, 90, 120, 250, 500, 1k, 2k, 5k or 10k. Power supply required. Can use the KL-4A/4B or KW-3. Each **$30**

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

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

SBK-C2: BALLARD ACTIVE CROSSOVER. [SB 3,4:82] Three-way crossover with variable phase correction for precise alignment. Kit includes PC board ($5^3/_8$ x $9^1/_2$"), precision resistors, polystyrene & polypropylene caps, crossover point at 400Hz & 3.5kHz. Requires ± 15V DC power supply—not included. Can use KL-4A/4B or KW-3. Two channel **$145**

AMBIENT SOUND

KH-1: THE WILLIAMSON SUPER QUADPOD. [1:77] Ambience decoder or encoder following the Blumlein system for recording. Parts for one four-channel board. 25V DC supply needed. Each **$38**

SULZER PREAMP POWER

KL-4A: OP AMP PREAMP VOLTAGE REGULATOR. [2:80] Sulzer ± 15V regulator, 30mA capacity. All parts and board. Requires ± 21V filtered DC input. No transformer, filter caps, or rectifier. Each **$40**

KL-4B: SULZER RAW DC SUPPLY. Triad F-91X transformer, diodes and two 5,000μF @ 35V capacitors. Will power two KL-4A supplies. Diagram supplied for construction. Each **$42**

KL-4C: SULZER RAW DC SUPPLY. Same as KL-4B but contains ILP ± 22V toroidal transformer. Each **$55**

KL-4D: ILP ± **22V** toroidal transformer only. Each **$52**

MIXING AND RECORDING

KB-2R: THE 4 + 4 MIXER. [2:71] All parts for eight mikes/inputs plus two extra faders for two added line-only inputs. Rotary fader pots, ICs & sockets and PC board included. Each **$90**

KB-2S: MIXER. Linear slide pot version. As above but includes ten slide pots (2³/₈" travel) with knobs in lieu of rotary pots. Each **$98**

KB-3: 4 + 4 MIXER POWER SUPPLY. + 12V DC, all parts. Each **$16**

KF-2: GATELY EQUALIZER [2:75]. Single channel, three-section equalizer. EG-1 circuit board, all parts including pots and prime LM301s. *Panel not included.* ± 15-18V power supply required.
Single channel kit **$88**
Two kits, as above **$164**

KF-3: GATELY POWER SUPPLY. Regulated ± 18V, 100mA per side. EG-2 circuit board, all parts, includes transformer and heatsink. Each **$48**

SYSTEM ACCESSORIES

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

KG-1: GATELY PEAK OVERLOAD INDICATOR. [2,3:76] Detects signal peaks and warns of overloading levels with LED. Use with or without meter. Firing sensitivity is adjustable down to 170mV. Needs ± 15V from system. Two channels. Each **$16**

KH-8: MORREY SUPER BUFFER. [4:77] All parts & board for two channel output buffer to isolate tape outputs in your preamp from distortion originating in a turned-off tape recorder. Many uses for this versatile matchmaker. Requires power supply. Each **$20**

KH-9: TONEARM MOUNT BOARD. For the Thorens TD-124 turntable *only*. Exact fit, unpainted fine grade hardwood. Three countersunk holes drilled to fit frame. Each **$3.25**

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

KM-10A: SWITCHBOX. [2:81] (6³/₈ × 4 × 2") black with white letters with 5-position DP silver contact, steatite switch, five inputs, and one output with RCA jacks, nickel-plated, and five ground posts. Each **$40**

KM-10B: Same as 10A but with 12 gold-plated jacks. Each **$50**

KP-4: BOAK HEADPHONE AMPLIFIER. [3:82] Two-channel headphone amplifier with two regulated power supplies as in KP-4B. Limited quantities. Each **$195**

KP-4B: BOAK HEADPHONE VOLTAGE REGULATOR. All parts for 12V or 24V regulator for one channel. Does not include transformer, rectifier, or filter capacitors. Each **$25**

KP-5: AUDIO SWEEP MARKER ADDER. [2:82] All parts and circuit board (no power supply) for adding two adjustable markers to a swept audio signal, 20Hz-100kHz. Each **$32**

SBK-D1: NEWCOMB PEAK POWER INDICATOR. [SB 1:83] All parts & board. No power supply required. Each **$7** Two for **$11**

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

AIDS & TEST EQUIPMENT

KE-2: JUNG REGULATED POWER SUPPLY. [4:74] ± 15V @ 1.5A. Lab quality device but excellent for powering system components. Includes board, all board mounted parts plus two LM395K regulators. Transformer and filter caps not included. Each **$45**

KE-5: OLD COLONY POWER SUPPLY. ± 18V @ 55mA balanced, includes all parts, board and transformer. Not regulated. Each **$20**

KH-7: PRECISION 101dB ATTENUATOR. [4:77] All switches, 1% metal film resistors to build Gloeckler's prototype. Chassis, input/output jacks not included. Each **$62**

AIDS & TEST EQUIPMENT

KJ-6: CAPACITOR CHECKER. [4:78] All switches, ICs, resistors, 4½" D'Arsonval meter, transformer and PC board to measure capacitance, leakage and insulation. Each **$90**

KK-3: THE WARBLER OSCILLATOR. [1:79] Switches, ICs, transformer, and PC board for checking room response and speaker performance without anechoic chamber. Each **$65**

KL-3: INVERSE STEREO RIAA NETWORK. [1:80] Two channels, 1% polystyrene capacitors and metal film resistors, gold jacks, cast aluminum box, solder lugs and alternate 600 or 900Ω R_2'/C_2' components. Each **$45**

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

KM-3: CARLSTROM/MULLER SORCERER'S APPRENTICE/PAUL BUNYAN. [2,3:81] Includes six boards and all parts including power supplies. No chassis or knobs. Two article reprints. Each **$280**

KP-2: TWO TONE INTERMODULATION FILTER. [1:82] All parts, circuit boards, 1% resistors included. Each **$26**

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

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

KV-4: HANSEN CIRCUIT-SAFE CHECKER [*CS* 1:86] All parts supplied, except test leads and battery. Each **$20**

PARTS

Tubes

Siemens 6DJ8 . . . . **$ 9.50***
Siemens 12AX7 . . . . **$ 9.50***
Mullard 12AU7 . . . . **$ 7.00***
National 6CA7 matched pair **$21.50***
National 6L6GC matched pair **$18.00***

Integrated Circuits

Signetics NE5534 .... Each **$4.75***
Signetics NE531 (8-pin DIL) .... Each **$4.25***
Signetics NE5535 (8-pin dual FET op amp) . . . . . . Each **$3.70***
Signetics LF351N (8-pin op amp) .... Each **$1.25***
Analog Devices AD-711 (8-pin DIL single FET op amp) . . Each **$2.40***
Analog Devices AD-712 (8-pin DIL dual FET op amp) . . . Each **$3.00***
Motorola MC34081P input op amp . . . . Each **$2.50***

Connectors

SCXT7: ROYCE AUDIO PLUG. RCA type phono plug custom made for Old Colony. Five part construction with excellent strain relief. Heavy 24K gold plate, accepts cable diameter up to 0.23".
Pair **$18.00*** Two or more pair Each **$17.00***

SCXT8: ROYCE AUDIO JACK. Counterpart to SCXT7. Mounts from front of panel (up to 3/16" thick, 1/16" if with insulators) in 5/16" hole. Nylon insulators are included. Pair **$16.00***
Two or more pair Each **$15.00***

PHONO JACK A. Mounts in 3/8" hole from rear of panel (up to 13/64" thick). External hex nut ensures tight installation. Gold-plated hardware included. Pair **$6.00***

NYLON INSULATING WASHERS. One flat/one shoulder, 10 pairs per set.
1/8" size—Fits Phono Jack A **$1.50***
1/4" size—Suitable for 1/4" phono jacks **$1.50***

SB7550B: PHONO PLUG. Gold-plated, fully shielded. Features spring strain relief. Accepts cable diameter up to .24" (such as Neglex 2534).
Pair **$6.50***

SCBPG: BRASS GOLD-PLATED BINDING POSTS. Red and black. 30A, 1000V AC, five-way. Pair **$6.50***

SCBNG: GOLD-PLATED BANANA PLUGS. Stackable, beryllium copper type. Leads held by internal set-screw. Red and black.
Pair **$6.50***

SCSLG: GOLD-PLATED SPADE LUGS. For 1/4" post, accepts 10-12 gauge wire. Solder or crimp. Pair **$1.50***

PARTS

Connectors

INDIUM PLATED SCREWS. 10/32 x ½ "Indium over chrome over brass. Indium provides superior electrical power contacts on large electrolytic terminals (POOGE-2, 4:81). Four **$4.75***

& Cable

518: APATURE SPEAKER CABLE. This heavy 12-gauge oxygen-free copper, linear crystal cable has an ultra flexible clear jacket. Terminate with SCBNG or SCSLG. Twin Lead, per foot **$1.50***

2534: NEGLEX AUDIO CABLE. Low capacitance, high performance interconnect made with OFHC wire by Mogami. Copolymer insulated with spiral shield. Available in blue, black or yellow (specify with length). Per foot **$1.25***

2477: NEGLEX SPEAKER CABLE. Low impedance, high definition cable made with Mogami OFHC wire and copolymer insulation. Per foot **$2.50***

TK2477: TERMINATION KIT. For 2477 cable, includes four gold-plated spade lugs and insulating sleeve. Per pair **$2.00***

2515: NEGLEX HOOK-UP WIRE. Oxygen-free copper, super flexible, 18-gauge with cross-link polyethylene insulation. 25 foot spool in red or black. Specify. Each **$10.00***

Potentiometers

Alps offers the finest in precision rotary controls. All mounting hardware is included.

VR-50K: Stereo volume control, no detents. Two sections matched within ± 2dB. Size: body 25H x 25.3W x 27D (pins project 5mm above height measurement). Shaft length 13mm beyond mounting bushing. **$27**

VR-100KB: Stereo volume control, 41 detents, log taper, tracking error less than 2dB. Dimensions as above. **$27**

VR-100KJ: Stereo volume control, log taper, 31 detents. Taps at 14k for a "loudness" control. Heavy cast aluminum 34 x 33.5 x 40mm case. **$20**

VR-100KD: Balance control, single center detent. **$27**

Et cetera

MSLD-32: OLD COLONY SOLDER. Manufactured by Alpha Metals, our product features a highly activated organic flux (2.2%) with a water-soluble resin derivative of rosin which allows post soldering residues to be removed with a water wash. An article reprint, *Soldering: The Basics* by Marc Colen is included. 63/37 Tin/Lead, .032" diameter. 30 foot spool is 2 oz. Each **$7.50***

Et cetera

KM-12: RESISTOR SAMPLE DESIGN PACK. Metal film resistors, ¼ W, ½ W, ± 1% tolerance. A total of 119 resistors in 31 of the most used values. **A $25 value.** Only **$15.00***

KM-11: RESISTOR SENIOR DESIGN PACK. Metal Film Resistors, ¼ W, ½ W, ± 1% tolerance. 108 values (total E24 series) distributed according to Old Colony's customer preferences over 18 months: 31 packs of 3 each; 28 packs of 5 each; 33 packs of 10 each; 12 packs of 15 each and 3 packs of 20 each. A total of 803 resistors. Regular value $165. Each **$125.00** *

KM-6: CRAMOLIN CONTACT CONDITIONER. Kit consists of one two-dram vial of Red for old contacts, one two-dram vial of Blue for new contacts, lint-free applicators and full instructions for use. This military grade contact cleaning compound dissolves and removes oxides and their effects on all non-soldered contacts in audio systems from cartridges to speaker terminals. Each **$20.00***

KM-9: WILLIAMSON RECORD CARE KIT. Deep cleaning and static removal treatments. (2,4:81) Contains ½ ounce of de-static liquid and 4 ounces of powder for making up coating cleaner. User needs isopropyl alcohol and distilled water as well as glycerin and a surfactant to make up the needed treatments. Full instructions included with each kit. De-static treatment for 1,000 disks, deep cleaner for 80 disks. We regret that due to postal regulations, this cannot be shipped outside of the U.S. Each **$10.00***

KM-9R: CLEANER REFILL. 4 ounces, Each **$6.95***

KM-9S: DE-STATIC LIQUID. refills ½ ounce. Each **$6.95***

KW-1: MAGNAVOX CD PLAYER MODIFICATION. Improves frequency response. Includes two Signetics NE5535, two Panasonic HF series 330μF capacitors and four 3.92k, 1% metal film resistors. Each **$12**

KW-2: MODIFICATION. As above, but with two AD-712 op amps in addition to the NE5535s. Each **$16.00**

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

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

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ORDERING INFORMATION: Prices, except as noted, include shipping via UPS, which requires a street address. If you cannot receive UPS delivery, please include an extra $2 for insured service via Parcel Post. We cannot accept responsibility for safety or delivery of uninsured Parcel Post shipments. For charge card orders under $10 please add $2 service charge. *UPS Next Day and 2ND Day Air available to some areas.* PRICES SUBJECT TO CHANGE WITHOUT NOTICE.

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Julian Hirsch—Stereo Review, Sept. '88

Cambridge SoundWorks has created Ensemble,™ a speaker system that can provide the sound once reserved for the best speakers under laboratory conditions. It virtually disappears in your room. And because we market it directly, Ensemble costs far less than previous all-out designs.

Henry Kloss, creator of the dominant speaker models of the '50s (Acoustic Research), '60s (KLH), and '70s (Advent), brings you Ensemble, a genuinely new kind of speaker system for the '90s, available factory direct from Cambridge SoundWorks.

The best sound comes in four small packages.

Ensemble consists of four speaker units. Two compact low-frequency speakers reproduce the deep bass, while two small satellite units reproduce the rest of the music, making it possible to reproduce just the right amount of energy in each part of the musical range without turning your listening room into a stereo showroom.

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No matter how well a speaker performs, at home the listening room takes over. If you put a conventional speaker where the room can help the low bass, it may hinder the upper ranges, or vice-versa.

What Henry Kloss tells his friends:

Every time I came out with a new speaker at AR, KLH, or Advent, my friends would ask me, "Henry, is it worth the extra money for me to trade up?" And every time I would answer, "No, what you've already got is still good enough."

But today, with the introduction of Ensemble, I tell them, "Perhaps now is the time to give your old speakers to the children."

Ensemble is a Trademark of Cambridge SoundWorks, Inc.

Ensemble, on the other hand, *takes advantage* of your room's acoustics. The ear can't tell where bass comes from, which is why Ensemble's bass units can be tucked out of the way—on the floor, atop bookshelves, or under furniture. The satellites can be hung directly on the wall, or placed on windowsills or shelves. No bulky speakers dominate your living space, yet Ensemble reproduces the deep bass that *no* mini speakers can.

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Unlike satellite systems which use a single large subwoofer, Ensemble features separate compact bass units for each stereo channel. They fit more gracefully into your living environment, and help minimize the effects of the listening room's standing waves.

cabinets ruggedly constructed for proper acoustical performance. We even gold-plate all connectors to prevent corrosion. An even bigger difference is how we sell it...

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Our toll-free number will connect you to a Cambridge SoundWorks audio expert. He or she will answer all your questions, take your order and arrange surface shipment via UPS. Your Cambridge SoundWorks audio expert will continue as your personal contact with us. We think you'll like this new way of doing business.

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Pair of M&K BE-II 12-inch subwoofers with cabinets, $200; one Polk LF14 subwoofer, $100; Dynaco ST-70 totally reworked with too many features to name. Call Jerry McNutt between 6 p.m. and 10 p.m. CST at (205) 821-8426.

B&K model 177 VTVM, immaculate condition with all manuals, asking $150 Canadian dollars; B&K Model E-2000 RF signal generator, very good condition but needs recalibration, asking $150 Canadian dollars. Greg Nawrocki, 21 Indiana St., Kitchener, Ontario, N2H-2A4, Canada, (519) 745-1579.

Madisound 1054, $40/pair; Radio Shack 40-1022 (old style), $20/pair; (1) Dynaudio 17W75EXT, $45; Philips 1065W8, $15/pair; Audax HDB25J2C12, $30/pair; Philips 1265W8, $15/pair; Vifa D75MX-31-08, $30/pair. Alan (302) 764-7479 evenings.

Unused pair Strathearn ribbons with transformers, $200. Call evenings, (616) 241-5324, Dean Price, 1616 Millbrook S.E., Grand Rapids, MI 49508.

DeCoursey 120-R two-way active crossover. 100Hz, rack mount, 18dB/octave, four level controls. Technics SH-8055 12-band graphic equalizer and RP-3800E microphone. Dave Schneider (319) 373-2275.

JBL 2234H 15" woofers, excellent condition, $350/pr; in studio enclosures; $450/pair; JBL 2221 10" midrange, excellent condition, $125/pair; Dyna PAT 4, $75. Call Steve (203) 397-3888 7-10 p.m. EST.

B&K Model 3304 Frequency Response Recorder (BFO Type 1014 and Level Recorder Type 2305), $700 or best offer. Johan Van Leer, 230 Bay St., Santa Monica, CA 90405, (213) 396-3005.

Everest's Critical Listening Course, basic and advanced, cassettes and manuals, $160; Audio Control SA-3050A spectrum analyzer, portable with 6 memories, etc., $750. Roadproof case for SA-3050A, $175. Technics SH-8000 acoustic test set with microphone, decibel meter and ⅓-octave warble tone generator, $160; Custom Audio crossovers, 4kHz, 12dB/octave, 12 for $250. Tom Young (203) 274-2202.

Hafler DH-101, thoroughly modified, Sulzer regulated with top quality parts, superbly clean sound, $175; Ariston RD40AC, excellent sound, excellent condition, $100. Eric Anderson, 308 NW Linden, Ankeny, IA 50021, (515) 964-7710.

WANTED

Old tube hi-fi components including tuners and amps such as Scott Sherwood, REL and speakers and/or enclosures from Electro-Voice. Also, old hi-fi magazines, catalogs and brochures. Jeofrey S. Bean, 9 Eisenhower St., #5, Coventry, RI 02816, (401) 826-3658.

Would like to talk with Bay Area owners of stacked quads. Joe (415) 897-5053.

Burwen 1201A and suggestions for crossover design for MGIIIA. For biamp with transistor low end, tube high end. C.E. DiCini, 1532 Chanslor Ave., #Q, Richmond, CA 94801.

Ionovac, Ionofane wanted. State price and condition. Henry Eng, 1330 Burrard, Suite 718, Vancouver, BC V6Z 2B8.

Power supply, regulated ±15V DC at 30mA and ±5V DC at 120mA. ±12V DC will work also. Please send any information to Mike Gollatz, 1303A Stone Mill Cove, Stone Mtn., GA 30083.

Dyna MK IIIs. Must be complete and in excellent condition. Don May, 129 Glenridge, East Aurora, NY 14052, (716) 652-6107.

Quad 303 power amp(s) in any condition. Call Jerry McNutt between 6 p.m. and 10 p.m. CST at (205) 821-8426.

Pro Plane literature (commercial/industrial). Old Klipsch catalogs, brochures, price lists. 1960's Klipsch K-77 tweeter, 1950's Klipsch University 4401 tweeter, Klipsch crossover/balancing network, air core coils. Western Electric mixer knobs. *Audio Engineering* and *High Fidelity* magazines Volumes 1-5. D.R. Schaller, 6704 Schroeder Rd., Suite 6, Madison, WI 53711.

CLUBS

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

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

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

JOIN AN AUDIO CLUB

Expand your horizons. Improve your system. Learn about the latest equipment and techniques. Share viewpoints and experiences. Don't miss out on the fun and value of belonging to an audio society.

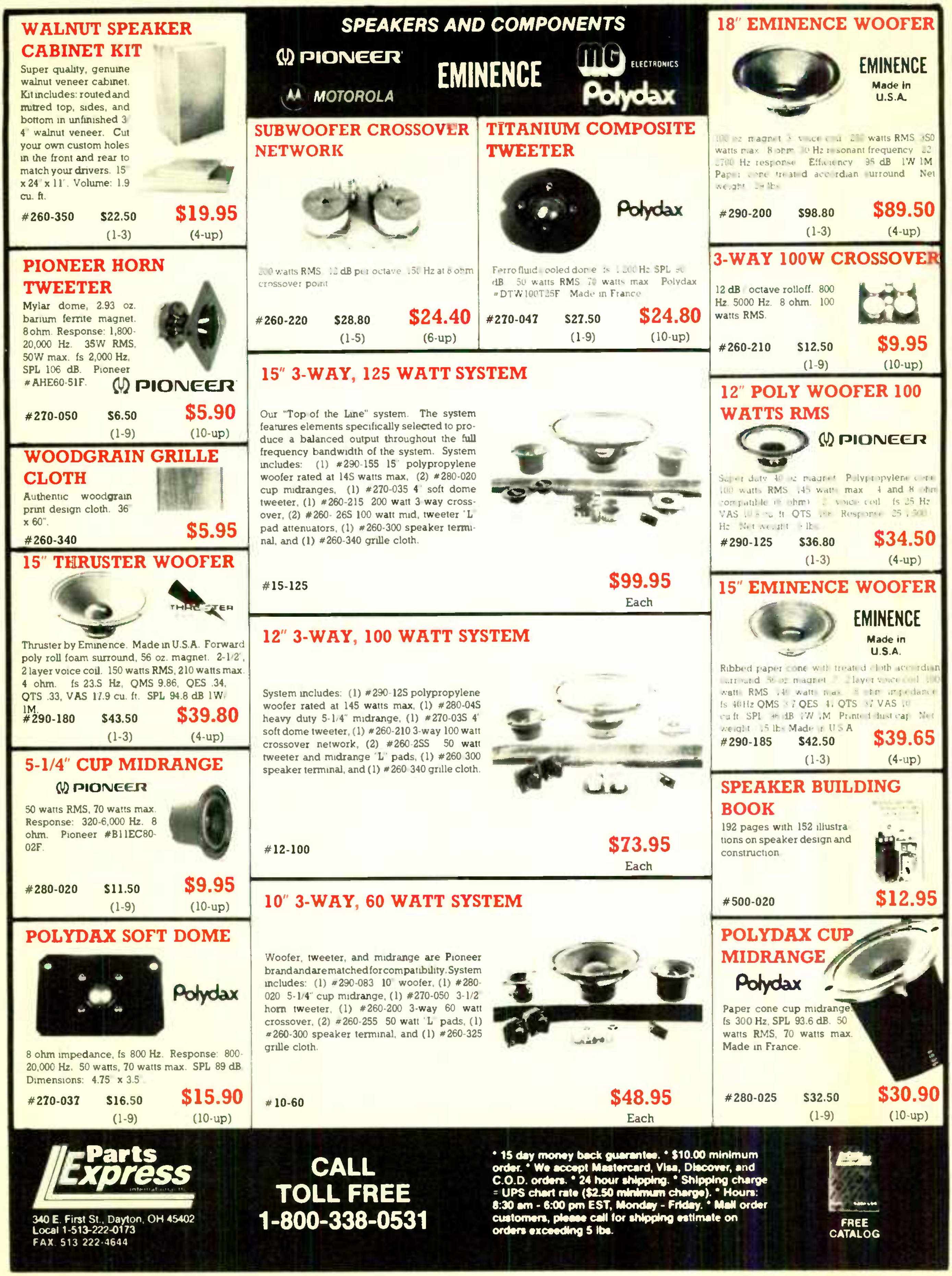
Typical activities include:

- ***Guest Speakers.*** Here's your chance to listen to and meet "superstars" of circuit design, prominent manufacturers, acoustical consultants, and recording engineers.
- ***Tours.*** Get a behind the scenes look at the equipment and talk with the people who operate it at local TV and radio stations, universities, research labs, recording studios, and factories.
- ***Newsletters.*** These publications are often of high technical quality and are full of worthwhile information even if you don't attend many meetings. Ads and reviews help you find the right equipment, the latest records, and the dealers who carry them.
- ***Evaluation and Testing.*** Frequently clubs sponsor clinics so you can bring in your equipment for checkups on test equipment most individuals don't own.
- ***Group Buying.*** This can be an effective way to obtain obscure items from abroad, including audiophile disks.

No club in your area?

Start one—with a **free classified ad** in *Audio Amateur* or *Speaker Builder.*

For more information, see the club listings in the Classified Ads of this issue.



Fast Reply #KC81

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

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

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

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

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

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

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

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

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

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

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

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, AB listening tests, equipment clinics, recording studio visits, annual picnic and audio fun. The club journal is *LC The SMWTMS Network*. Corresponding member's subscription available. Call (313) 477-6502 (days) or write David Carlstrom, SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

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.

Advertising Index

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

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

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

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

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

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

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

THE ATLANTA AUDIO SOCIETY, a club for men and women interested in high quality home audio/video systems and recordings. Monthly meetings are conducted where special programs and guest speakers are often featured. Guests have been: Jack Renner, Richard Vandersteen, William Conrad, William Johnson, Louis Lane, John Cooledge and others. Annual dues are $25 and includes a monthly newsletter. For information call Chuck Bruce, (404) 876-5659 or write Atlanta Audio Society, PO Box 361, Marietta, GA 30061.

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Office use, leftover	2,054	1,801
Return from news agents	0	0
Total	11,269	11,003

I certify that the statements made by me above are correct and complete.

Publication No. 01997920 Edward T. Dell, Jr.

morel

high fidelity range

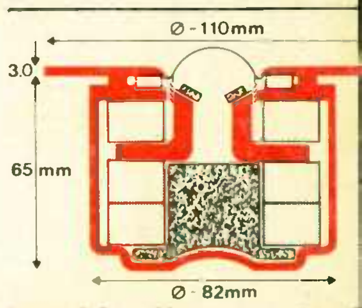
The MDT 33 is an extremely fast Tweeter, using a 28mm (1⅛") diameter voice coil and a chemically treated soft dome, and is ideally suited for two way systems with the possibility of a lower than normal crossover frequency, as well as for three and multiple way systems.

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

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

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

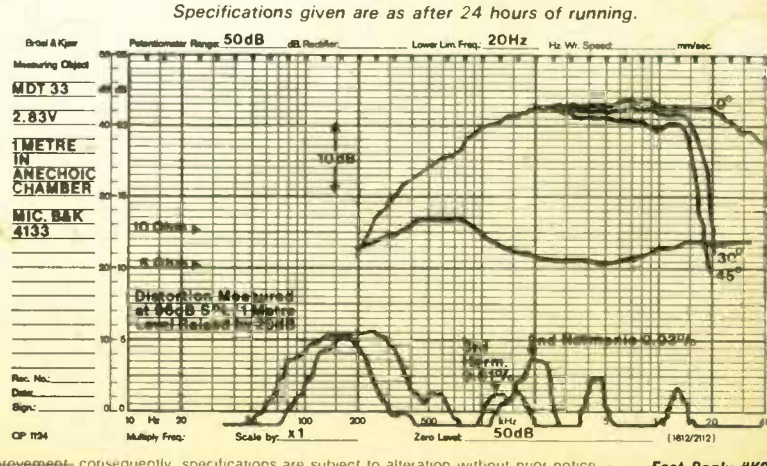
To utilise the dome at the lower than normal crossover frequency available makes it necessary to have a sharp roll off below 1400Hz of minimum 12dB per octave to protect the tweeter from mechanical damage. This makes it ideal for use with active systems.



Specification

Overall Dimensions	Ø - 110mm x 68mm	Vas	0.016
Face Plate Thickness	3mm	Moving Mass including Air Load	0.44 gr
Voice Coil Diameter	28mm (1⅛")	Effective Dome Area	8.5
	Hexatech Aluminium	Dome Material	Treated F
Voice Coil Former	Aluminium	Frequency Response	1400-20000 ± 0.
Number of Layers	2		(1000-40000 –
DC Resistance	5.2 ohms	Resonant Frequency	50
Nominal Impedance	8 ohms	Power Handling Din:	
Voice Coil Inductance @ 1 Khz	0.09mh	X-Over 1400 Hz	1
Air Gap Width	0.75mm	X-Over 5000 Hz	5
Air Gap Height	2.5mm	Transient Power 10ms	150
Voice Coil Height	2.7mm	Sensitivity	92.5dB (1W/
Flux Density	1.95T	Rise Time	1
Force Factor (BXL)	4.76 WB/M	Intermodulation Distortion	
Rmec	2.09ns/m	for 96dB SPL	<0.
Qms	0.66	Harmonic Distortion	
Qes	0.38	for 96dB SPL	<0.
Q/T	0.24	Nett Weight	1.2

Specifications given are as after 24 hours of running.



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