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Speaker Builder

THE LOUDSPEAKER JOURNAL

Ms. Speaker

ESOTAR TWEETER

T330D

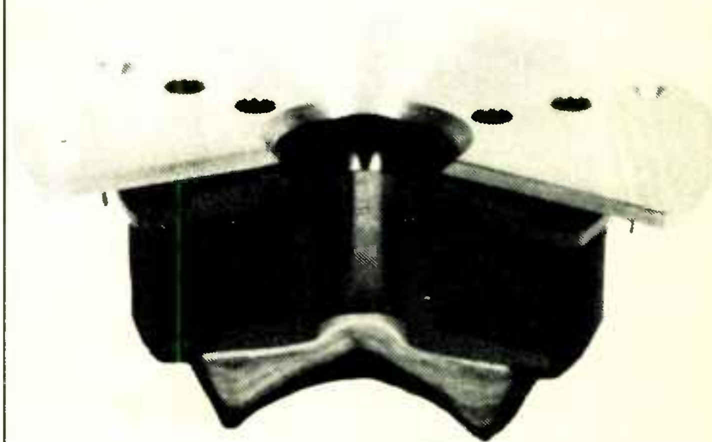
DYNAUDIO OF DENMARK

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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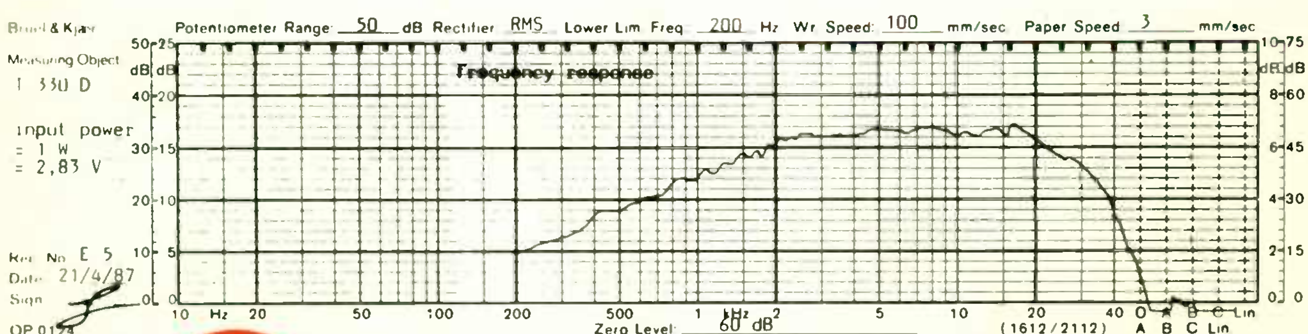

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Fast Reply #ED683

Good News

Roy Allison, President of **ALLISON ACOUSTICS**, has announced the sale of the loudspeaker company to an investor group comprised of former executives from Acoustic Research. Joining Allison, who will remain as President of the company, are Frank Romagnoli, Vice President of Production, Bob Barr, Executive Vice President, and Ron Falkenstein, Vice President of Sales.

The new investor group/management team will combine their sales and marketing expertise with Allison's engineering skills. At Acoustic Research, as engineer of record, Allison developed the original AR 3a and the LST, thus earning his reputation as an engineering innovator.

According to Falkenstein, "We are controlling our distribution to enable key dealers to develop a protected, profitable speaker line. We are encouraging dealer input on product development as well as on marketing strategy. And, we are actively working to develop long-term relationships with our dealers."

Allison Acoustics designs and manufactures loudspeakers aimed toward faithful sound reproduction in a home environment. For further information contact: Bob Barr, Allison Headquarters, 1590 Concord St., Framingham, MA 01701, (508) 788-1500; 1-800-225-4790; Fax: (508) 877-6740.

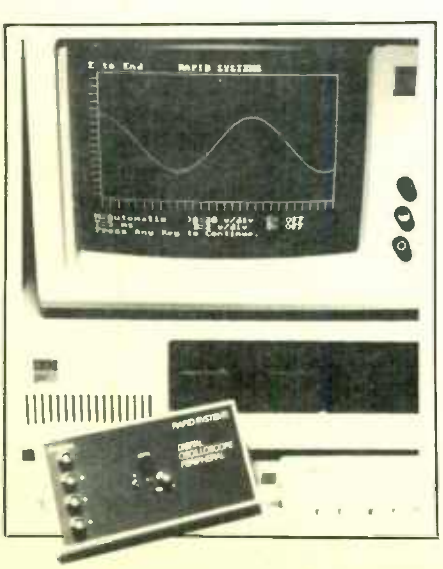
Fast Reply #ED327

LMP (Loudspeaker Modelling Program) has been updated to the *graphics* version LMPG. This version gives superior frequency response plots, compared to the original line-printer-style output. Resolution is 640x200 pixels for graphics-equipped IBM PCs and compatibles (graphics card and graphics monitor are required), and 480x300 pixels on the Apple Macintosh. The graphics version is *not* presently available for other computers.

All LMP disks for the Apple Macintosh and IBM PC sold by Old Colony (CSK-C2 and CSK-C3) now contain the graphics version (IBM PC disks also contain the original version for compatibility with non-graphics systems). However, if you have previously purchased LMP for one of these two computers, you may obtain the upgrade by sending the following to Ralph Gonzalez, PO Box 54, Newark, DE 19711:

1. A disk containing LMP (do not include any personal files);
2. Return postage;
3. $5 handling fee (check or MO);
4. Return envelope no more than 6-inches tall, with a piece of cardboard and/or label: Do Not Bend.

RAPID SYSTEMS has released the R414 Digital Oscilloscope Peripheral for IBM PC/XT/AT and compatible computers. This instrument comes with hardware and software, including source code, to turn your PC into a four-channel data acquisition unit with selectable sample rates from 1kHz to 500kHz. The hardware also allows user selectable gain from 10mV to 320V peak-to-peak, for 8-bit accuracy.



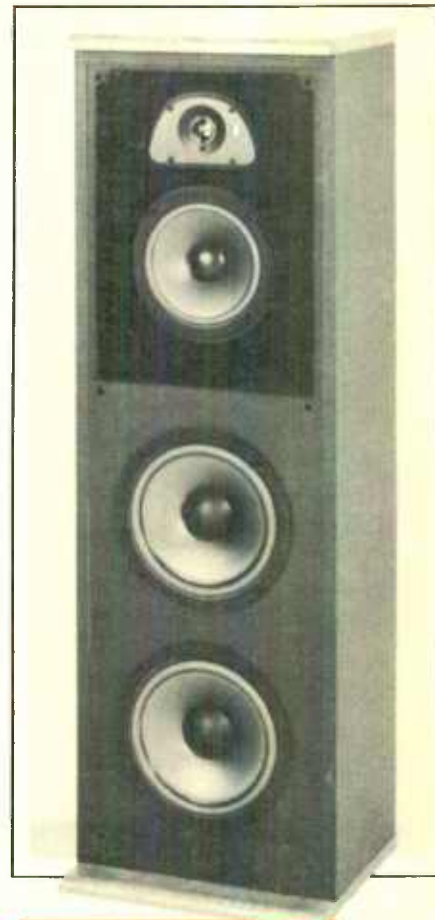
Features include: 8-bit 500kHz A/D converter, 2048 point data memory, external or internal analog triggering, executable routines for waveform capture and display, and sample program listings with a complete user's manual.

Options include: enhanced oscilloscope display software, spectrum analysis software, and digital signal processing hardware.

Rapid Systems also announced their new full line of PC-based FFT spectrum analyzers. Covering a range of one to four channels, 8- and 12-bit accuracy and 10Hz to 10MHz, these spectrum analyzers are applicable to audio, modal analysis, vibration, electronics and biological uses. Prices range from $1,495 to $4,995.

The R414 is in stock now for $295. For more information contact: Rapid Systems Inc., 433 N 34th St., Seattle, WA 98103, (206) 547-8311, TLX 265017 UR.

Fast Reply #ED948



AR has filled the gap in their TSW Series, between the 610 and the 810 floor standing speakers. The 710 is a tall, slim loudspeaker which only requires approximately one square foot of floor space.

The three-way design uses a 6½-inch polypropylene cone midrange, ¾-inch titanium liquid cooled tweeter with AR's Tetra-Helix mounting plate, and twin long-throw polypropylene cone, 8-inch woofers. The woofers are mounted in their own acoustic suspension enclosure.

Also, the new AR TSW 315 loudspeaker is a three-way system using two 6½-inch polypropylene cone woofers working simultaneously in the bass region, with the lower woofer rolling off sooner than the upper woofer/midrange unit. This technique is also used with the TSW 410 speaker.

These bookshelf speakers are 22½ inches high and are designed for optimum listening on 20–24-inch stands. Enclosures for both models are black vinyl veneer with solid hardwood top and bottom (walnut or oak).

Suggested retail price: TSW 710, $650 each; TSW 315, $275 each. Available from Teledyne Acoustic Research, 330 Turnpike St., Canton, MA 02021.

Fast Reply #ED909

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.

EJ JORDAN USA announces that effective immediately, all sales of Jordan ACT loudspeaker drivers in North America will be handled directly by Ted Jordan in England: EJ Jordan Designs UK, The DAK, Manorbier, South Pembrokeshire, SA70 8QR, England; phone (011) 44-834-871-209. Direct sales from England will offer lower driver cost to North American buyers. All warranty and replacement of Jordan ACT drivers will be handled by EJ Jordan Designs in England, effective immediately.

EJ Jordan USA will continue to offer basic technical backup, enclosures and components for Jordan systems. Owners can contact EJ Jordan USA for advice and expediting service work. For information please send SASE to EJ Jordan USA, Inc., 301 N. Harrison St., Building B, Suite 252, Princeton, NJ 08540.

Fast Reply #ED563

GOLD SOUND's 1988-89, 39-page catalog is now available, featuring speaker kits and components. Over 30 kits for home, car and pro are listed. New kits include compound-loaded subwoofers and symmetrical arrayed three-way systems. Frequency response and impedance graphs are included for over 100 speaker components, including diagrams and charts on crossover components.

New electronics products include test equipment and portable ⅓-octave real-time analyzers. New 24dB Linkwitz-Riley electronic crossovers are phase coherent, available for biamping and triamping, and optionally available with bass equalization and subsonic filter for sixth-order alignments; crossover frequency is sweepable from 40Hz–4kHz (model 224) and 40Hz–9kHz (model 424). Level controls are provided for left and right inputs and low and high outputs. All crossover models are rack mount, and have a Plexiglas cover.

Contact: Gold Sound, PO Box 141, Englewood, CO 80151, (303) 761-6483.

Other new items are 6μF and 7μF, 100V Mylar capacitors, $2 each, and JBL Speaker System kits are priced as follows: #11, $1,038; #2005, $1,295; #2018, $1,595; #14, $2,088; and JBL Pro 15, $739.

Fast Reply #ED149

INFINITY SYSTEMS, INC. announces the SM 80, the first bookshelf loudspeaker in their Studio Monitor Series.

A compact design incorporating an eight-inch polypropylene coated woofer, it is enclosed in a tuned bass reflex cabinet with a vinyl finish. A polypropylene foam one-inch Polycell tweeter provides flat response to 27kHz.

The SM 80 will be available at a suggested retail price of $219 each.

Infinity Systems Inc., 9409 Owensmouth Ave., Chatsworth, CA 91311, (818) 709-9400.

Fast Reply #ED255

Fast Reply #ED29

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About This Issue

Reid Woodbury's Ms enclosure, a fresh, and probably unique loudspeaker format leads off our first issue of *Speaker Builder*'s tenth year. If you need both channels from one box—this is the one. **Arthur Brown** has been in and out of audio for a long time. His return and a plunge into speaker tech makes good reading starting on page 16.

David Long has put together yet another simplified Thiele/Small design program on his Commodore 64, a favorite machine whose low price belies its capabilities (p. 22). Contributing Editor **Bruce Edgar** is back this time with all the news you need about those wonderful cordless drill/driver gadgets, as well as the fast fasteners to go with them (p. 25).

Our French connection, **Jean Margerand**, brings us Part 2 of his elegant Delta project based on a format he calls the "Third Dimension," beginning on page 27. And Evan Struhl of Polydax has arranged for a few of the no-longer-manufactured Audax bass drivers to be available to US builders who wish to tackle the Delta (see the end of the article for details). One of our Swedish readers, **Leif Ryden**, shares his article, written for a British journal on how to acoustically tune large halls (p. 36).

Bob White does an excellent, comprehensive review of Scientific Design Software's program for computer-aided design of speaker technology,starting on page 42. The second part of **Peter Muxlow**'s Technology Watch column on subwoofers is found on page 52.

Don't miss **Dick Pierce**'s newest saga of life in hi-fi land on page 70. **Dave Pitt** is our Craftsman this time. See his handiwork on page 40, plus Vintage Designs and more.

Cover photo by Dail White

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THE LOUDSPEAKER JOURNAL

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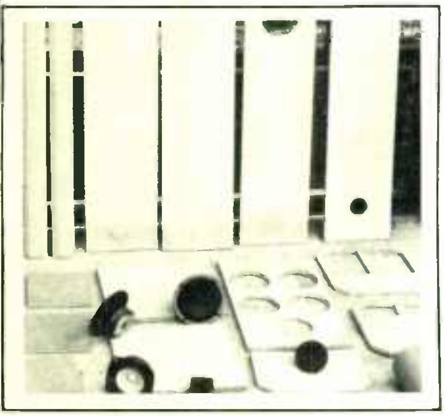

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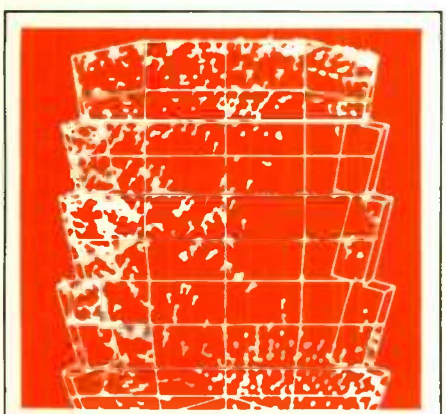

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FEATURES

DEPARTMENTS

"They Were Designed To Play Music–And Make It Sound Like Music. This They Do Very Well, In A Most Unobtrusive Way, At A Bargain Price... It's Hard To Imagine Going Wrong With Ensemble."

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.

Your listening room works *with* Ensemble, not against it.

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.

Not all the differences are as obvious as our *two* subwoofers.

Unlike seemingly similar three-piece systems, Ensemble uses premium quality components for maximum power handling, individual crossovers that allow several wiring options and

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...

The best showroom of all: your living room.

We make it possible to audition Ensemble the *right* way–in your own home. In fact, Ensemble is sold *only* by Cambridge SoundWorks directly from the factory. Listen for hours without a salesman hovering nearby. If after 30 days you're not happy, return Ensemble for a full refund.

At only $499–complete with all hardware and 100' of speaker cable–Ensemble costs hundreds less than it would in a retail store.

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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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Editorial

A Modest Proposal

Every magazine publisher has one continuous built-in, inescapable headache: The United States Postal Service. President Nixon attempted to remedy the most serious structural flaw in the system, the political "pork barrel" component which tempts members of Congress to make rewards out of postal stations and jobs. Inevitably that supposed independence of the Postal Service has been eroded by Congress. Local voter pressure can still affect Postal Service administrative decisions.

Several years ago the government decided the telephone service was, in effect, a monopoly of the American Telephone & Telegraph Co. After an expensive and lengthy court battle, the Justice Department forced a process called "divestiture" on that company with results that are very, very mixed, to say the least. Now the telephone system under the Bell hegemony was one of the best in the world. It was more expensive than it needed to be, but it was the best of its kind. The effects of divestiture are still in a shake-out period but long distance rates are going down, and I suspect, local rates will follow in time.

The period since divestiture by the telephone people is dotted with other events of note having to do with competition in business. The airlines are now de-regulated and fares are certainly lower. The trucking business is de-regulated and the rates there are much lower. De-regulation and divestiture are making many large industries much more competitve than they have been in the past.

If the telephone decision was made in the interest of the consumer, why hasn't the government examined whether the monopoly which the USPS enjoys is beneficial to the consumer?

During that same period we have seen the steady rise of competition in the one segment of the communication business where competition is strictly controlled: While it is illegal to establish a letter carrier business in this land of free enterprise, fortunately it is not illegal to establish a parcel delivery business. The contrast between the efficiency and profitability in the letter delivery business and the parcel delivery business is startling, to put it mildly. The performance of the USPS is improving to some extent, but it is permanently hampered by the inevitable fact that it is not a business but a bureaucracy.

The custodian at our local Post Office retired a couple of years ago. The Postmaster requested a replacement and was sent forms from the area administrative office that had to be filled out to justify hiring a new one. It took him three days to complete the forms, which included measuring every pipe in the building, reporting both the length and diameter of each. After months of delay, a part-time custodian was authorized. Obviously the pipe measurements were important in determining whether this would now be a full- or part-time job.

Bureaucrats are traditionally not interested in profits, efficiency or speed. They are interested in larger budgets and departments and in their next promotion. In business you either make a profit or you find out why you haven't.

Those of us in publishing are keenly aware that we have a problem with postal administration. Decisions about interpreting the three volumes of rules which are supposed to govern mail delivery and fees are made by local Postmasters. Their judgements about the rules, and even their knowledge of them, varies widely. They lose dozens of magazines each issue because of ignorance, carelessness, or both, and we have no recourse. Magazines are not forwarded properly, even *correctly* addressed magazines are destroyed by the Postal Service and marked as undeliverable—often with no recompense of either the postage to deliver them or the cost of the magazines.

And almost invariably, the subscriber considers the publisher responsible and certainly insists that we bear the cost of replacing the magazine. We do so if the Postal Service is at fault because it is cheaper to do this than argue with the bureaucracy. As far as second class mail is concerned, we effectively have no recourse for magazines that are lost, stolen or destroyed in error. Our local postal people do everything in their power to rectify what they can, but it is evident that they fight a losing battle and often there is nothing to be done.

I hope with this overview in mind, you will make an effort to keep your local Postmaster informed if your mail is lost, strays, or is stolen. And I hope that if you have not received your magazine, you will ask first whether you have told us that you moved, and then question your Postmaster as to why you have not received it. He needs your feedback if he is to improve his office's performance.

We have spent a large amount of money to install an efficient computer system to see that your magazines are delivered correctly. We go to great lengths to keep up with your moves and we are one of the few publications that can and does change addresses as little as forty-eight hours before mailing. If we receive a change of address request on Tuesday morning—that change is on your label Thursday morning in Fulton, Missouri, where our printer mails *Speaker Builder*.

A growing number of people in Washington are beginning to rethink the question of whether the government should be running many operations that are, in fact, businesses—or should be. The director of the Office of Budget and Management recently suggested that the USPS, or at least some portions of it, be sold to private enterprise. It should be glaringly obvious that if the private sector were allowed to compete with the Postal Service for third or second class delivery, the result would very likely be comparable to what has happened with parcel delivery.

If the new administration in Washington is committed to competition in American business, then I for one believe it is time the government divested itself of the Postal Service. Unfortunately it is a bit late. The Democrats are in power in both houses of Congress and are unlikely to go along with such a proposal from the President. But the business style Postal Service would work a lot better, and faster, even if it became somewhat more expensive. Think about it. And pass along the idea to your Congressman if you believe it has merit. When Barry Goldwater was nominated for the Presidency during the 1960s a joke circulated in the form of a reply to the candidate that went as follows:

"But Mr. Goldwater, the government has always delivered the mail! We cannot *sell* the Post Office."

It seemed like a joke then. It isn't anymore.—E.T.D.

M-S SPEAKER SYSTEM

BY REID WOODBURY

You can add this design to your "gee wiz" stack of novelty speakers. It is based on the M-S (middle-side) stereo miking technique, which is popular for many reasons, but few people seem to understand how this method realizes stereo signals.

The secret with the singular stereo speaker I designed is that the front driver receives the sum of the left and right channels, and the difference is sent to the rear drivers, which are wired out-of-phase, like the M-S microphone (see "Theory"). My original belief was that the left and right channel program material would hover to the left and right of the speaker, but no such luck. I've discovered it's similar to Lauridsen's method (*Fig. 1*) for producing pseudo-stereo from a mono source[1] which my speaker does with some success. However, my single speaker does have an *incredible* sense of depth.

The M-S is my first and only speaker construction project and I continue to startle friends with the sound.

ORIGINAL DESIGN. I came up with the design in 1983, reflecting that I was rarely positioned front and center while listening to my stereo system, but either working on some project, homework, cleaning or cooking, and not positioned optimally to hear the stereo image, often being in the next room. My design produces the same acoustic energy as standard stereo, and since I am generally economical, I figured I shouldn't pay for more speakers than I absolutely needed. For normal stereo I'd just wear headphones.

I started this project just prior to subscribing to *Speaker Builder*. I'd never built any kind of furniture and most of the electronic projects I'd ever finished were kits. I modified my original design as my reading and building experience progressed.

DRIVER SELECTION. I chose drivers only by specs, reputation, and by listening to similar drivers. These weren't my first choices, and in some cases the third or fourth choice, due to availability at the time.

The "front" drivers use a standard three-way design with a Thiele/Small aligned enclosure. The recommended enclosure specifications were supplied by the retailer (McGee Radio & Electronics), which I found very useful and fit closely with Bullock and White's BOXRESPONSE program.[2,3]

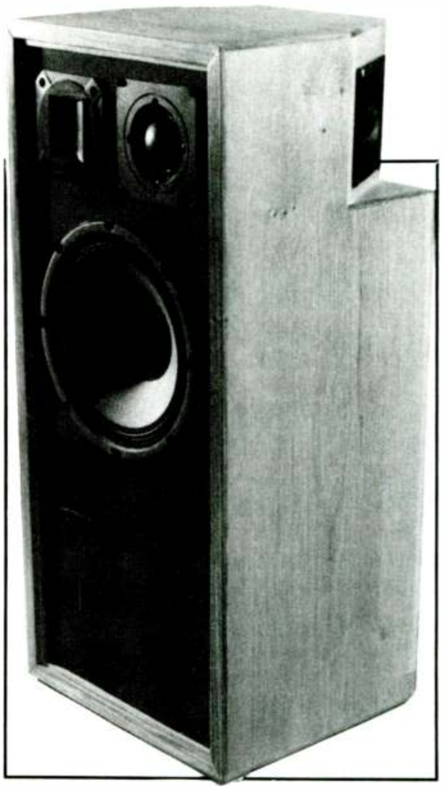


PHOTO 1: Beauty shot of finished speaker.

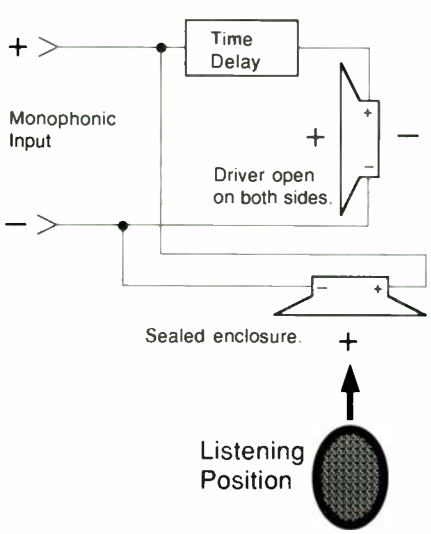


FIGURE 1: Lauridsen's method for pseudo-stereo (from *Sound Recording*, p. 74).

ABOUT THE AUTHOR

Reid Woodbury is a freelance audio-for-video (sweetening) engineer in Los Angeles, working with sound for TV and film, and has done sound designs for various live theaters in the US. He graduated from the University of Missouri at Kansas City Conservatory of Music and also studied two years of electrical engineering. He recorded recitals, concerts and radio programs for the Missouri Repertory Theater and the Conservatory's recording department, and currently runs his own location classical recording service.

1. Eargle, John, *Sound Recording*, Van Nostrand Reinhold Co., New York, 1980, pp. 73–79.

2. Bullock, Robert M., and Bob White, "Box-Response," *SB* 1/84, pp. 13–18, 42.

3. I took the Bullock and White program and typed it into a powerful spreadsheet program which has built-in graphing (Excel™ for the Macintosh). I've spent many hours subtly changing a speaker enclosure's parameters with a particular driver and studying the numbers *and* frequency response graphs. I modified the input of the program, i.e., instead of entering the compliance/box volume ratio, I enter just the box volume and the spreadsheet calculates the ratio. I enlarged my enclosure slightly and I always use the 30Hz rumble filter in my preamp as the program suggests.

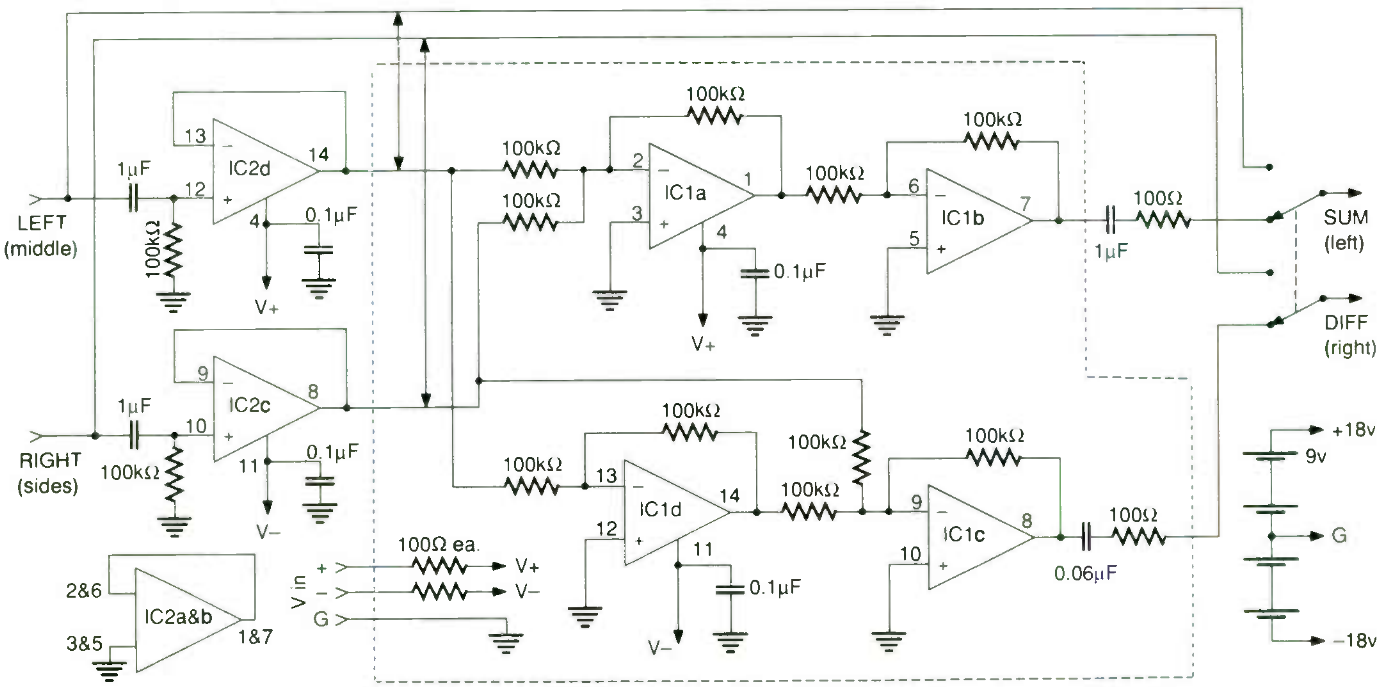


FIGURE 2: Sum and difference circuit with low rolloff. Labels in parentheses mark connections for M-S microphone.

I chose a 10-inch woofer (Eminence EM40WOAE) for my design because I thought it was the best compromise between full-sounding and tight bass. I chose a dome midrange (Peerless PMT51) for its smooth sound and wide dispersion, and a leaf tweeter (Foster E110T08) for its clarity[4]

The crossover is a stock second-order design, rated at 150W, with crossover frequencies of 500Hz and 7kHz. I chose a wide midrange bandwidth because the PMT51 has a much greater power rating (120W) than the other drivers (85W and 50W, respectively).

For the rear drivers, I tried full-range Philips AD5061/M8, but they were not of high enough quality, power, or sensitivity. An album of *Carmina Burana*, with its lush chorus, has so much out-of-phase material I was getting distortion from the amp trying to drive them, and the small speakers were bottoming out. I replaced them with two of the same dome midrange drivers (PMT51) I used for the front.

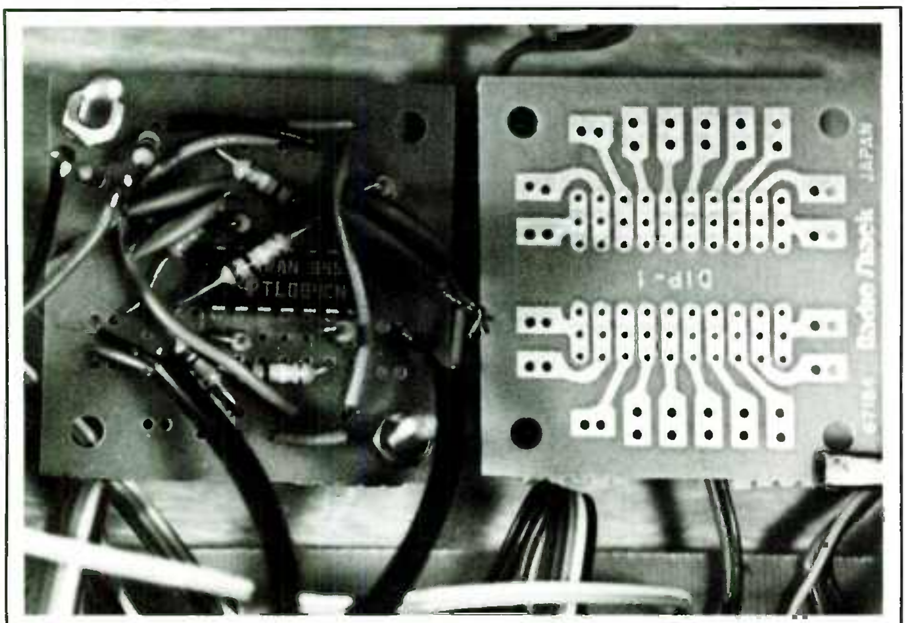


PHOTO 2: Finished board installed in EQ enclosure with an unstuffed board, foil side. Signal hook-up wires are two-conductor shield (left, right, and ground).

PARTS LIST

2	TL084/74s
12	100kΩ resistors
4	100Ω resistors
4	0.1μF ceramic capacitors
3	1.0μF nonpolar electrolytic capacitors
2	0.03μF ceramic capacitors
1	switch, DPDT
1	circuit board, Radio Shack #276-159

Misc.: hardware, screws, nuts, spacers, bipolar power supply or batteries, box, RCA input and output connectors.

In my enthusiasm while choosing the drivers, I'd somehow lost the idea of keeping the cost down. I was learning how much is involved to make a good sounding speaker an excellent sounding speaker. Changing the rear drivers and adding adjustable L-pads to the more efficient front midrange and tweeter were two major changes.

ELECTRONIC CIRCUIT. I naively tried to come up with a passive design to achieve the sum and difference, but this would have caused a tremendous signal loss. The active circuit I designed (*Fig. 2*) is simple and anyone with a little knowledge of op amps could design one. The op amp can be any good quality device that is unity-gain stable or stabilized with proper compensation.

4. McGee Radio & Electronics Corp. *Catalog*, Kansas City, MO, 1983.

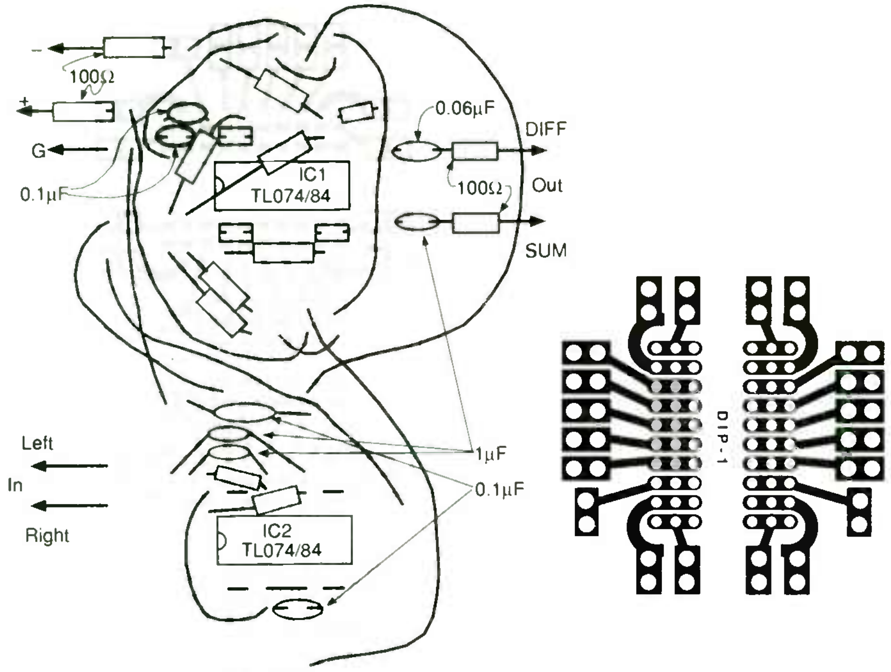


FIGURE 3: M-S circuit pattern and component layout. Resistors are rectangular and capacitors are oval. All resistors are 100kΩ except where marked.

The design works well and parts are readily available (all can be found at a Radio Shack).

Basic Circuit. I added my original circuit to a parametric equalizer kit I'd built some time before. I chose the TL084 quad op amp because it's similar to the TL074s which are used in the rest of the circuit. Not very scientific, but this saved me the trouble of building a power supply and enclosure for such a simple circuit, and I knew this supply would work. I've since discovered many types of IC op amps with faster and quieter specs.[5]

I used a pre-made Radio Shack IC experimenter's board (*Photo 2* and *Fig. 3*) for the project. The "difference" circuit in the *Photo* has about 10dB of gain (330kΩ feedback resistor on IC1c) because the rear drivers on my speaker are wired in series. I suggest you wire in parallel and keep the gain at unity as in the schematic, depending on the load your own power amp can handle. This is one likely cause of the amp's clipping distortion when I was using the AD5061s.

The circuit can be powered by any power supply giving ±6–18V, or by two or four 9V batteries. D'Appolito's power supply is suitable also (*SB* 4/88, p. 20).[6] Be sure not to connect power to the ICs backwards, or this will fry them.

Use metal film resistors for lowest noise. I originally used carbon resistors, but I did hand match the values of the resistors around each op amp to 0.1%. The circuit adds noise to my system, but it is below all my program material sources (LPs, FM stereo, my own reel-to-reel, and digital masters).

The circuit associated with IC2 can be eliminated altogether if you know the output of the device feeding it, and the connecting lines are short enough (less than 2 feet). Inverting amplifiers should be isolated from the "outside world" by a noninverting buffer (IC2c, IC2d—*Fig. 2*). In the schematic, the section within the dotted line is shown in the photo, the rest of the circuit is on the original EQ board. The input buffers are identical to the ones on the EQ board. I picked up the signal after the input buffer on the EQ board and returned it to the same point after the cut in the trace. If you're unsure where the output of the sum/difference circuit will be connected, include the 100Ω resistors and 1µF capacitor.

***BASSIC* CONCERNS.** An advantage of this speaker is the summed version of the bass is the actual program material (see "LP Mastering"). Turntable rumble tends to be out-of-phase, which cancels itself in the front driver and remains in the rear where it can't be heard. This does eat up your system headroom, though, and should be filtered out. I use the matrix circuit in a tape loop, instead of between the preamp and power amp, with the left channel as the front and the right as the rear. This allows me to use the right channel bass control to cut the out-of-phase turntable rumble and control the balance between the front and rear drivers.

In the original circuit, I added two parallel .03µF capacitors to the output of the difference amp (0.06µF) to help roll off the bass. Calculations show the circuit to be 3dB down at 27Hz when feeding a 100kΩ load. The resistance and capacitance can be slightly smaller, depending on the desired low rolloff.

CONSTRUCTION. The speaker cabinet is constructed of ¾-inch particle board (*Fig. 4*). I've included the entire blueprint so you can compare my actual design with my description of the sound. I attached the corners with Elmer's glue and counter-sunk cabinet screws. I added cleats later, after tests with the woofer installed indicated air leaks. I lined the interiors with fiberglass. I mounted all the drivers with bathroom silicone sealant.

I designed a square port because the area required would have been an unusual diameter. I've since learned the port can be almost any diameter with a corresponding change in port length.[7]

The upper chambers were for the original 5-inch full-range drivers.

Initially I thought the speaker would sound best in a corner. This improved the bass response but gave the ambient rear signal a constrained character. The "stereo" image is much wider against a flat wall. I chose a 30° angle for the drivers so they wouldn't be pointed directly into the walls of a corner or, conversely, be too close to the line of sight on the sides mounted perpendicular to the front drivers. I soldered all the electrical connections within the enclosure with 16-gauge stranded wire.

LISTENING IN. This speaker seems to work best with any source material that has major program material coming equally and in-phase from the left and right channels (panned center). This includes M-S miking, stereo trio miking, or pop music where leads and bass are panned hard center, and backgrounds and effects are left and right. This creates a halo of sound around the center, letting the more expensive drivers handle

5. Pennington, Terry, and Larry Winter, "Understanding Circuit Principles," *Recording Engineer/Producer*, March 1987, pp. 60–65.

6. D'Appolito, Joseph and James W. Bock, "The Swan IV Speaker System," *SB* 4/88, pp. 9–21.

7. Bullock, Robert M., and Bob White, "Box-Response," *SB* 1/84, pp. 13– 18, 42.

the important part of the sound. Changing the rear drivers to PMT51s subjectively improves the overall frequency response and seems to give a more even sense of depth, possibly because the front and rear drivers are more closely matched.

I suggest you use a dome tweeter rather than the E110T08. The ribbon's dispersion is so narrow you must be positioned directly in front to really hear it. Otherwise it is a very nice driver with a clean, clear sound.

One problem with playback of recordings made with wider spaced microphones: more out-of-phase material is present (see "Theory"). When the level is balanced between the front and rear, the front drivers will sound weak. If the front level is increased, the image sounds restrained.

Even though the direct sound from the rear drivers wraps around directly to the front, a greater spaciousness is apparent when the speaker sits farther from a wall or corner. Besides the initial direct sound, you hear reflections from more distant walls and surfaces in the room.

THEORY. I prefer minimal miking, like a stereo pair with no highlight mikes, for recording classical ensembles. I've also heard of jazz bands which are recorded with just a stereo trio (left-center-right, LCR). Even with minimal stereo miking, differences in sound can be great, particularly recordings of classical music in a concert hall, caused by different stereo miking techniques, but different musicians and halls add variables, also.

In designing sound for live theater (sound effects, music and live miking), I know speakers and their placement must fit the miking techniques used to record the effects or music. I've seen *SB* articles on driver and speaker design, but nothing on the program material they reproduce. Let's review the many different ways sound is recorded, and presented in the home; you can adapt my speaker design, or any other design, accordingly.

STEREO MIKING. Two audio cues help us perceive direction between two loudspeakers: relative *intensity*, which is the most significant, and relative *arrival time*. The sound will appear to come completely from the right speaker when the intensity of the right channel is 20dB greater *or* the arrival time is 2msec sooner than the left. This time difference is also known as the *precedence* or *Haas* effect.[8] Lowering the level in the right speaker about 8–10dB, while delaying the left up to about 25msec will keep the stereo image centered, but will broaden the image between the speakers proportional to the amount of delay.[9] Beyond 25msec you start to hear a distinct echo.

Different stereo miking techniques, such as X-Y, ORTF,[10] Blumlein,[11] and M-S, use coincident (one microphone directly on top of the other), slightly spaced, and widely spaced pairs; and trios.

Slightly spaced pairs use both time delay (the Haas effect) and level differences to give localization. Widely spaced pairs generally use omnidirectional (equal pickup in all directions) mikes, therefore only the Haas effect produces localization. The time delay is created by the distance between the two mikes, most apparent with sound coming from the sides.

Coincident miking uses only level differences for localization because there is no delay between mikes for sound traveling from the sides.

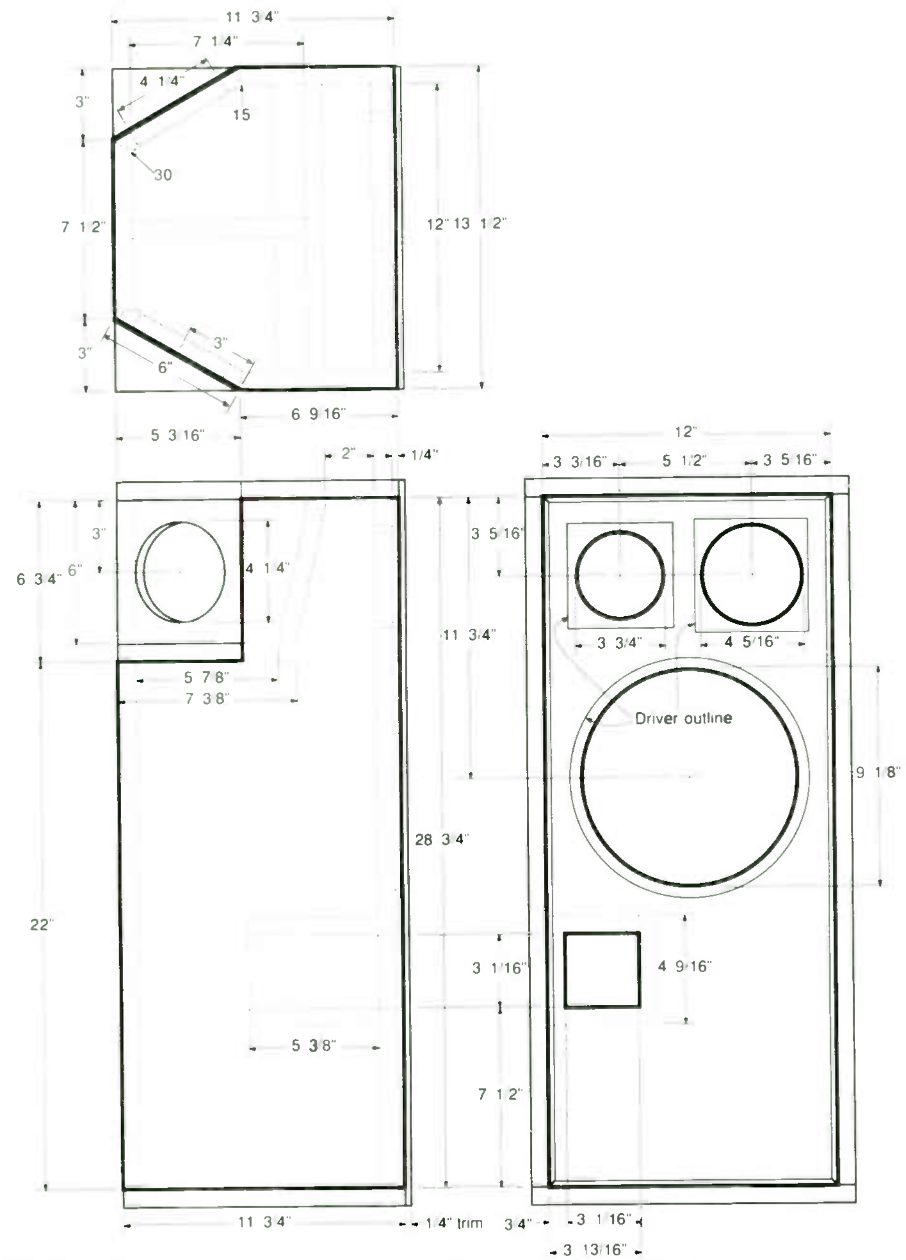


FIGURE 4: Blueprint of enclosure. Panels were cut from ¾-inch particle board.

8. *db: The Sound Engineering Magazine* Dec. 1979, pp. 42–46.

9. Eargle, John, *Sound Recording*, Van Nostrand Reinhold Co., New York, 1980, pp. 41–42.

10. ORTF is a microphone setup specified by the French Broadcasting Organization where two cardioid mikes are at a 110° angle and 17cm apart.

11. Blumlein or "stereosonic" miking uses two coincident figure-eight mikes at a 90° angle.

X-Y generally refers to a coincident or near-coincident stereo pair of mikes, with one panned hard left and the other panned hard right, which can be applied to all these techniques except M-S (which uses a simple electronic matrix to derive the left and right channels).

Music in a studio is generally recorded with a large number of single mikes and each is assigned a position in the stereo image by panning or adjusting only the relative level between the two channels. Signal delay is used only as an effect, that is, reverberation, echo, flanging, and chorusing.

Middle-Side. The M-S method uses a coincident pair, however one is a cardioid pickup pattern mike pointed directly toward the ensemble (middle), and the other is a perpendicular (side-to-side) figure-eight pickup pattern mike (*Figs. 5a* and *5b*); the positive lobe is usually pointed to the left.

The electronic matrix is a simple sum and difference, unity gain mixer. The sum of the two mikes is panned hard left and the difference is panned hard right. Varying the level between the two microphones allows you to adjust the width of the stereo image (*Figs. 5c–f*). The points where the microphone pickups are equal, or where their pattern lines cross, indicate where a musician would be positioned to be heard from the left or right speaker. The negative rear lobes which result from combining the two mike signals, put some reverberation out-of-phase in the stereo signal, so the reverberation seems to come from outside the normal stereo stage between the speakers.

The closer the microphone elements are matched, the clearer the stereo image; the best results are obtained with stereo microphone where the elements are factory matched.

Inverting the phase of one microphone will cause the left and right signals to exchange outputs. This is handy if you accidently place the figure-eight mike backward or hang the stereo mike upside down.

M-S stereo (or other coincident techniques) is also perfectly mono compatible. When you add left and right channels together, the figure-eight sides cancel, leaving only the middle cardioid mike signal, as if only one mike was used. Commercial FM radio also uses this matrix to broadcast in stereo and be mono compatible. M-S recordings are more likely to broadcast unscathed.

My M-S speaker matrix circuit can also be used for the M-S microphone matrix. The signal from the microphones must go through an appropriate mike preamp *before* going through this matrix. This design is too noisy for professional applications, though. If there's a demand I'll make a more esoteric design available, but the corresponding parts may be hard to find. In addition, I suggest you read Walt Jung's *Audio IC Op-Amp Applications*.[12]

X-Y miking can sometimes suffer from a "hole in the middle" effect. Performers situated at the center of the microphones are off-axis from both, and the recording sounds different than the live performance. One cure for this hole is to add a third, center mike with its signal going equally to both the left and right channels.

M-S miking doesn't suffer from this problem; the middle element is pointed toward the center. Both M-S and stereo trio miking have the advantage of three on-axis directions, giving a much more pleasant recording of the concert hall sound. While M-S requires only two microphones or one stereo mike, it does require the electronic matrix. Even though Blumlein's method has four on-axis directions, a soloist is still off-axis from both microphones. Interestingly, performers tend to prefer M-S, while engineers prefer ORTF.[13]

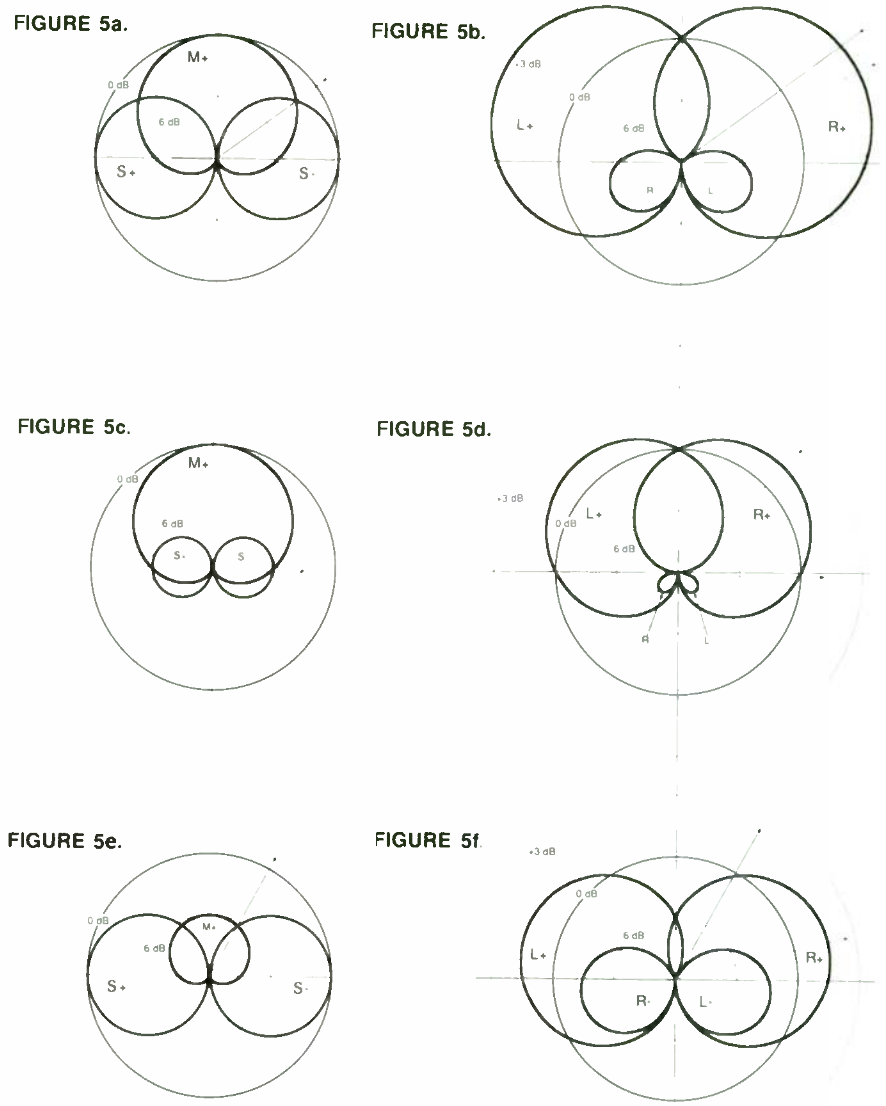


FIGURE 5: Cardioid (M) and figure-eight (S) microphone pickup patterns and their resulting matrixed patterns (L and R). Drawings are on a linear scale. The dotted arrow is the "on-axis" direction of the right matrix output. The solid arrow is the performer's physical direction from the mikes, to be heard only from the right speaker in standard stereo.

LP MASTERING. I believe you can't detect the direction of a signal source below about 200Hz, except by moving around and listening closer and farther away from the source. The wavelength is long enough to go around the head and affect both ears equally. The same effect occurs when recording with coincident or near-coincident microphones. Also, excessive out-of-phase bass, which on LPs could cause the stylus to skip out of the groove, is filtered out during disc mastering, using an elliptical equalizer and leaving only the in-phase bass.

Left and right (lateral) stylus motion is center program material, and up and down (vertical) motion is out-of-phase or ambient material. The popular Neumann system has switchable turnover frequencies of 150Hz and 300Hz.[14, 15]

Even on CDs, because of the miking techniques and LP engineering habits, the low-frequency information tends to be the same in both channels. In regular stereo (or P.A. setups, for that matter), out-of-phase bass will sound weak for the same amount of program power. Still, I prefer stereo using two woofers. Recordings are likely to be monitored and adjusted with two equally full-range speakers and then checked on one full-range speaker for mono compatability, rather than on an odd stereo setup with different frequencies getting a different number of speakers. Boosting the bass of the front driver of my M-S speaker at all frequencies below that of the rear drivers (about 300Hz with the PMT51s) takes care of this consideration, but is not a perfect solution.

TV STEREO. Many movies are broadcast or released on video tape with the original surround encoding of a Dolby stereo optical print. The two-channel recording is decoded to four channels (Dolby started by using a technique similar to the Sansui QS quadrophonic matrix[16]), but, instead of speakers in the four corners of a room, the new format specifies three speakers across the front—left, center, and right; the surround channel is the fourth.

Soundtracks must pinpoint information across the front and keep the center material in the center because phantom center material of two-channel stereo drifts toward you if you sit at the sides of a listening space. The closest speaker is louder and its sound arrives sooner. This is not as big a problem in the home, though dialogue and effects in stereo television shows are kept fairly close to center in the stereo field to keep the sound of the on-screen action from running off the small screen. The 70mm, six-track stereo format does not use this encode/decode process.

When a theatrical show is mixed, the dialogue program material is always given to the center channel; never panned from speaker to speaker. Through psychoacoustics (the study of how people perceive sound), movement, even following the characters across a screen, was found to be distracting. Panning is only used for an effect. On the set, the action and dialogue are recorded by a shotgun or other narrow pattern microphone. If a stereo recorder is used, one track gets the main mix and the other is used for backup—rarely a stereo mix. Replaced dialogue is recorded mono and effects are recorded mono and stereo. Room tone and other atmospherics may be recorded in stereo after the main "take." On the simplest stereo shows, only the music is stereo.

The surround channel also has a very limited bandwidth, 500Hz–7kHz.[17] (Matching these frequencies with the front crossover of my speaker was unintentional.) Surround information is kept to a minimum for compatiblity with mono-only theater systems. Any bass information comes only from the front speakers. The 7kHz low-pass filter keeps distracting optical noise out of the surrounds. You can design your own home surround speakers with a limited bandwidth, also.

The simplest surround decoder takes the left and right recorded signals (left total and right total) and sends them to the left and right speakers; makes a sum and sends it to the center speaker; and takes the difference signal and sends that to the surrounds. At the other end of the spectrum, professional theater decoders use a complex phase detection, steering circuitry and surround-channel delay to increase separation between speakers. Home units fall somewhere in-between. Interestingly, M-S recorded material is easily encoded and decoded through the Dolby process. The middle mike is sent to the center speaker, matrixed left and right to the left and right speakers, and the surrounds are fed by the figure-eight mike (which picks up mostly reverberation).

This mix type works well with my M-S speaker. It's great for television or stereo video, since sound effects tend to run off the picture with a normal stereo setup. For movie theater systems, recordings are mixed with the left and right speaker behind the screen, not on each side. With the M-S speaker, the action takes place in the center, and the music and atmospherics take place all around. As with the M-S microphone, the M-S speaker has a direct center element so center program material has an actual driver and not a phantom combination of other elements to produce the sound.

My 13-inch color television sits comfortably on my speaker, but the magnets in the speaker cause the color to change. The sky turns red, grass turns yellow, and faces turn green. I stacked three large phone books under the television and this works fine. If you're going to add this speaker to your television, be sure to include magnetic shielding.

OTHER IDEAS. The left and right signals don't recombine on the left and right sides of the speaker because of many inaccuracies, such as diffraction problems from the recessed front panel, and drivers not vertically aligned. I've come up with a few possible cures you might try, also, the best application I've thought of for this design—a "ghetto-blaster."

Front Delay. Mark Rumreich's electronic time delay project (*SB* 3/88) would be perfect to try on this speaker.[18] I agree with his philosophy that sound from each driver must reach the listener at the same time for the best imaging. Try to delay the front signal so it will reach you at the same time as the rear signal, for my design. The amount of delay *may* depend on the distance between your front and rear drivers *or* on the length of the sound's path from the rear drivers, reflecting off a wall, to the plane of the front drivers. If you align the front signal to match the reflections of a speaker sit-

Continued on page 64

12. Jung, Walt, *Audio IC Op-Amp Applications*, 3rd ed., Howard W. Sams & Co., Indianapolis, IN, 1987.

13. Cross, Lowell, "...Assessments of Studio Microphones," *Recording Engineer/Producer* April 1984, pp. 120–129, and Dec. 1985, pp. 70–85.

14. Runstein, Robert E., *Modern Recording Techniques*, Howard W. Sams & Co., Inc., Indianapolis, IN, 1974, p. 300.

15. Eargle, John, *Sound Recording*, Van Nostrand Reinhold Co., New York, 1980, p. 298.

16. Blake, Larry, "Mixing Dolby Stereo Film Sound," *Recording Engineer/Producer*, Feb. 1981.

17. Blake, Larry, "Mixing Techniques for Dolby Stereo Film and Video Releases," *Recording Engineer/Producer*, June 1985, pp. 94–107.

18. Rumreich, Mark, "Electronic Time Delay Line for Speakers," *SB* 3/88, pp. 15–21, 67.

A STEREO SYSTEM ODYSSEY

BY ARTHUR BROWN

I started building speakers in 1950, with a back-loaded horn driven by a cheap 12-inch driver and a horn-loaded high frequency speaker. I attempted to utilize the room corner to extend the low-frequency horn's mouth opening. I thought the design should work great, but single frequency input came out with 100% second harmonics at frequencies below the driver resonant frequency, and nothing I tried improved this.

My friends were building speakers also; infinite-baffle, bass-reflex (ported) and tuned-pipe enclosures. We frequently got together to make measurements and our group concluded the bass reflex gave the cleanest sound and widest range, perhaps because it had the best driver. I continued to try to improve my speaker but finally gave up and bought an AR-1 in 1960. I added a stereo receiver and phono system, but never believed full stereo was worth having until 1986, when my son took me out to listen while he auditioned systems to buy.

I became hooked again. Stereo sounded like I thought it should, rather than like disconnected sound from two separate boxes. After that, my son borrowed several copies of *Speaker Builder* for me to read. One issue included the BOXRESPONSE article, so I set off to build myself a stereo speaker system.

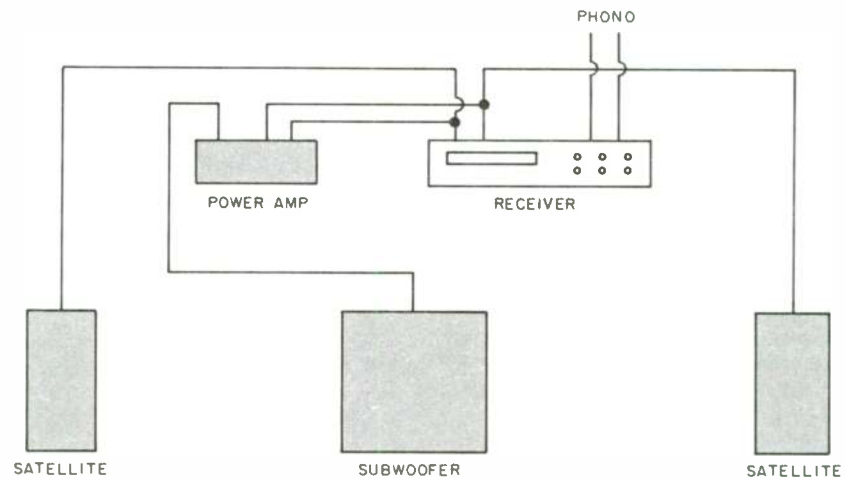


FIGURE 1: System concept.

CONCEPT. My son did buy a sound system, and this gave me the following equipment to work with: a 35W stereo receiver, my son's old 35W mono amplifier, the stereo phonograph player and the AR-1 speaker. In my design, shown in *Fig. 1*, the receiver drives two satellite speakers and the summing inputs to the subwoofer power amplifier.

I planned on using the AR-1 as the subwoofer. Its response was clean, and extended to the low frequencies. I was aware that the stereo content of music was minimal at the low frequencies and decided to drive the subwoofer with the mono amplifier, using a 100 to 150Hz electronic low-pass crossover, a part of the new preamp that also provides the summing and gain adjustment.

In selecting drivers for my satellites, I wanted the best transient response I could find. A further objective was to design the speaker enclosure to provide *the low-frequency cutoff*, to avoid a low-frequency passive crossover network for the satellite speakers. Finally, I wanted drivers capable of crossing over at 2–3kHz at the high end. With this frequency span available in the one driver, I believed I would be covering the range in which the principal music content occurs.

SATELLITE DESIGN. I proceeded to develop the design using BOXRESPONSE. I also used Max Knittel's Maximum Effort Software program (*SB* 1/83). I set up my Lotus spreadsheet to calculate the box's physical design. I chose the Focal 5N401 driver for the low end. It had a free-air resonant frequency of 43Hz and a high-end response well above 3kHz. I chose a Dynaudio D-28AF for the tweeter. Its resonant frequency is around 700Hz and well-damped, lending itself well to the selected crossover. These two drivers have some of the highest acceleration factors of all those I could find, promising very good transient response.

I purchased two of each and, after a break-in period, tested them for all of the driver design parameters. The measurements were very close to the published data. I completed the design on the computer and built the boxes. The box for the 5N401 and D-28AF (*Fig. 2*) is made

ABOUT THE AUTHOR

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

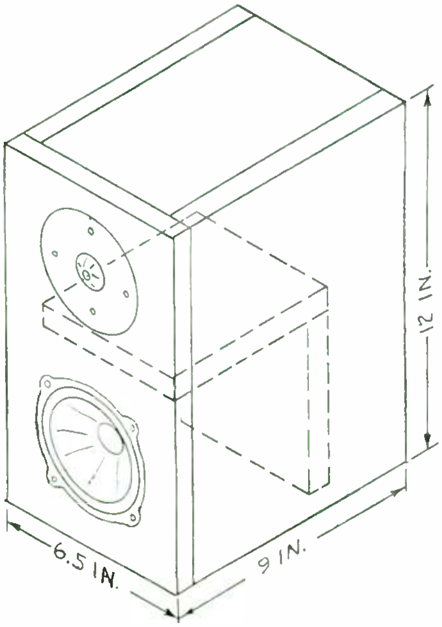


FIGURE 2: The satellite's initial design.

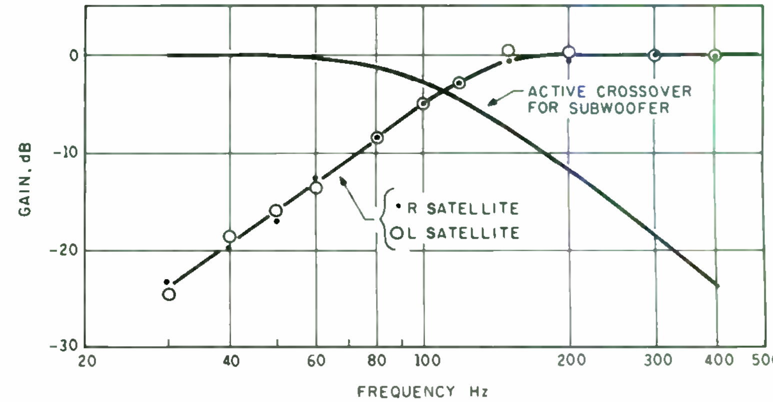


FIGURE 3: Crossover data.

of ¾-inch industrial-grade particle board. I kept all wall areas of the 5N401 enclosure below $25in.^2$ to minimize wall vibration. This phase of adjusting the size of the enclosure was easier with the Lotus program. I lined the walls with ¼-inch sheet asphalt compound as damping material and added long-hair sheep's wool for high frequency absorption.

I mounted the D-28AF driver just above the 5N401 in an open-back structure, which also provides space for the crossover components. I adjusted the volume of the boxes to get the high-pass crossover curves to within ±1dB of each other. I obtained a crossover frequency (3dB down) of 110Hz with a 62.5-in.^3 box volume. The data, which I used to design my active crossover network, is shown in *Fig. 3*, with the crossover response.

I ran frequency response data on both drivers in their enclosures in my listening room using a small electret condenser mike—a Panasonic P9932, purchased from Digi-Key. The individual response curves were quite variable when I placed the mike 39 inches from the driver (room effects and speaker irregularities, I thought). I repeated the frequency response test with the mike within 1.5 inches of the front baffle to eliminate most of the room effects, and obtained smoother data.

I still had considerable variability and elevated response from the D-28AF. I therefore chose to use a third-order passive crossover to be sure I could transfer between drivers quickly and avoid superimposing the apparent irregularities of the drivers on each other. My crossover attenuates the D-28AF driver 5dB to bring the two drivers to the same apparent sound pressure level.

I also chose to use a Zobel to handle the 5N401 driver's impedance rise. I used a computer program, purchased from Old Colony Sound Lab, to calculate the component values for the crossover and Zobel design. I developed the Zobel design in breadboard, and then modified the components, based on actual tests to obtain the flattest impedance curve on the driver.

I built the crossover and after listening, I thought high frequencies were missing. I then tried the crossover without the attenuation and decided the sound was more balanced. The final crossover design, *Fig. 4*, is for 2,250Hz. All parts are high quality components purchased from Madisound Speakers.

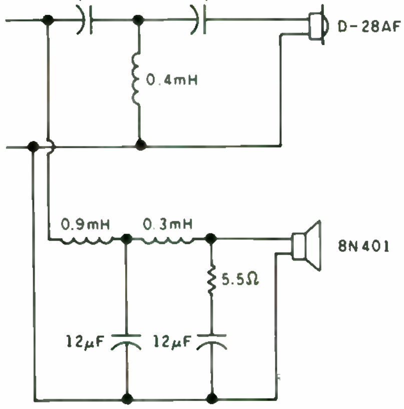


FIGURE 4: Crossover.

ALIGNMENT. My reading convinced me I should align the drivers by time. I tested the satellite drivers by placing a mike in front of and at a fixed distance from the mounting panel of each driver and drove each speaker with a square-wave signal. To view the square wave to the speaker and the response of the mike, I used a dual-channel oscilloscope and photographed the display. From the scope sweep calibration (msec/cm.), I calculated the time from the square-wave signal to the mike response and multiplied this by 13,500 in./sec, to obtain the distance from the mike to the apparent sound source for each driver. From the data on each driver I determined the distance between the apparent sound sources.

The signals from the drivers are shown in *Fig. 5a*. The D-28AF driver responded first, then the 5N401 without the crossover. I interpret this apparent 1.1-inches to be the distance from the the D-28AF dome to the 5N401 voice coil dome. The distance corresponds approximately to the measurements of the parts.

When I looked at the 5N401 driver

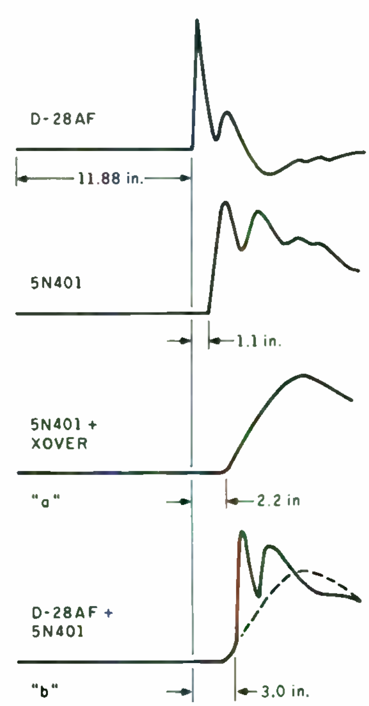


FIGURE 5: Time delay; (a) driver response differences and (b) matched response.

PHOTO 1: Satellite enclosures, reworked to set back the D-28AF.

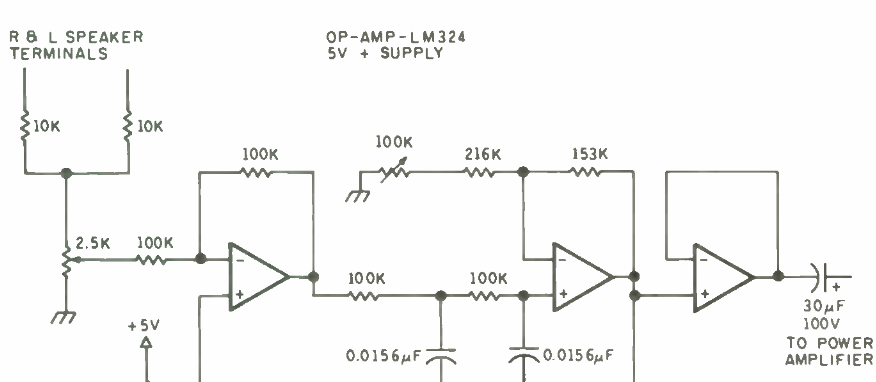


FIGURE 6: Preamplifier.

with the crossover network in place, I found the response was heavily lagged; the apparent distance had increased to 2.2 inches, caused by the crossover. I decided that I could set the D-28AF driver back 2.2 inches or more, depending on how I wanted to match up the two transient responses. I chose a 3.0-inch setback to superimpose the two curves (*Fig. 5b*).

I reworked the original satellite enclosures to set back the D-28AF, shown in *Photo 1*, and placed ½-inch felt around the drivers and on the "shelf" to avoid radiating surfaces. Without the felt the sound was alive and as evident at the sides as at the front. There was less difference between loud and soft passages than I expected. When I added the felt to the tweeter and shelf, the sound stage became more stable and had more depth and dynamic range. I added the felt around the 5N401 driver and the whole speaker seemed to become transparent.

The satellites reproduce sound clearly; I can hear the nuances of the instruments and the recording hall ambience on the record for the first time. I judge the satellites to be excellent sound sources for an LEDE (live-end dead-end) type listening room, to which I'm partial.

Photo 2 shows the crossover components mounted in the rear of the box. Eventually, I must build a screen to cover this weird box so my wife will allow it in the house on a permanent basis.

In my time delay exercise, a single-order crossover may impose a less severe lagging effect on the 5N401 driver and improve the transient response of the driver, if the drivers' frequency response curves are smooth enough to allow this. I have not investigated this idea. Please see the sidebar, *Commercial Speaker Alignment by Time*, for another application of this technique.

SUBWOOFER AMPLIFIER. To use the AR-1/power amplifier as a subwoofer, I built a preamp (*Fig. 6*) that sums the right and left inputs, and provides gain adjustment and the low-pass active filter for the crossover. The preamp uses three sections of an LM-324 quad op amp, and a +5V power supply. I designed the active crossover using the specific approach described in Don Lancaster's *Active Filter Cookbook* (Sams). I heartily recommend this text for this kind of circuit design.

The first stage attenuates the signals from the receiver's left and right speaker connections, which reach 16.73V at 35W; the input to the subwoofer amplifier needs only 1.0V for full power. I use the gain control to adjust the subwoofer sound pressure level to match the satellite sound pressure level. The summed signal then passes through the active filter amplifier and a buffer amplifier prior to going to the power amplifier. The active filter circuit includes an adjustment to set the damping of the crossover curve.

The system sound with the two satellites and the AR-1 subwoofer was very satisfactory. I placed the satellites on bookshelves, about 4.5 feet above the floor (I prefer 3 feet) and 10.5 feet apart. The sound stage is strong and stable, 5–18 feet from the satellites, in my listening room, 19-feet long. I positioned the subwoofer on the floor between the satellites; the driver is 12 inches above the carpeted floor.

I kept wondering, however, if I could improve the subwoofer's performance. I knew that the AR-1 subwoofer has very low efficiency and was limiting the system's maximum power. I also thought I could improve the transient response of the subwoofer. I first experimented with an Isobarik configuration by mounting a 5N401 in front of a satellite. This configuration extended the low frequency cutoff by about 5Hz.

I used the computer to select suitable drivers for an Isobarik, assuming that a well-designed acoustic suspension enclosure would be the proper backup for the Isobarik. I wanted to try the Focal 10C02 driver, but I ended up buying two 8N401 drivers because I could not find my first choice. I built the Isobarik into a new box with the drivers facing forward. The driver's resonant frequency was 44Hz in the box, compared to 32Hz free air. I operated with it for a while. It appeared to be better in transient response than the AR-1 and had some extended low frequency response. It was still limiting the system power level because I now had to divide the amplifier power between two drivers.

I went back to the computer and looked at a ported enclosure, a design I had not liked in earlier times. However, now I could see benefits: full power to the one driver and somewhat improved low-frequency response compared to the Isobarik design, using an 8N401 driver in the same box I built for the Isobarik. This is my current subwoofer. It has a box volume of 2.41 ft.^{3} and a 3-inch diameter by 7-inch long port. The port is mounted on the front panel of the enclosure, 9 inches from the 8N401. I made this enclosure from the same ¾-inch industrial-grade particle board I used for the satellites, with braces to limit panel vibration. I applied sheet asphalt to the inside walls for damping. I also attached felt across the face of the enclosure except the driver and tube openings.

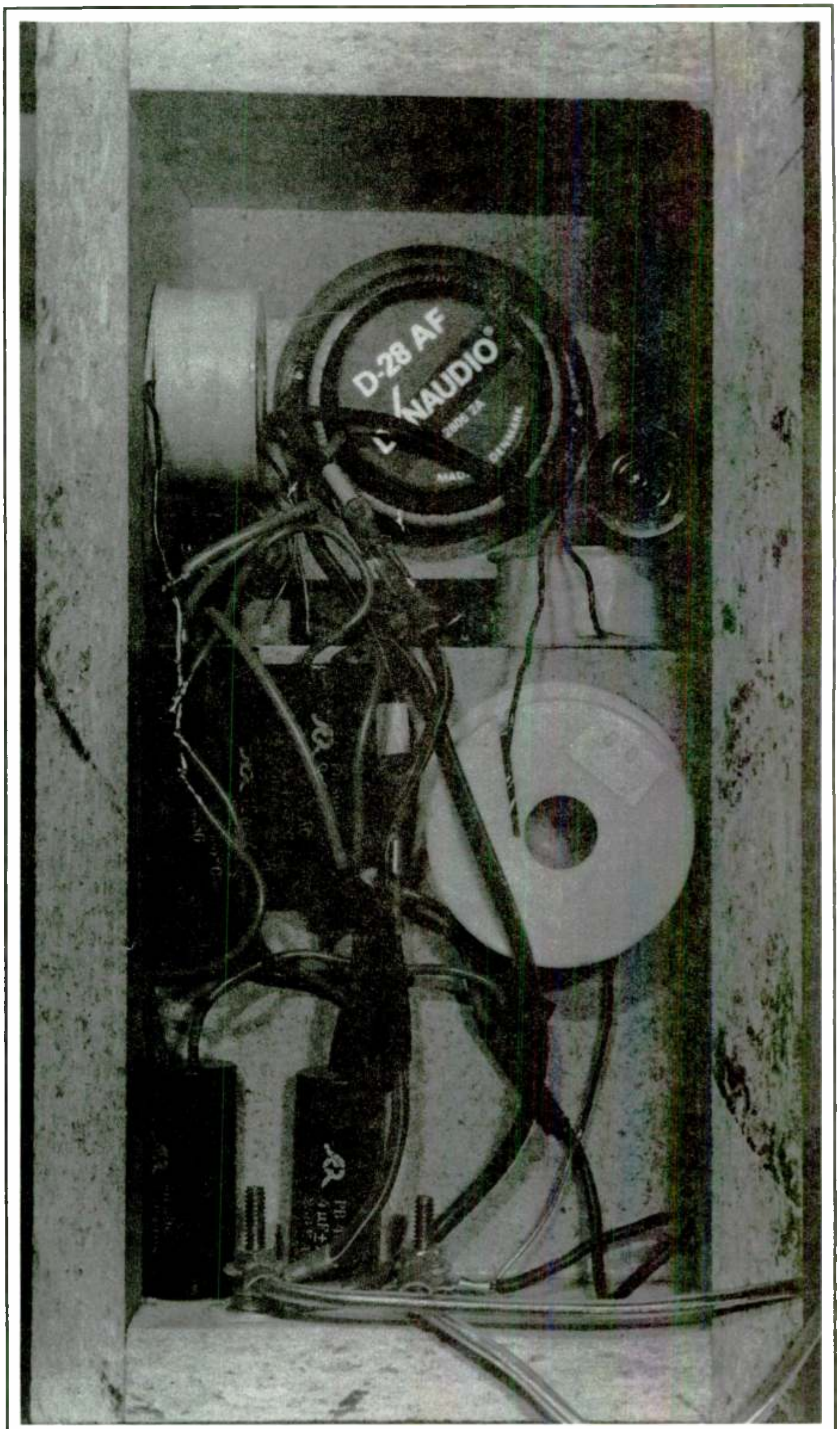


PHOTO 2: The crossover components mounted in the rear of the box.

The overall system frequency response is shown in *Fig. 7*. I measured this data with a Radio Shack sound pressure meter (No. 33-2050); typical calibration curve shows a rise of 6dB, starting above 3kHz, peaking in the 7-8kHz region, and a rapid fall-off from there. Therefore, I conclude the high frequency data is reflecting the sound pressure meter response and is probably reasonably smooth. I believe the dip in the subwoofer response at 40Hz is the crossover between the port and the cone output, and might be eliminated by resizing the enclosure.

I measured the satellite data with the sound pressure meter 9 inches in front

Commercial Speaker Alignment By Time

PHOTO A: The modified speaker without its grille cloth.

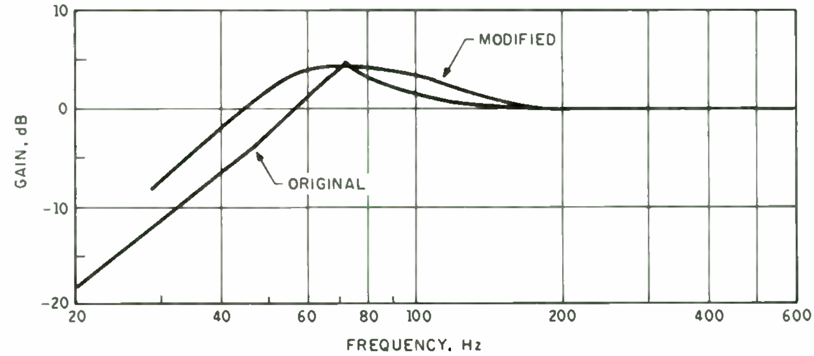


FIGURE A: Effect of alignment by time on the AR-30.

The speaker system my son purchased—an AR-30, Connoisseur Series—is a two-way type, with a 10-inch woofer, crossing over to a 1-inch dome tweeter at 1.8kHz through a second-order crossover. The enclosure is intended for vertical mounting with the tweeter just above the woofer driver, both mounted flush to the front panel. The front panel has rounded edges but no other treatment except the grille cloth.

The frequency response is specified as 46Hz–22kHz, with a Q of 1.1. In the audio showroom these speaker gave the best balance and non-imposing performance of any in my son's price range. In his home, however, he heard more bass boom than he liked for the music he played. He also was not happy with the imaging or clarity of the sound.

I had just finished aligning my satellites. My son had heard them and thought they were great. I suggested we could do the same for his AR-30s, using an extension to move the woofer forward of the tweeter, by the amount we wished for the alignment. We also anticipated a second benefit; that the extended cabinet's added volume would reduce the Q of the system and extend the frequency response.

We made frequency response, impedance and time delay measurements on the speakers as they came out of the box. The data suggested that the displacement of the two drivers should be three inches. Note that the 10-inch woofer has a deeper cone and its apparent sound source is physically farther from the tweeter dome, compared to my satellite. But since the system uses a second-order crossover, the lag-time is less and thus, the combination resulted in a 3-inch time delay, the same displacement as my satellite.

I built 3-inch extensions for the woofers, using mounting brackets so we could use the same T-nuts on the cabinet originally used for mounting the woofer (*Photo A*). We could reverse ourselves if we did not like the new sound. Once installed, we added ½-inch felt around the tweeter and on the shelf to improve the imaging. We tried felt on the edges of the woofer extension with no clear gain in sound performance, and later removed it.

We ran the frequency response, impedance and alignment tests on the modified speaker. The time delay measured as planned. The resonant frequency shifted to 51Hz from 57Hz. The frequency response is shown in *Fig. A* and shows a −3dB frequency of 38Hz, compared to 46Hz before the modification. The rise due to the damping was the same. Listening to the modified speaker, however, the boom seemed to be less evident, perhaps because the rise in the modified speaker is extended over a wider frequency range. The sound was crystal clear, without the muddiness that was apparent on some types of music. The modified speaker appeared to be significantly better.

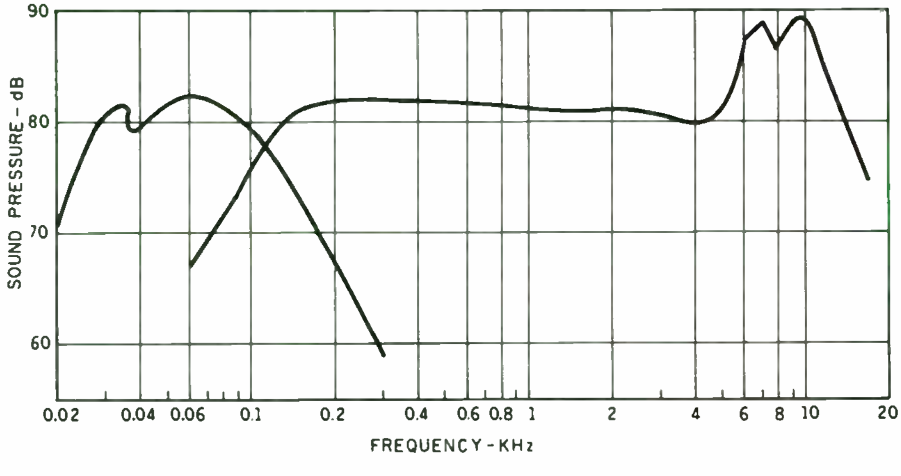


FIGURE 7: System response.

of the enclosure, between the drivers. For the subwoofer, I placed the sound pressure meter at 9 inches, but more in front of the tube than the speaker. I found no difference in the dip in the curve or in the level and shape of the 8N401's signal as I moved the mike around. I found the sound pressure higher from the tube with the mike more in front of the tube. I thought this data might be more representative of the sound going into the room.

The sound from the subwoofer "supports" the stereo signal and the dip is not noticeable in the concert and jazz music I play, or in voice programs on FM. I

Continued on page 64

A T/S DRIVER PROGRAM

BY DAVID LONG

A computer can greatly decrease the amount of time spent crunching out a series of mathematical equations, compared to using a hand-held calculator. The following program, which I wrote for the Commodore 64, will make the job of determining the Thiele/Small parameters of raw drivers much easier and faster.

I also present a program, CLOSED BOX, based upon OPTIMUM BOX (a vented box program) that determines the response of a driver in a sealed box. Additions to OPTIMUM BOX will be given that are used in CLOSED BOX to help make using it, along with T/S DRIVER PROGRAM and BOXRESPONSE, a more complete and interactive package.

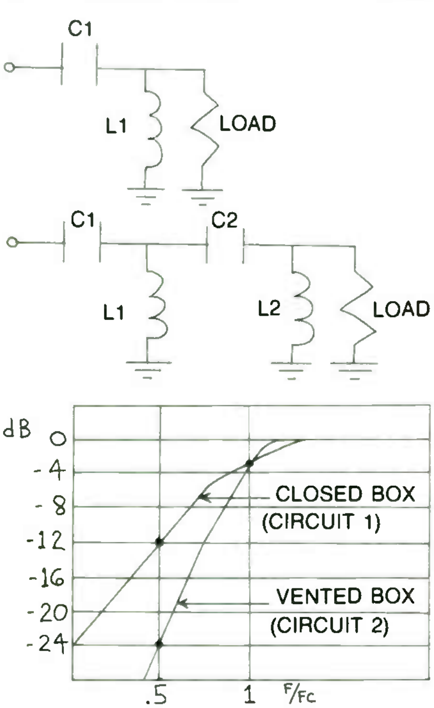


FIGURE 1: (a) Circuit 1, 12dB/octave; (b) Circuit 2, 24dB/octave.

A SHORT HISTORY. The low-frequency response of a driver-box combination can be shown to be the same as a high-pass electrical filter (*Fig. 1*). A closed-box response (Q_{tc} = 0.707) rolls off at 12dB/octave (second-order filter) and a vented box rolls off at 24dB/octave (fourth-order filter).

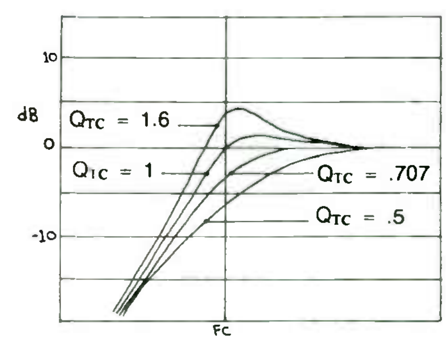
If you place a driver in a closed box that is smaller or larger than "optimum," then the Q_{tc} of the system changes. As *Fig. 2* shows, a system with a high Q_{tc} will have a large peak in the bass response and will probably sound "boomy." A low Q_{tc} will have reduced bass output and may sound "thin." By adjusting the box volume you can obtain any number of Q_{tc} values. A Q_{tc} of 0.707 is considered "optimum" because the −3dB point has decreased as low as possible in a closed box.[1]

By placing the same driver in a vented box, the −3dB point will be lower, but at the expense of a larger box. Some closed-box alignments, however, with a Q_{tc} below 0.707 would require a larger box than a vented system (see the sample driver later in the article).

To know what size enclosure to build for the best performance with any given driver, you must measure the electrical and mechanical parameters of the driver (Q_e, Q_m, Q_t, V_{as}, and so forth). Methods for measuring these parameters were first introduced by L.L. Beranek and J.F. Novak back in the 1950s.[2] A.N. Thiele published the most detailed work describing the driver/vented box, low-frequency response and the mathematical equations that are involved.[3] Richard Small further refined Thiele's work and went on to publish a series of papers describing the response of a driver in a closed box.[4] Thus, Thiele and Small are the most well-known of a number of people who have contributed to our knowledge of box/driver-as-filter theory.

For a more complete and updated method of driver measurement, I highly recommend the fine articles written by Robert M. Bullock that have appeared in previous issues of *Speaker Builder*,[5] and

1. Knittel, Max and Rod Rees, "Manrique Response," *SB* 3/81, pp. 38–39.
2. Beranek, L., *Acoustics,* McGraw Hill, 1954.
3. Thiele, A.N., "Loudspeakers in Vented Boxes," *JAES*, May–June 1971.
4. Small, R., "Closed-Box Loudspeaker Systems," Part 1–2, *JAES*, Jan.–Feb. 1973.
5. Bullock, R., "Thiele, Small and Vented Loudspeaker Design," *SB* 4/80, 1–3/81, 1/82.



EXAMPLE DRIVER
QTC

	.5	.707	1	1.6
FC=	27.7	39.2	55.4	88.7
F3=	43	39.2	43.5	56.8
VB=	5.56	1.73	.73	.26
PEAK=	0	0	1.25	4.5
FREQ=	-	-	78.3	98.8

FIGURE 2: Driver bass response in a closed box that is smaller or larger than "optimum."

INPUT

```
DRIVER ID # ?? DRIVER FROM TABLE 1
DRIVER D.C.RESISTANCE?? 4.3
CALIBRATION RESISTOR DCR?? 5
CONSTANT VOLTAGE OUTPUT?? .100
FREE AIR RESONANCE?? 20.5
CURRENT IN AMPS AT FS?? .003
I1= 8.3526907E-03 AMPS
FREQ. BELOW FS WHERE I=I1 ?(F1)? 12.4
FREQ. ABOVE FS WHERE I=I1 ?(F2)? 34.2
F'S= 20.5932028 HZ
IS F'S OK (Y,N)?? Y
VAS CALCULATIONS
1=CLOSED BOX,2=ADDED MASS? 1
BOX RESONANCE (FC)?? 35.3
CURRENT IN AMPS AT FC?? .005
I2= .0107832773 AMPS
FREQ. BELOW FC WHERE I=I2 ?(FL)? 24.2
FREQ. ABOVE FC WHERE I=I2 ?(FH)? 52.3
F'C= 35.58 HZ
IS F'C OK (Y,N)?? Y
TEST BOX VOLUME=(FT3)?? 2
DRIVER RADIUS IN INCHES=? 4
AMP RESISTANCE =? .1
XOVER RESISTANCE=? .35
```

OUTPUT

```
DRIVER FROM TABLE 1

RE  = 4.3 OHMS            RG  = .1 OHMS
FS  = 20.5 HZ             RX  = .35 OHMS
IM  = 3E-03 AMPS          Z   = 33.333 OHMS
F1  = 12.4 HZ
F2  = 34.2 HZ             MD  = 67.51 GR
F'S= 20.59 HZ             CMS= 8.93E-04 M/N
QM  = 2.63010173          VAS= 4.6116 FT3
QE  = .389532862          EFF= 2.754E-03
QT  = .339283123          SPL= 86.4 DB
QES= .430297929           LOSSES=-.86 DB
QTS= .369797234           SPL= 85.54 DB

ANOTHER DRIVER?(Y,N)? N
```

FIGURE 3: T/S DRIVER PROGRAM input and output diplays using the sample driver.

The Loudspeaker Design Cookbook by Vance Dickason.

If you want to get into the "nitty-gritty" of loudspeaker design, purchase *Loudspeakers* (Vols. 1 and 2) from the Audio Engineering Society (AES), 60 E. 42nd St., New York, NY 10165; members—$27, nonmembers—$30, each. Both volumes cover over 30 years of design and should be a valuable reference, with articles by famous authors and lots of mathematical equations.

T/S DRIVER PROGRAM. When measuring the T/S parameters, the test setup requires that you use the CONSTANT VOLTAGE test method as described by Bob Bullock,[6] and when determining the V_{as} of the driver, you are limited to using either the SEALED BOX or ADDED MASS method as described by Joe D'Appolito.[7] (Note that if you choose the ADDED MASS method, your frequency counter must be able to resolve down to 0.1Hz for accurate measurements.

Wherever possible, I have tried to use the "standard" symbols that have become the norm in describing driver parameters (Q_e, Q_t, Q_{ts}, and so on), but because the computer sees data differently, I have substituted slightly altered versions to let the program run properly. This in no way affects the accuracy, and in the final output screen, all values revert back to "standard."

The program starts by asking you the driver name, driver resistance, value of the calibration resistor, constant voltage output, driver free-air resonance, and the current output at F_S. Note that you can use the value for the calibration resistor and constant voltage output that your test setup requires.

The program then displays on your screen the value of I1 in amps so you can then proceed to measure F1 and F2 (*Fig. 3*). The program will calculate F's and ask if the value is satisfactory. (F's should be within 1Hz or 2% of F_S) If F's is not satisfactory, recheck your test methods and try again.

The program then goes to the V_{as} determining portion; you choose either the SEALED BOX or ADDED MASS meth-

6. Bullock, R., "Thiele, Small and Vented Loudspeaker Design," *SB* 4/80, 1–3/81, 1/82.
7. D'Appolito, J.A., "More Driver Tests," *SB* 4/82, pp. 41–44.

```
100 PRINT "[control chars]" TAB(9)"[control chars]"
110 PRINT TAB(9)"      T/S DRIVER       "
120 PRINT TAB(9)" PARAMETERS USING      "
130 PRINT TAB(9)"    THE CONSTANT       "
140 PRINT TAB(9)"  VOLTAGE METHOD       "
150 FOR PAUSE = 1 TO 7000:NEXT:PRINT""
160 REM A PROGRAM BY DAVID LONG
170 GOTO190
180 PRINT CHR$(147)
185 REM DRIVER DATA INPUT
190 INPUT "DRIVER ID # ?";D$
200 INPUT "DRIVER D.C.RESISTANCE?";RE
210 INPUT "CALIBRATION RESISTOR DCR?";RC
220 INPUT "CONSTANT VOLTAGE OUTPUT?";VO
230 IC=VO/RC
240 IE=IC*RC/RE
250 INPUT "FREE AIR RESONANCE?";FS
260 INPUT "CURRENT IN AMPS AT FS?";IM
270 Z=IE*RE/IM
275 Z=INT((Z*1000)+.5)/1000
280 RO=IE/IM
290 I1=IE/SQR(RO)
300 PRINT "I1=";I1;"AMPS"
310 INPUT "FREQ. BELOW FS WHERE I=I1 ?(F1)";F1
320 INPUT "FREQ. ABOVE FS WHERE I=I1 ?(F2)";F2
330 FO=SQR(F1*F2)
335 REM DRIVER ONLY,QM,QE,AND QT
340 QM=SQR(RO*F1*F2)/(F2-F1)
350 QE=QM/(RO-1)
360 QT=QM/RO
370 PRINT "F'S=";FO;"HZ"
380 INPUT "IS F'S OK (Y,N)?";K$
390 IF K$="N" THEN GOTO310
400 PRINT "VAS CALCULATIONS"
410 INPUT "1=CLOSED BOX,2=ADDED MASS";AB
420 IF AB=1 THEN GOTO480
430 INPUT "FSAM=?";FM
435 REM FSAM IS RESONANCE OF DRIVER WITH    MASS (GR) ADDED TO CONE
440 INPUT "ADDED MASS IN GRAMS=?";GR
450 MS=GR/1000
460 CMS=((FS+FM)*(FS-FM))/((FS↑2)*(FM↑2))*((1/(39.478418*MS)))
470 GOTO680
475 REM CLOSED BOX VAS MEASURMENTS IN A     BOX OF KNOWN VOLUME(VT)
480 INPUT "BOX RESONANCE (FC)?";FC
490 INPUT "CURRENT IN AMPS AT FC?";IN
500 RP=IE/IN
510 I2=IE/SQR(RP)
520 PRINT "I2=";I2;"AMPS"
530 INPUT "FREQ. BELOW FC WHERE I=I2 ?(FL)";FL
540 INPUT "FREQ. ABOVE FC WHERE I=I2 ?(FH)";FH
550 FD=SQR(FL*FH)
560 FD=INT((FD*100)+.5)/100
570 PRINT "F'C=";FD;"HZ"
580 INPUT "IS F'C OK (Y,N)?";L$
590 IF L$="N" THEN GOTO530
600 QA=SQR(RP*FL*FH)/(FH-FL)
610 QB=QA/(RP-1)
620 INPUT "TEST BOX VOLUME=(FT3)?",VT
630 INPUT "DRIVER RADIUS IN INCHES=";IN
640 SD=(3.1415927*IN↑2)/1550
650 VAS =VT*(((FC*QB)/(FS*QE))-1)
655 VAS=INT((VAS*10000)+.5)/10000
660 CMS=VAS/((1.39E5*SD↑2)*35.31)
670 GOTO710
680 INPUT "DRIVER RADIUS IN INCHES=";IN
690 SD=(3.1415927*IN↑2)/1550
700 VAS=(1.39E5*CMS*SD↑2)*35.31
701 VAS=INT((VAS*10000)+.5)/10000
705 REM COMPUTATION OF DRIVER EFFICIENCY
710 NO=(VAS/QE)*(FS↑3)*2.7E-8
711 NO=INT((NO*1000000)+.5)/1000000
715 REM DRIVER SPL REFERENCED TO 1 METER    AT 1 WATT INPUT
720 SPL=(LOG(NO)/LOG(10))*10+112
725 SPL=INT((SPL*100)+.5)/100
730 RA=SQR(SD/3.1415927)
740 CMS=INT((CMS*1000000)+.5)/1000000
750 INPUT "AMP RESISTANCE =";RG
760 INPUT "XOVER RESISTANCE=";RX
770 MD=1000/((39.47*CMS)*FS↑2)
800 MD=INT((MD*100)+.5)/100
805 REM XOVER AND AMPLIFIER LOSSES
810 DL=20*LOG(RE/(RE+RG+RX))/LOG(10)
820 DL=INT((DL*100)+.5)/100
825 REM DRIVER SPL WITH XOVER AND AMP       LOSSES
830 CA=DL+SPL
835 REM DRIVER QE'S AND QTS(EFFECTS OF      XOVER AND AMPLIFIER)
840 QS=((RG+RX+RE)*QE)/RE
850 QU=(QS*QM)/(QS+QM)
860 FO=INT((FO*100)+.5)/100
870 PRINT CHR$(147)
880 PRINT D$
890 PRINT"  "
900 PRINT "RE =";RE;"OHMS" TAB(21) "RG =",RG;"OHMS"
905 PRINT "FS =";FS;"HZ" TAB(21) "RX =";RX;"OHMS"
910 PRINT "IM =";IM;"AMPS" TAB(21) "Z  =",Z;"OHMS"
920 PRINT "F1 =";F1;"HZ"
930 PRINT "F2 =";F2;"HZ" TAB(21) "MD =";MD;"GR"
940 PRINT "F'S=";FO;"HZ" TAB(21) "CMS=";CMS;"M/N"
950 PRINT "QM =";QM TAB(21) "VAS=";VAS;"FT3"
960 PRINT "QE =";QE TAB(21) "EFF=";NO
970 PRINT "QT =";QT TAB(21) "SPL=";SPL;"DB"
980 PRINT "QES=";QS TAB(21) "LOSSES=";DL;"DB"
990 PRINT "QTS=";QU TAB(21) "SPL=";CA;"DB"
1000 PRINT "  "
1010 INPUT "ANOTHER DRIVER?(Y,N)";B$
1020 IF B$="Y" THEN GOTO180
```

FIGURE 3b: Program listing.

```
71 A=V/VB

72 H=FB/F

90 PRINT "VB=";VB;"FT3""       ""ALPHA=";A

160 PRINT"FB=";FB;"HZ""      ""H=";H
```

FIGURE 4: OPTIMUM BOX added lines.

od. (Please refer to the references at the end of this article for more information.) The rest of the V_{as} portion is self-explanatory.

The final display screen will show, in addition to the usual parameters, the effects of amplifier and crossover losses on the driver (Q_e, Q_t, Q_{es}, Q_{ts} and SPL), driver impedance at F_S, the driver's moving mass (M_{MD}), the driver's suspension compliance (C_{MS}), and efficiency (η).

CLOSED BOX. This program was created to complement OPTIMUM BOX and BOXRESPONSE software from Old Colony Sound Lab, disk SBK-E3CD. Like OPTIMUM BOX, CLOSED BOX will calculate the box size, tuning frequency (F_C), and −3dB down point. One of this program's advantages is that it allows

```
CLOSED BOX VOLUME

ENTER FS(HZ),QTS,QTC,AND VAS(FT3)

? 20.5,.3697,.5,4.6116

OPTIMUM BOX VOLUME
VB= 5.56207249 FT3    ALPHA= .829115408

-3DB DOWN POINT
F3= 43.0786672 HZ

BOX TUNING FREQUENCY
FC= 27.7251826 HZ
AGAIN (Y OR N)?
? N
```

FREQ IN HZ	RELATIVE RESPONSE IN (DB)	MAXIMUM POWER INPUT IN WATTS	MAXIMUM INFINITE BAFFLE RESPONSE IN (DB)
*****	********	*******	********
5	-30.03	2.25	59.59
10	-18.78	2.7	71.63
15	-12.9	3.53	78.68
20	-9.31	4.89	83.68
25	-6.96	6.95	87.55
30	-5.36	9.96	90.72
35	-4.23	14.22	93.4
40	-3.41	20.07	95.72
45	-2.79	27.92	97.76
50	-2.33	38.22	99.59
55	-1.97	51.49	101.25
60	-1.68	68.28	102.76
70	-1.26	100	104.83
80	-.98	100	105.11
90	-.79	100	105.31
100	-.64	100	105.45
150	-.29	100	105.81
200	-.17	100	105.93

FIGURE 5: CLOSED BOX display with BOXRESPONSE output.

MANUFACTURERS' SPECIFICATIONS

Driver	10" diameter (8" nominal)
R_e	4Ω
F_S	22Hz
X_{max}	0.125" peak
power	100W

you to find the −3dB point, regardless of the Q_{tc} of the enclosure; previously I had to look up the −3dB point on a chart, which had no provision for in-between Q_{tc} values. Do not try to input an impossible alignment (Q_{tc} smaller than the driver Q_{ts}).

Unlike OPTIMUM BOX, this program will output the compliance ratio (A) so that you will have the data on hand when you run the alignment through BOXRESPONSE.

Similarly, I have added more data lines to OPTIMUM BOX so that it will output the compliance ratio (A) and the tuning ratio (H). This makes data transfer to BOXRESPONSE easier. Note that OPTIMUM BOX assumes the box loss factor (Q_L) is 7. If your system measures differently, then you may prefer to use the design tables by Bob Bullock.[8]

If you already own BOXRESPONSE, which includes OPTIMUM BOX, then add only the part of the lines I have included (*Fig. 4*) that are underlined.

Table 1 lists a sample driver used to demonstrate T/S DRIVER PROGRAM. Also shown are the input and output screens of the program (*Fig. 3*), and the screens of CLOSED BOX, OPTIMUM BOX (the revised version), and the output of BOXRESPONSE, all using the sample driver (*Figs. 5* and *6*).

In the example for CLOSED BOX, the Q_{tc} is critically damped (.5) and actually requires a larger box than when the driver is placed in a vented box ($Q_L = 7$).

BOXRESPONSE confirms that the alignments above are in close agreement. Note that BOXRESPONSE[9] does not take into account crossover and amplifier losses so that the last column (maximum infinite baffle response in decibels) will

```
10 PRINT "       CLOSED BOX VOLUME "
20 PRINT "ENTER FS(HZ),QTS,QTC,AND VAS(FT3)
30 PRINT
40 INPUT F,Q,QT,V
50 PRINT
60 A=((QT/Q)↑2)-1
70 FC=F*SQR(A+1)
80 F3=FC*SQR(((1/QT↑2-2)+SQR((1/QT↑2-2)↑2+4))/2)
85 VB=V/A
90 PRINT "OPTIMUM BOX VOLUME"
100 PRINT "VB=";VB;"FT3""   ""ALPHA=";A
110 PRINT
115 PRINT
120 PRINT "-3DB DOWN POINT"
140 PRINT "F3=";F3;"HZ"
150 PRINT
160 PRINT
170 PRINT "BOX TUNING FREQUENCY"
180 PRINT "FC=";FC;"HZ"
190 PRINT "AGAIN (Y OR N)?"
200 INPUT C$:IF C$="Y" GOTO 10
```

FIGURE 5b: Program listing.

```
OPTIMUM BOX VOLUME

ENTER FS(HZ), QTS, AND VAS (CU FT)

? 20.5,.3697,4.6116

OPTIMUM BOX VOLUME
VB= 3.97805527 FT3    ALPHA= 1.15925991

3DB DOWN POINT

F3= 21.4653502 HZ

BOX TUNING FREQUENCY
FB= 21.0833028 HZ     H= 1.02845379
```

FREQ IN HZ	RELATIVE RESPONSE IN (DB)	MAXIMUM POWER INPUT IN WATTS	MAXIMUM INFINITE BAFFLE RESPONSE IN (DB)
*****	********	*******	********
5	-49.8	.76	35.09
10	-26.26	1.26	60.86
15	-12.68	3.39	78.71
20	-4.4	45.03	98.23
25	-1.36	19.53	97.64
30	-.74	12.92	96.47
35	-.61	14.93	97.23
40	-.54	19.66	98.5
45	-.48	26.71	99.88
50	-.43	36.31	101.27
55	-.38	48.91	102.61
60	-.34	65.03	103.89
70	-.27	100	105.83
80	-.22	100	105.88
90	-.18	100	105.92
100	-.15	100	105.95
150	-.07	100	106.03
200	-.04	100	106.06

```
VENT DIAMETER (IN.)? 3

VENT AIR SPEED MACH NO.   .0488
VENT TUBE LENGTH (IN.)    8.62
```

FIGURE 6: OPTIMUM BOX display with BOXRESPONSE output.

not be correct unless you subtract the "losses=" value in the T/S DRIVER PROGRAM from that column. For example, in the T/S DRIVER PROGRAM output screen, LOSSES=0.256dB should be subtracted from every number listed in the last column of BOXRESPONSE.

Also notice that in the example for T/S DRIVER PROGRAM, I1=8.6602540E−03, is the same as 0.00866025404 (E−03 means you must move the decimal point three places to the left). The same applies to EFF and IM in the output screen.

CONCLUSION. I hope you find these programs useful in your search for the "perfect" loudspeaker low-frequency alignment.

The programs this article describes are available on a Commodore 64-format disk, 5¼", SS/DD; available through Old Colony Sound, PO Box 243, Peterborough, NH 03458, (603) 924-6371; FAX: (603) 924-9467; price is \$25. This disk includes programs for Q_{LA}, Q_{TC} and room resonance.

8. Bullock, R., "Thiele, Small and Vented Loudspeaker Design," *SB* 4/80, 1–3/81, 1/82.

9. Bullock, R. and Bob White, "Box-response," *SB* 1/84, pp. 13–18.

THE CORDLESS REVOLUTION

BY BRUCE EDGAR
Contributing Editor

Probably the one tool I could not get along without is my cordless screwdriver. Three years ago a neighbor lent me his and told me, "be careful, it's addictive." He was right! I've used it to literally replace my hammer and screwdriver for most projects. My collection of cordless screwdrivers has now grown to three.

BACKGROUND. If you purchased a cordless drill 5–10 years ago, you were probably disappointed by its wimpy performance and long recharging time. However, over the years, battery recharging time has dropped from 16 hours to one hour. The driving power has also improved with the higher capacity NiCad batteries.

So what is the effect on the individual speaker builder? The cordless screwdriver allows you to build boxes with tight butt joints, that won't pull apart as do enclosures built using nails. Construction time for prototype boxes is reduced. Jigs are easily constructed. I built most of my bass horn projects with screws only and without glue, except for the back chamber.

CHOOSING A CORDLESS. Occasionally you see the $20-$30 cordless screwdriver on special at the local hardware store or on television. Don't waste your time on one. They simply don't have the power to drive screws into wood. They are great for screwing machine screws into threaded holes such as found in electronic equipment. The smallest tool in *Photo 1* is a good example of this class.

The price threshold for a cordless screwdriver suitable for speaker construction seems to be about $100. The middle screwdriver in *Photo 1* is a good example. It is made by Skil, but marketed by Sears in their professional tool line. The Sears Professional screwdriver served me well for three years but developed a problem of gear jump which eventually caused failure of the low-speed gear. Because Sears had discontinued that model, I had to replace it. I eventually settled on the top of the line Skil "Top Gun" cordless screwdriver which I was able to buy for $150 at a tool discount store. Recently I saw this model on sale for a little over $100.

Another feature of the pro class of cordless screwdrivers is a Jacobs chuck and a high speed gear for drilling. However, I would not consider the cordless drill/screwdriver as a true dual tool because it takes time to change bits, unless you have an interchangeable bit system. When my Sears machine stripped its low speed gear, it became my cordless drill since its high speed gear was still functioning. Now I can drill and screw without having bothersome electrical cords all over my floor and project. My construction efficiency is greatly improved.

The Skil Top Gun, as well as many high quality models, comes with two batteries. You might think the extra battery is a luxury, but the NiCad batteries will give warning of a needed recharging only a few screws before the battery is completely drained. If your reserve battery is ready, a quick interchange will allow you to continue working. With the Sears machine, I only had one battery which seemed to run out at the most inopportune times. So if your choice of cordless screwdriver has only one battery, order another one for efficient operation.

Makita's machines, such as their model #6093DW, are comparable to the Skil models. Again, you should consider spending more than $100 to ensure obtaining a cordless machine rugged enough for furniture or speaker building.

SCREWS. The wide availability of the wallboard screw has propelled the cordless revolution (*Photo 2*, the one with

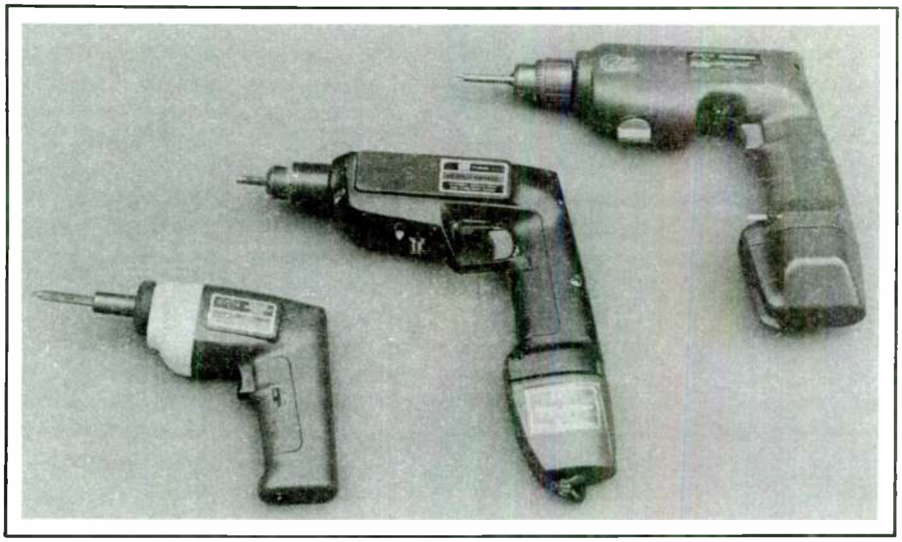


PHOTO 1: The author's collection of cordless screwdrivers.

small threads). This screw is designed to penetrate a wall board and screw into a 2 by 4 stud (wood or metal) without drilling a hole. The wallboard screw is heat treated for hardness and has a black phosphate coating. They come in a variety of lengths, from 1–3 inches. I use the 1-inch screw for attaching drivers and jig building. The 2-inch screw can be used for fast butt joining of underlayment particle board without drilling holes.

For serious horn building I use a hardened furniture screw or "grippit" (*Photo 2*) which has larger blades on the shank. When screwing plywood or particle board together, wallboard screws will strip their threads if you are not careful, but the grippit type will clinch the joint almost airtight.

TIPS. While the versatility of the cordless screwdriver is impressive, a few warnings should be noted. If you don't predrill your holes, you will encounter some splitting. I always predrill for the furniture screws. But you should experiment with your favorite materials.

The bugle head Phillips-head screw seems to be the industry standard, but the square-drive screw head offers advantages in power transmission to the screw. If your screwdriver tends to jump out of the Phillips-head groove, you must learn to hold the screwdriver directly in line with the screw shank while applying pressure.

Another helpful tip is to have a variety of different screwdriver lengths to poke into those hard to reach areas. Also, see if your hardware store carries hardened bits, which tend to stay better in Phillips-head slots and won't wear as badly as unhardened ones.

CONCLUSIONS. The cordless revolution has changed my way of speaker building, and I'm sure it will change yours after you try a cordless tool. A recent *Los Angeles Times* article mentioned the television and movie industries use cordless drills and screwdrivers almost exclusively in building and tearing down sets. Similarly, I now reuse the plywood from my prototype enclosures by just unscrewing them, rather than sawing them up for scrap.

The cordless power tool market has reached up to $200 million and promises other useful tools. Hmmm—that cordless saw looks inviting.

ADDENDUM. As I was finishing this article, magazine surveys of cordless screwdriver/drills appeared in *Consumer Reports*, (August 1988, p. 522) and *Fine Woodworking* (Sept.–Oct. 1988, p. 52). The *Consumer Reports'* survey recommended the top-of-the-line Skil and Makita models I noted in this article. *Fine Woodworking* offered pros and cons on various professional models and gave test results for the number of holes drilled/screws driven per battery charge. Most models tested averaged 100 holes drilled or screws driven. Both articles mentioned a budget model, the Sears #11132, which gave very good performance (the highest rank in the drilling/screwing test) for a lower price ($70 list, $50 on sale, with one battery).

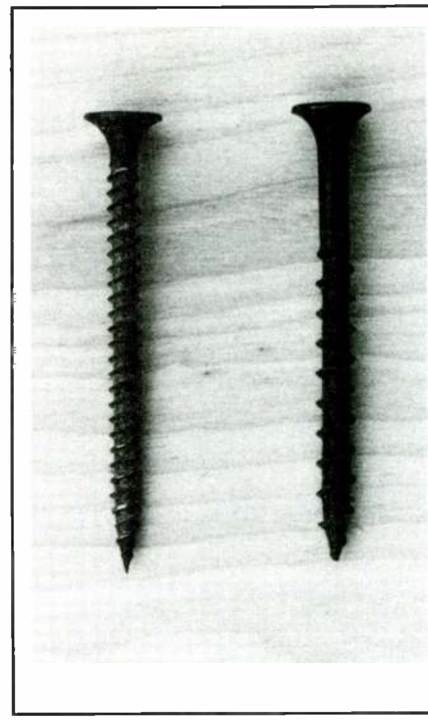


PHOTO 2: Left, the hardened wallboard screw; right, the hardened furniture or grippit screw.

SOURCES

Trend-lines
375 Beacham St.
Chelsea, MA 02150

Equality Screw Co.
PO Box 1645
El Cajon, CA 92022

(The hardened furniture screw that is typically not found in hardware stores. Trend-lines offers small quantities of screws for reasonable prices. Equality caters more to the buyer interested in orders greater than $35. Smaller orders have a $5 service charge tacked on. However, I use the Equality screws almost exclusively and have found them to be very high, commercial grade quality.)

THE THIRD DIMENSION:

THE SYMMETRICALLY LOADED DELTA

BY JEAN MARGERAND

Briefly, the symmetrically-loaded driver principle involves placing a woofer inside an enclosure, one face loaded by a closed box, the other by a vented box. The sound is emitted through the vent only. The frequency response is identical to that of a pass-band system. For a given driver, the reproduced range of frequencies and the sensitivity of the system depend on the two cavities' volumes and the vent-tuning frequency. For more details refer to the first part of my article (*SB* 6/88).

SPECIFICATIONS. My aim was to design an aesthetic shape and I wanted reasonable volume, to reproduce the bass frequencies at significant sound levels with the greatest impact possible.

For the bass section, my favorite speaker retailer advised me to choose a Polydax HIF12EB, 12cm (4.8-inch) diameter driver. Its major qualities are a low F_S/Q_T ratio of 130Hz which involves a possible low cutoff frequency, a high efficiency level (92dB) and an excellent accelerating factor, Γ, of 1,500. (Γ is the ratio of BL by M_{ms}, the moving mass). The greater the Γ, the better the transient response. For 12cm-diameter drivers, the Γ varies from 800 to 1500. Also, this driver is inexpensive, $20 each.

For the midrange driver, I selected the "standard" model Fostex FE103 (the "sigma" model is designed for better bass reproduction, which is not our concern for this driver in my design). This driver is characterized by its extraordinary resolution capability (see *Fig. 8*).

For the high-end reproduction, I use a well-known unit, the Polydax TW51A.

For the low frequencies, I chose the symmetrical-loading principle, which will draw the best performance from this small diameter bass driver, with greater power handling than a vented box. The Polydax HIF12EB manufacturer's specifications are:

F_S, 56Hz;
Q_T, .43;
V_{as}, 15.9 liters;
SPL, 92dB W/M.

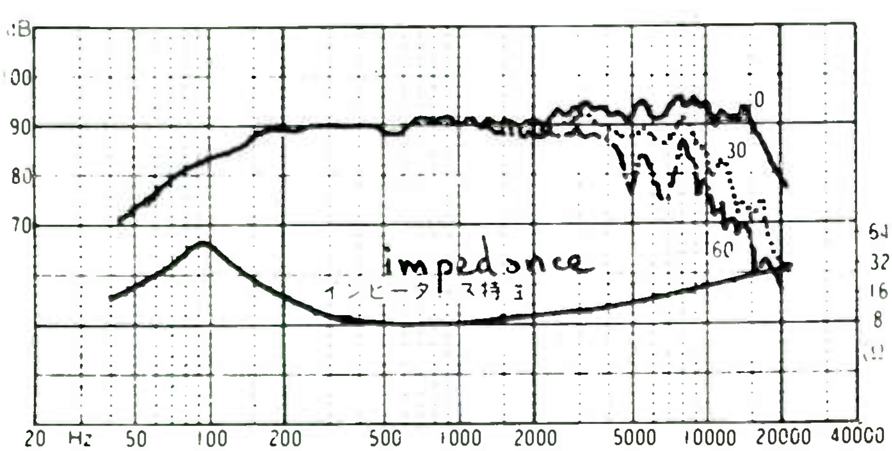


FIGURE 8: Midrange specs for the Fostex FE103: F_S, 80Hz; Q_{TS}, 0.35; V_{as}, 5.2l; SPL, 89dB.

For the bass section design, I follow the procedure of Philippe Augris and Dominique Santens, the authors of the symmetrically-loaded system theory.

A. for the choice for the value of S, I decided to push the frequency response toward the low range as much as I could, and accept a 1.25dB frequency response modulation. I chose a value of S = 0.5.

B. To calculate the front compartment volume:

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

$(2 \times 0.5 \times 0.43)^2 \times 15.9$; so $V_F = 2.94$ liters.

C. For the choice of the −3dB low cutoff frequency F_L, I chose 37.5Hz. For:

$$F_S/Q_T = 56/.43 = 130.23\text{Hz};$$

Then,

$$F_L / (F_S/Q_T) = 0.2879$$

D. For the value of Q'_T (see *Table 3, Part I*), we find:

$$F_L/(F_S/Q_T) = .2867,\ .2879,\ .3010;$$

and

$$Q'_T = .6683,\ x,\ \text{and}\ .6879.$$

You interpolate for x: $Q'_T = 0.6699$.

E. For the calculation of the −3dB high cutoff frequency, F_H;

where S = 0.5, and (bandwidth):
$$\Delta F/(F_S/Q_T) = 1.27115;$$

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

$$0.2879 + 1.27115 = 1.55905;$$

so

$$F_H = 1.55905 \times F_S/Q_T = 203.4\text{Hz}$$

F. To determine the back and total volumes, V_B and V_T:

$$V_B = \frac{V_{as}}{\left(\frac{Q'_T}{Q_T}\right)^2 - 1} = 11.13 \text{ liters}$$

$V_T = V_F + V_B = 2.94 + 11.13 = 14.07$ liters.

G. Tuning frequency of the vent:

$$F_0 = Q'_T \times F_S/Q_T = 87.24\text{Hz}.$$

H. Vent dimensions—for the chosen diameter:

$$D_V = 5\text{cm};$$
$$S_V = 19.63\text{cm}^2$$

the tuning frequency

$$F_0 = 87.26\text{Hz}$$

the front volume

$$V_F = 2.94 \text{ liters}$$

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

so $L_V = 22.3$cm.

I. Sound Pressure level (SPL):

for

$Q'_T = .6683, .6699, .6879;$

and

SPL(dB) = −7.0, y, and −6.5;

the interpolation gives you:

$$y = -6.96$$

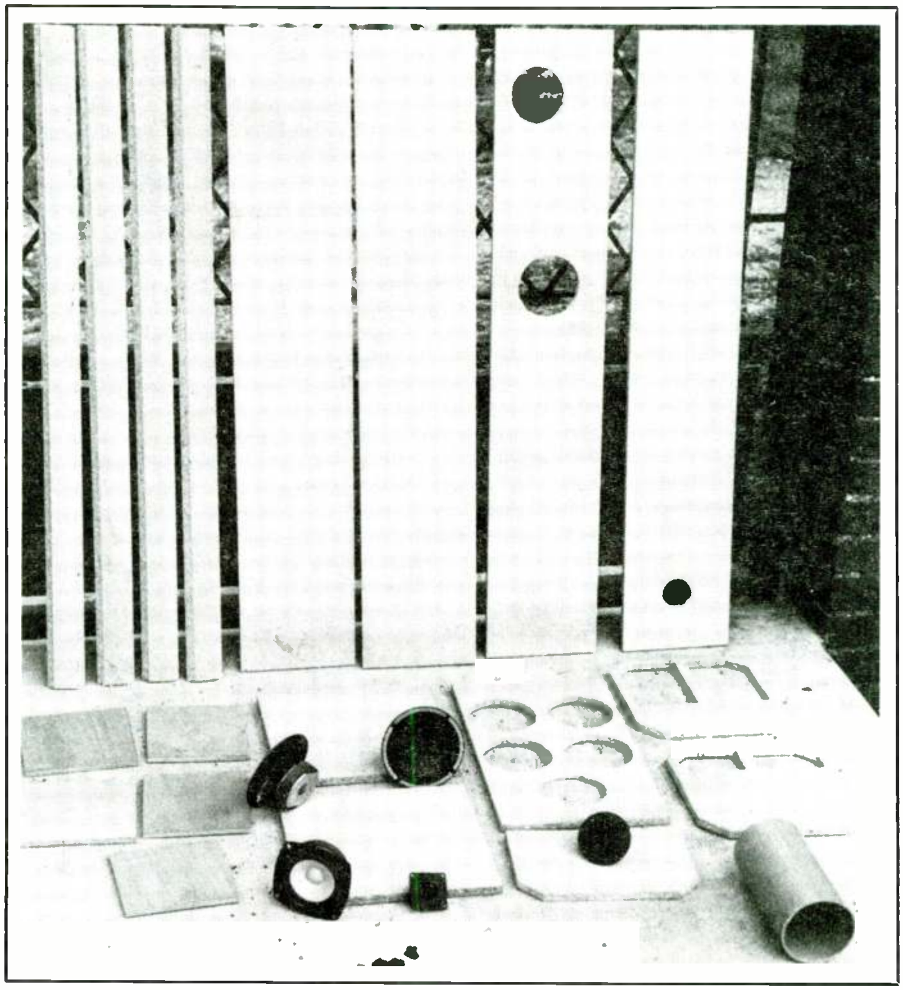


PHOTO 1: The essential components.

The SPL is 6.96dB lower than that of a conventional enclosure (closed or vented).

Obviously, the pressure level of just a single 4.7-inch diameter driver is limited in the 30–40Hz range. To keep the transient quality of small drivers and increase pressure output, I decided to use four 4.7-inch diameter drivers. Thus, I obtain a 10-inch diameter driver equivalent surface. And yet, unlike one disadvantage of a multi-driver system, the output in the lower range forms its coherency inside the enclosure. For the listener, the bass emits from a single source—virtual bass cone of the vent. Furthermore, in a push-pull configuration, by reversing two drivers (two with coil up, two with coil down), the nonlinear distortion is diminished.

As *Table V* shows, using four drivers instead of one involves no change for the values of S, F_L, F_H, Q'_T, F_0, or L_V. However, the vent area and all the volumes are multiplied by four, so: $V_F = 11.76$ liters, $V_B = 44.56$ liters, $V_T = 56.32$ liters, and $S_V = 78.52$ cm^2.

The alignment calculations predict a −6.96dB attenuation for one driver. The four series parallel connected bass drivers bring a +6dB power or voltage sensitivity. The final result is a −0.96dB attenuation. An interesting result is a +12dB increase for the maximum acoustic output when you use four woofers instead of one (*Table V*).

TABLE V

DELTA SYSTEM ALIGNMENT

	HIF12EB (1)	HIF12EB (4) series parallel con.
F_s (Hz)	56	56
Q_T	.43	.43
V_{AS} (liters)	15.9	63.6
S	.5	.5
Q'_T	.6699	.6699
F_0 (Hz)	87.26	87.26
F_L (Hz)	37.5	37.5
F_H (Hz)	203.04	203.04
SPL (dB)	−6.96	−.96
V_F (liters)	2.94	11.76
V_B (liters)	11.13	44.52
V_T (liters)	14.07	56.28
D_v (cm)	5	10
L_v (cm)	22.3	22.3
Max. output	PAR	PAR+12dB

Drivers Connections	1 Driver	2 Drivers in Parallel	2 Drivers in Series	4 Drivers in Series Parallel
Power Sensitivity dB/1W/1M	$S_o = 92$	$S_o + 3$	$S_o + 3$	$S_o + 6$
Voltage Sensitivity 2.83V input (dB)	$S_o = 92$	$S_o + 6$	$S_o + 0$	$S_o + 6$

DESIGN. To incorporate my desired acoustic result and aesthetic design, I chose a slim, elegant octagonal column (*Fig. 9*). The outer dimensions are 28 by 28cm and 105cm high, with beveled corners to improve the diffraction factor and aesthetics.

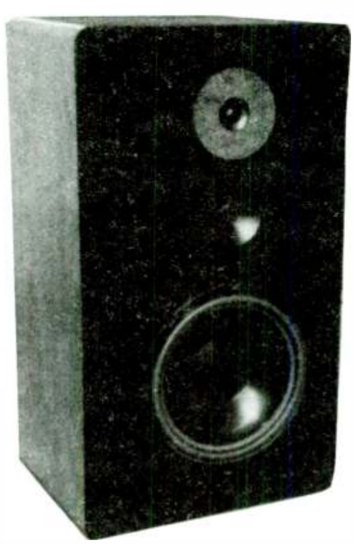
On the Delta's front baffle I place the midrange at the top; the tweeter at ear height (sitting); and the bass vent at 60cm, to avoid ground reflections. The

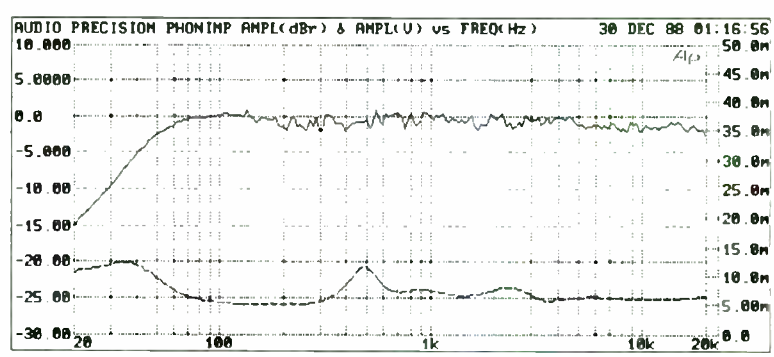
Madisound Presents

The Phoenix

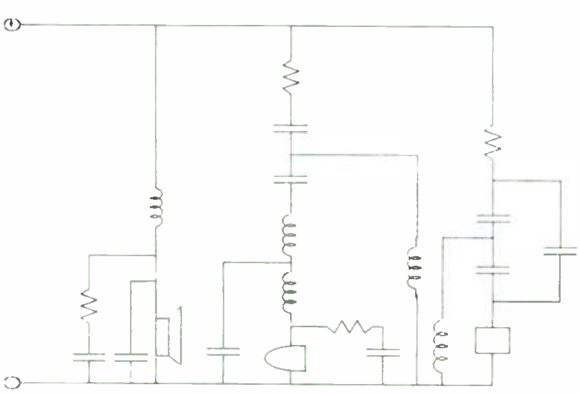
Madisound's *Phoenix* is a flawless 3-way speaker system. A 3-way requires four elements: deep bass, clear midrange, precise high frequencies, and a seamless crossover. We started out with the Madisound 10208 woofer. This is a proprietary polypropylene woofer optimized for low bass in a moderate sized enclosure. The *Phoenix* has an f_3 of 36hz, with a Qtc of .65 (adjustable from .5 to 1.0). Using this kind of woofer gives a very tight, bass with no boominess. We chose the Peerless 1615 (TO105F) midrange for its efficency and clarity. The Vifa D20TD05 ferrofluid tweeter is exceptionally smooth and clear without losing its high end extension. The crossover is the key to this system. Tom Roberts, our chief engineer, spared no expense on this design. By a series of listening tests, precise measurement with our new Audio Precision test equipment and adhering to an uncompromising design philosphy, he has come up with a filter design that is frequency, phase, and impedance correct. This kit is easy to assemble. The cabinet is finished and pre-cut for the drivers. The crossover is assembled, and the grill cloth is stretched on the frame which attaches easily with snap-on headlocks provided. The assembly does require soldering ability.

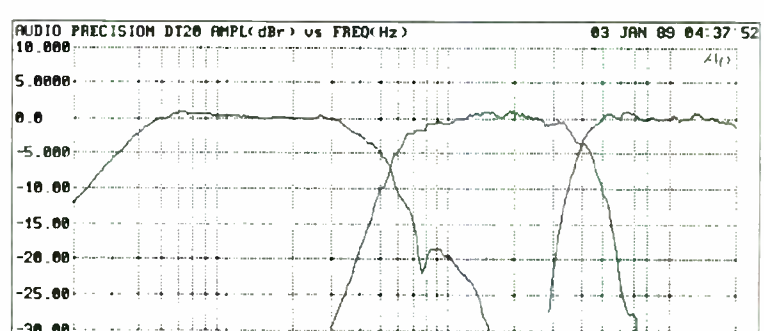
Impedance	8 Ohms nominal	Cabinet	Edges Solid Oak Quarter Round
Sensitivity	88 db		Oak Wood Veneer, Clear Stain Finish
Power Required	40 to 200 Watts		14.5" x 25" x 12"
Frequency Range	35 - 20K Hz +/- 2db		
Crossover	18 db, impedance compensated, min. imp. 4.6 ohms at 190 hz ChateauRoux polypropylene capacitors on tweeter section		
Price	$600 /pr		





Anechoic Response 1W/1M & Impedance Curve


Proprietary 18 db Crossover



Near field Response of Individual Drivers

Fast Reply #ED20

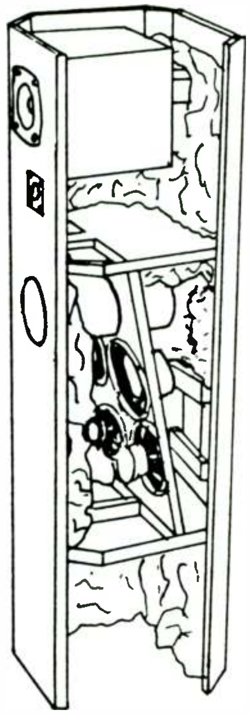


FIGURE 9: General view of the Delta.

right enclosure is a mirror image of the left one. In the back panel, a removable trap door (P) leaves access to the woofers and the crossover.

Three partitions (H, I and J) inside the enclosure create, with the back panel, a 11.76-liter (V_F) compartment (*Fig. 10*), which communicates with the outside through a vent. I mount the four woofers and the vent on the angled partition (I). The rest of the inner volume, except the separate midrange enclosure, represents the closed volume, V_B (44.52 liters). You may notice that the "front" volume, V_F, as mentioned in the theory, is located in back. This particular construction is due to the required vent length of 22.3cm. Finally, the woofers' mounting panel is not parallel to any enclosure wall to avoid standing waves.

CROSSOVER. I believe simple is beautiful, regarding crossover components. For the acoustic cutoff frequency, a 12dB/octave slope on each side of the 203.04Hz bass-midrange, I use, on one hand, the principle of the symmetrical loading, and on the other hand, a closed-box midrange enclosure. The required midrange enclosure is only .48 liters, and yields a 1.21 Q_{TC} and a 1.33dB ripple.

To be on the safe side, I design a much larger midrange enclosure with a 3.77-liter internal volume, to give me full freedom for future modifications. I place polystyrene blocks (3-liter equivalent volume) in the enclosure and, subtracting the 0.1-liter volume occupied by the

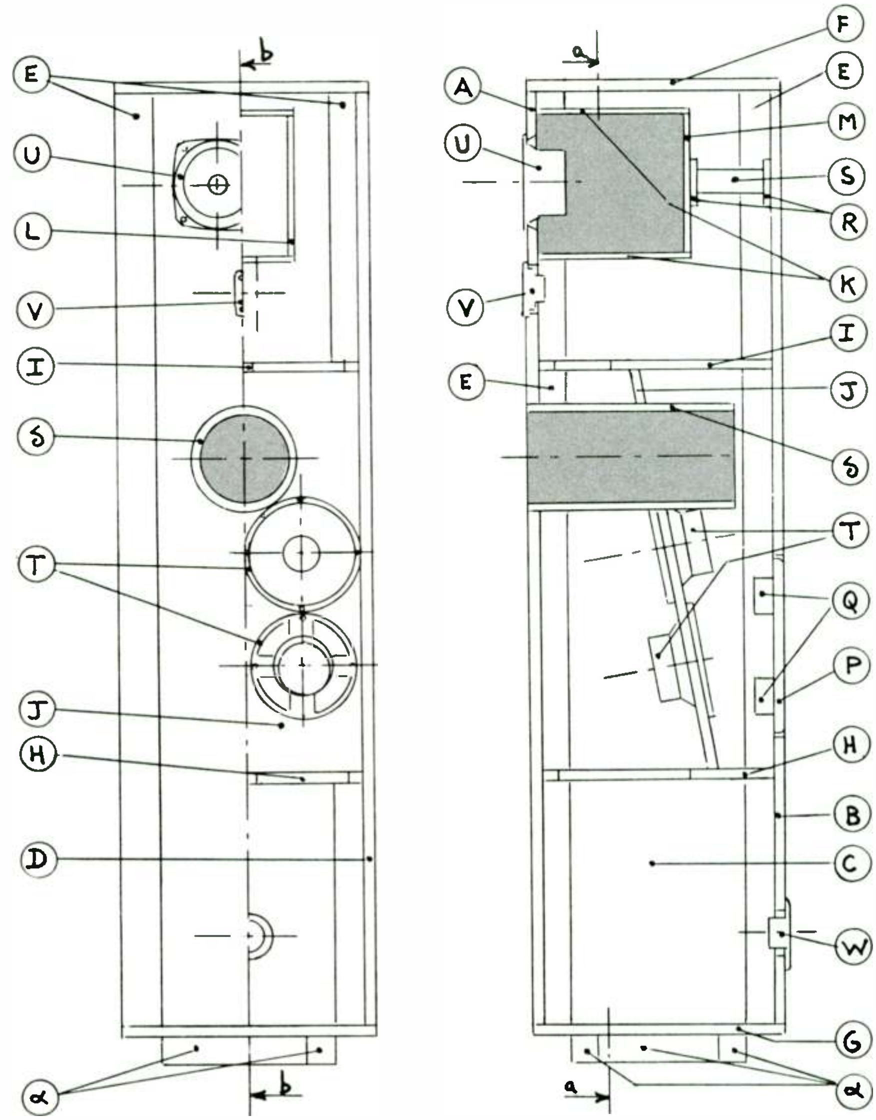
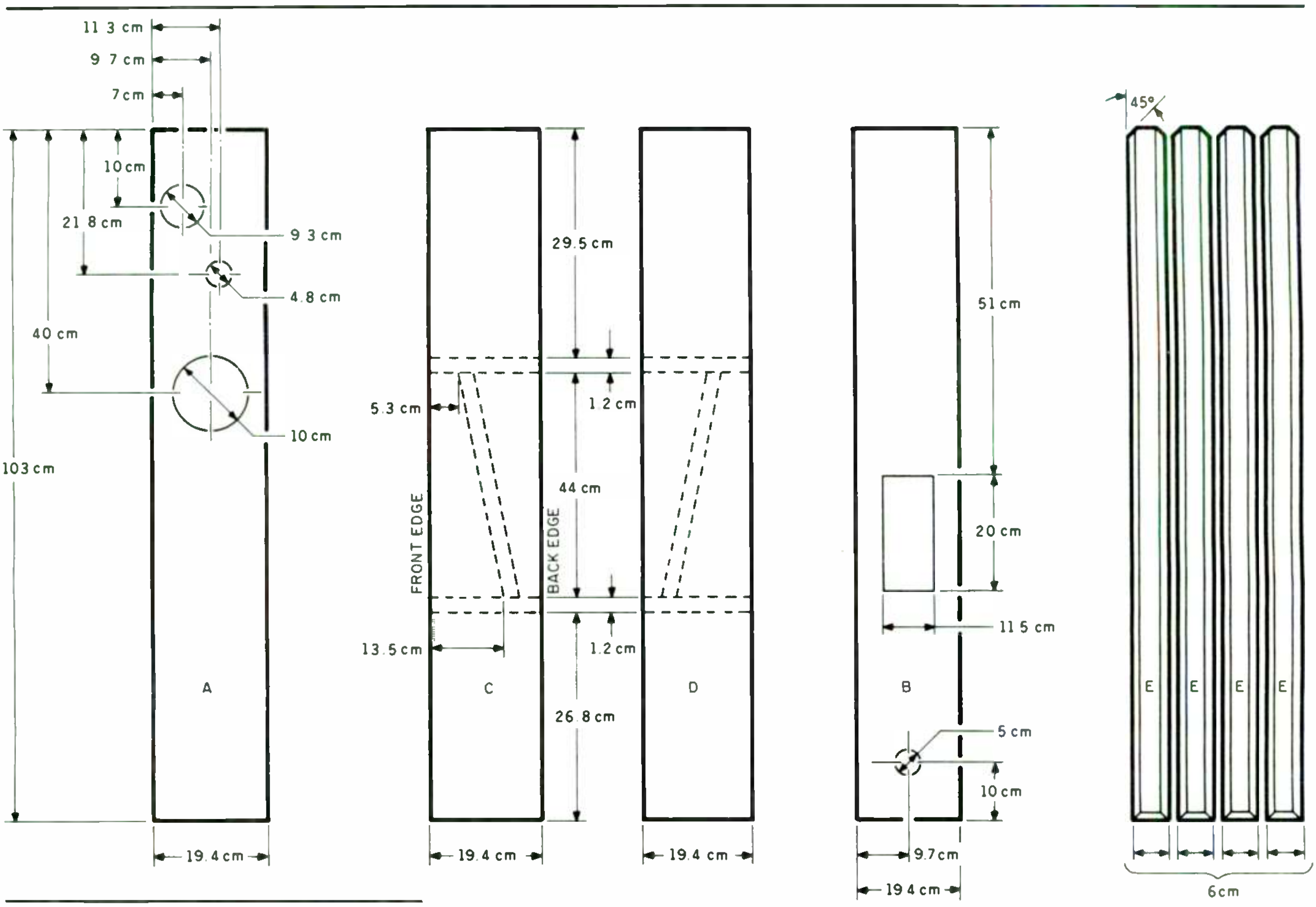


FIGURE 10: Delta system—front and side view.

PARTS LIST FOR A PAIR OF DELTA

	DESIGNATION	DIMENSIONS (in cm)	QUANTITY
• Particle Board			
A B C D	Wide panels	103.0 x 19.4 x 1.23	8
E	Narrow panels	103.0 x 6.0 x 1.23	8
F G	Top and bottom panels	28.0 x 28.0 x 1.23	4
H I	Horizontal partitions	25.6 x 25.6 x 1.23	4
J	Angled partition	44.8 x 25.6 x 1.23	2
P	Trap door	20.0 x 11.0 x 1.23	2
K	Midrange driver enclosure walls	16.7 x 16.7 x 0.8	4
L	"	16.7 x 15.4 x 0.8	4
M	"	15.4 x 15.4 x 0.8	2
• Hardwood			
α	Enclosure base	19.0 x 3.0 x 3.0	4
Q	Trap door brace	16.5 x 4.0 x 2.0	4
R	Brace plate	8.0 x 4.5 x 0.6	4
S	Brace	7.5 x 2.5 x 2.5	2
O	Triangular section molding	107.0 x 1.8 x 1.8	4
N	Square section molding	16.4 x 1.8 x 1.8	8
• Wild Cherry Veneer Cabinet Quality			
	sheets	280 x 30 x 0.3	3



midrange basket, I arrive at the necessary volume.

I connect the bass drivers in series parallel to maintain a final 8Ω impedance. I used a well-known low-cost tweeter, the Polydax TW51A, but you can use a more sophisticated driver, such as the Focal T120 or KEF T27.

For the midrange and tweeter separation I have chosen a simple 6dB/octave slope crossover network because of the excellent linearity of the two drivers. The cutoff frequency is 7kHz. For best results, I compensate for the midrange and tweeter impedance rise around 7kHz. The midrange impedance varies from 5.9Ω to 6.1Ω in the 5kHz- 20kHz range; and for the tweeter, I obtained a variation from 4.0Ω to 4.3Ω in the 2kHz-15kHz range.[2,3,5]

CONSTRUCTION. First, cut the wood pieces as shown in *Figs. 10–12*. For the narrow side pieces (E), you must cut the edges for the 45° angle. Then, you must cut partitions (H, I and J) as precisely as possible.

If you do this initial work carefully, what follows will be very easy. For the top and bottom pieces (F, G) do not break the corners yet. Drill the holes in

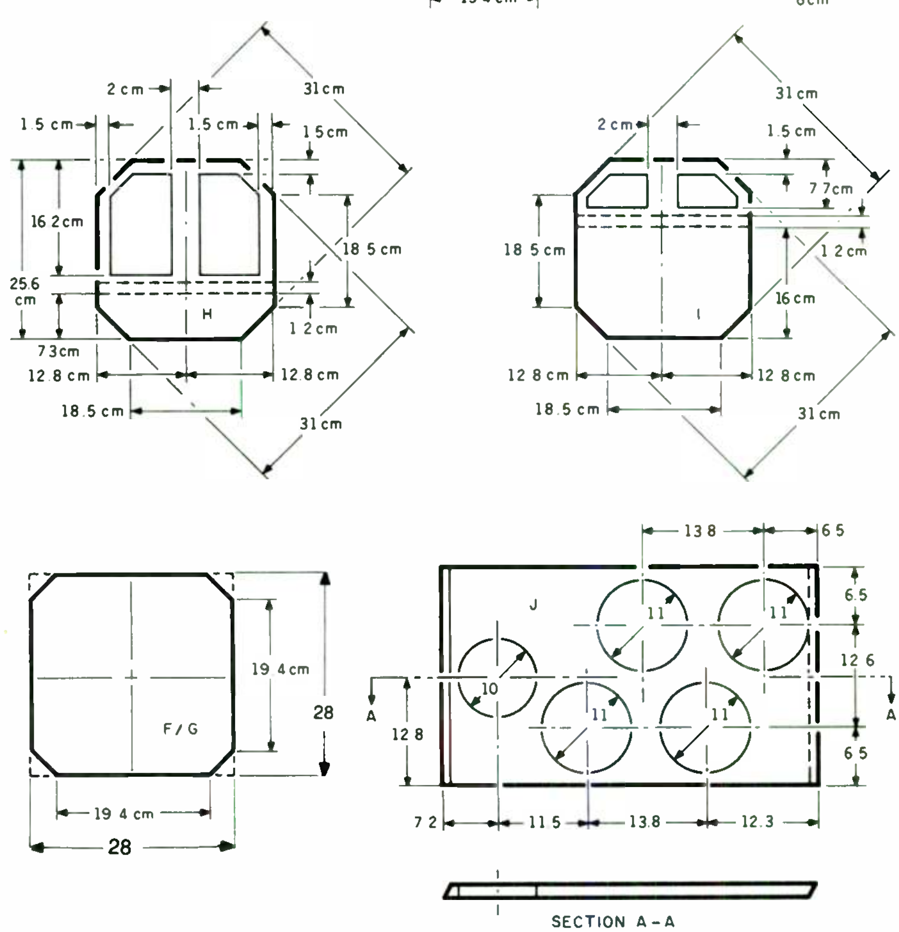


FIGURE 11: Construction guide.

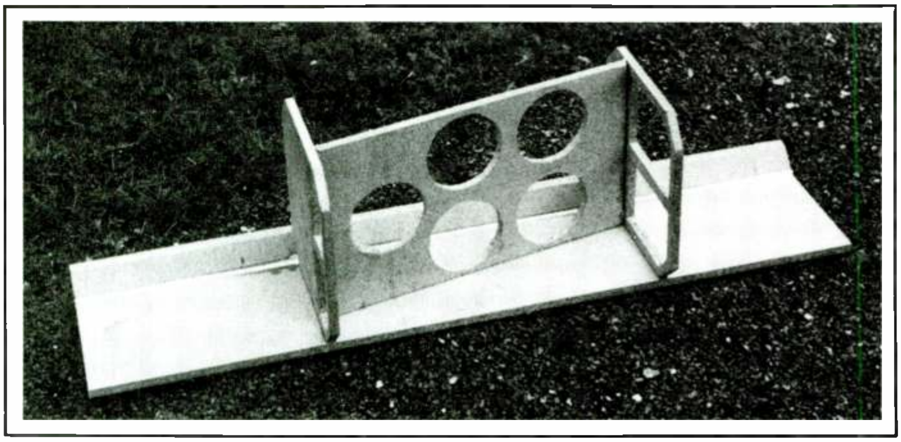


PHOTO 2: Initial assembly—the "heart of the system."

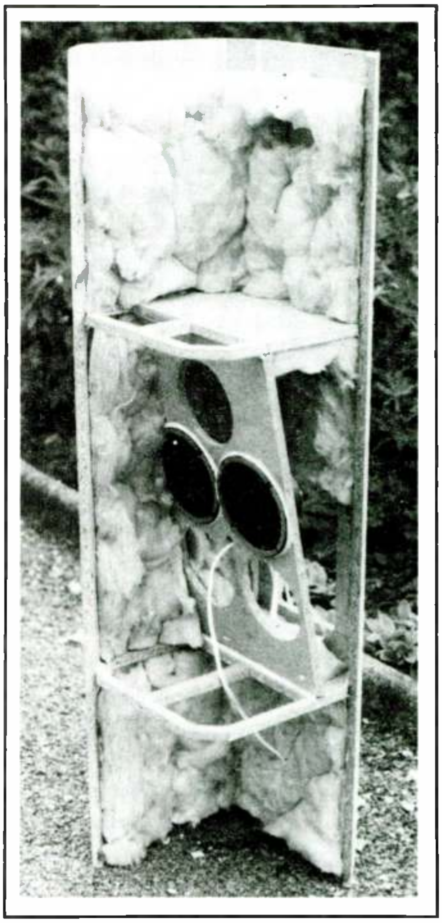


PHOTO 3: The partitions with the cutouts for the four woofers and vent.

the front face (A) for the midrange (9.3cm), tweeter (4.7cm) and the vent (10cm). Remember the left enclosure is a mirror image of the right one. On partition J, drill four holes for the bass drivers (11cm) and one for the vent (10cm). In the back panel, cut the opening for the rectangular trap door and for the circular, push type speaker wire terminals. Of course, you must adapt the aperture dimensions if you modify the speakers or the terminal types. Chamfer the inner edges of the drivers' holes.

The first assembly is this construction's key element, so I suggest you follow my recommendations exactly. Lay the right side panel (C) on your work table and draw on the panel the locations of partitions H, I and J. On panels H and I, draw the partition (J) edge location.

Chamfer the edges of panel J (small sides), for the correct angles between this panel and partitions H and I. Assemble, temporarily without gluing, partitions H, I and J and position this on panel C at the places noted by the pencil drawing. Hold this preassembly with small nailed wood blocks.

Once all is stable and solid enough I suggest you try to preassemble (without gluing) the sides, and the front and back panels to check whether they fit well together. First cut two squares 10 by 10cm, 3-4cm thick. Cut each square to form triangles with 45° angles. These devices will be used as follows. On each side of panel C, place panel E, and support these pieces with the triangular blocks. Then position the front and back panels, the two remaining panels (E) and finally the other side panel (D). Now, if you discover any gaps, file a side of pieces H and I. On the other hand, if there is not enough room, file an edge of a panel (D, for example).

Disassemble the construction. Now glue and nail pieces C, H, I and J, for final assembly of the heart of your enclosure (*Photo 2*). Apply a silicone bead on all inside joints to ensure the enclosure is airtight.

Solder a 1.5m speaker wire to each of the four bass drivers, the midrange and the tweeter, and to the flush mount, push-type speaker wire terminals. Mark the positive wires with red masking tape. Mount the four bass drivers on the partition (J) with screws and a large amount of silicone around the baskets. Install two drivers with the coil facing the front, and two with the coil facing the rear.

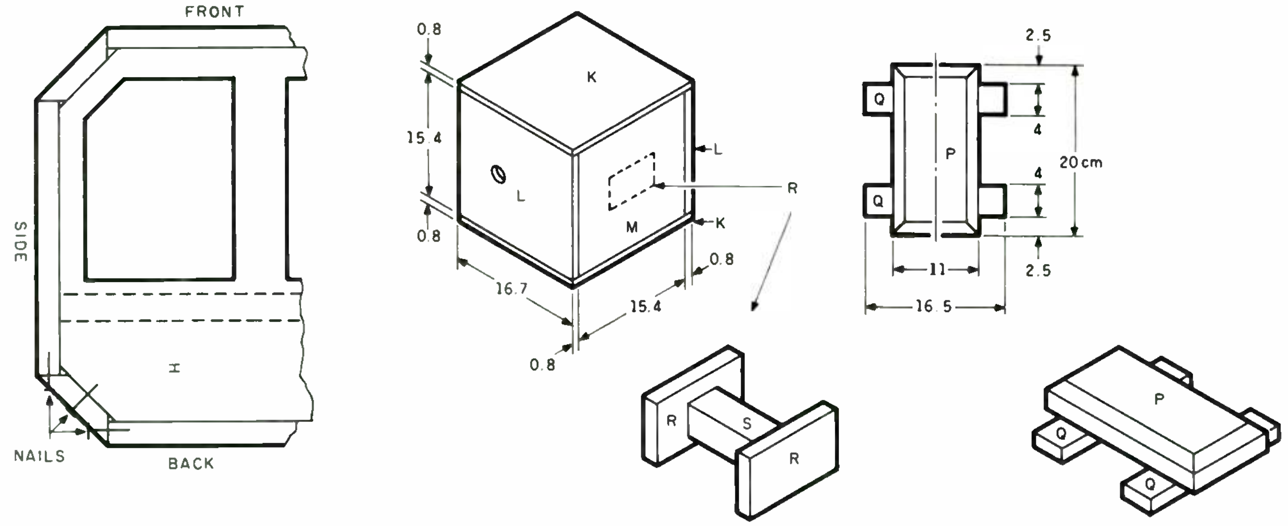


FIGURE 12: Assembly details.

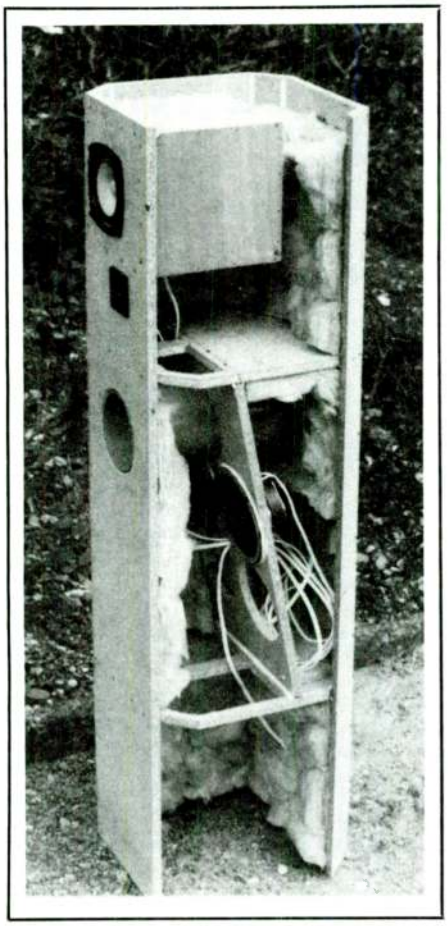


PHOTO 4: The internal configuration showing the midrange enclosure and vent. Initially, the author installed fiberglass in the front volume section, but later obtained better results by removing it.

Drill a small hole (1.5cm) in the middle space between the drivers to feed the wires through. Glue and nail two narrow panels (E) on each side of panel C. The angle between the surface of panel E and the nail is 45° (*Fig. 12a*).

Build the midrange enclosure (K, L, M) as shown in *Fig. 12b*. Drill a 0.8cm-diameter hole on a side panel (L) for the speaker wires. Position the box to the front face, assemble R and S and position it to the rear of midrange box.

Next, glue and nail the two bracings (Q) on the removable trap door, P (*Fig. 12c*). The outer face edges of panel P must be chamfered.

Cover every wall of the enclosure with fiberglass, including top and bottom (F and G). You do not need to install fiberglass in the cavity, V_F, or in the vent.

Place the front panel (A) in position. Glue and nail the support (R) for the midrange box on the back panel. Position the vent, ensuring air tightness by a circular silicone bead on panels A and J.

Then, place some fiberglass in the midrange box and mount the midrange driver after you pass the midrange wires through the hole in panel L. Use a 2m length of wire (which will be later connected to the crossover).

Mount the tweeter. Solder all the connections between the speakers and the terminals according to the crossover scheme (see *Fig. 13*). Due to the very simple network, you can build it with or without a crossover board. You must attach the L-pad on the inner face of the trap door, P, so that the axis goes out of the enclosure. You can modify the crossover if you wish. This one works well and is simple. Remember that the cavity volume, V_F, is critical and every supplementary component installed in the compartment will diminish the volume and modify the alignment. If you wish to have access to the filter (by the removable trap door) you must pass all the wires coming from the various drivers

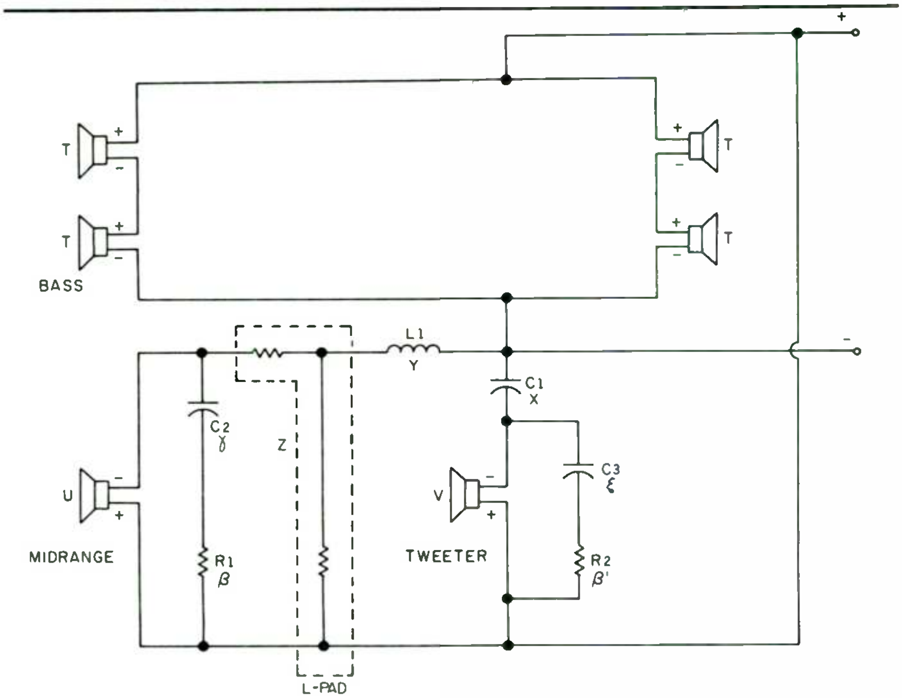


FIGURE 13: Delta system crossover.

PARTS LIST II

CROSSOVER

	DESIGNATION	VALUE	QUANTITY
X	Capacitor	4.02μF	2
Y	Coil	0.19mH	2
Z	L-pad attenuator	8Ω	2
β	Impedance compensating resist.	7Ω	2
β'	Impedance compensating resist.	11Ω	2
γ	Impedance compensating cap.	8.75μF	2
ξ	Impedance compensating cap.	16.0μF	2

• **Miscellaneous Parts (dimensions in cm)**

2	Δ Vent, inner diameter 10.0, length 25.0
8	Casters, 5cm in height
2	Black grille cloth, 107.0 x 22.0
12	Grille fasteners, male and female
500	Flat head nails, L = 2.5
5	16-gauge stranded insulated wires (a 10m long roll)
1	roll masking tape
1	Wood glue
2	container of white silicone
1	gasket material
1	black paint bomb
1	fiberglass roll

• **Electrical Components**

T	Polydax HIF12EB-8Ω woofers		8
U	Fostex FE103-8Ω midranges		2
V	Polydax TW51A-8Ω tweeters		2
W	Circular flush mount	diameter 7.5	
	Push-type speaker wire terminals		2

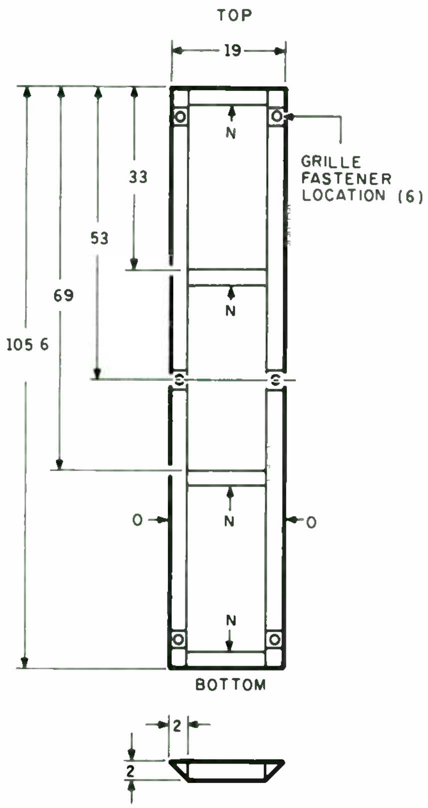


FIGURE 14: Grille frame construction.

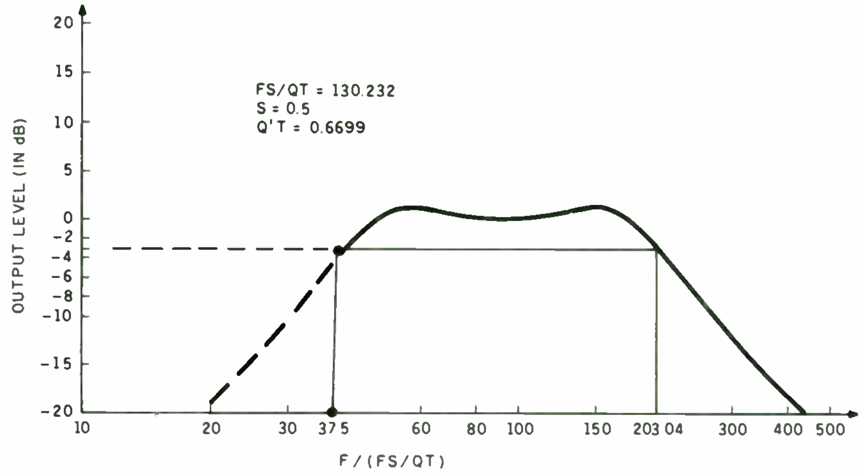


FIGURE 15: Frequency response of the bass section.

and from the terminals through the 1.5cm-diameter hole in panel J, and solder all the connections there.

Now assemble panel (E) along the front face (A) and also a side panel (D). Install the last panel (E). Don't forget to glue and nail the back panel to the midrange enclosure brace (S and R).

Install the circular flush mount push-type speaker wire terminals on the back panel (B). Position the trap door. Screw the battens (Q) on panel B. Ensure air tightness with a silicone bead.

Nail two battens (L) which constitute the enclosure base, unless you prefer to use casters. You can also use this space to place the filter. It is a matter of taste.

For the finish, I use cabinet-quality wild cherry veneer. *Figure 14* gives you the grille frame dimensions.

HOW DOES IT SOUND? The bass is there, and in just the right amount. I am however, impressed by the bass depth, particularly considering the enclosure volume is less than 60 liters and the drivers are inexpensive. The Delta performs admirably on all types of music but particularly well with recordings of large orchestras. I detected no significant overhang on drums and other percussion. The acceleration is fantastic and the bass sound extinguishes at once. The Delta is exceptionally clean and images well. Finally, I am able to listen for long periods without fatigue.

DRIVERS. The HIF12EB, manufactured by Audax in France, is not available in the USA. I believe the Fostex FE103 and the TW51A are available in the USA. Anyway, two drivers stores in Paris are ready to sell to American clients. The price, without shipping cost, would be around $20 for each HIF12EB, $40 for the Fostex and $20 for the TW51A, or $280 total for a pair of Deltas.

If you prefer to buy your drivers on the American market, I list in *Table VI*, alternatives to the HIF12EB. I have chosen similar drivers, so the performances should be almost the same. With my personal computer-aided speaker design program, I have calculated the required vent area and length. Of course, I kept the cavities' volume, V_F and V_B unchanged. Since 22.3cm is the vent's maximum length, compared to the enclosure depth, I suggest you to use a 20–22cm vent.

Finally, all these drivers have a different sensitivity than the Polydax so you will have to adjust midrange and tweeter damping. I know the KEF and the HIF-13FSM are much better quality than the HIF12EB, but they cost twice as much.

At the last minute, I checked the HIF12EB sensitivity, which is 88dB in the bass range and 92dB for the midrange; so you must damp the Fostex FE103 for 4dB more.

SUMMING UP. I hope this article has been useful to the home constructor, and I am always willing to answer questions and assist readers who may be having problems. I would appreciate a SASE if you write, with two international postal coupons, if you wish a reply.

CONCLUSION. My aim in writing this article, is to give speaker builder enthusiasts the opportunity to enlarge their design field. Two audiophile champions, Briggs and Small, have taught us that miracles do not exist in audiophile land. The symmetrically-loaded system is only one way to attain some goals (frequency response, power handling, transient response, and so on). I do not pretend

Continued on page 64

TABLE VI

ALTERNATIVE DRIVERS FOR HIF12EB*

	ALIGNMENT PARAMETERS						VENT	
	S	Q'_T	F_L (Hz)	F_H (Hz)	F_0 (Hz)	SPL (dB) atten.	D_v (cm)	L_v (cm)
Polydax HIF13FSM	0.537	0.6047	35.21	189.66	81.72	−1.5	9.1	21.4
Peerless TO 125F	0.574	0.6345	36.96	162.72	77.55	+0.49	8.6	21.5
KEF B110 SP1003	0.565	0.5507	28.78	158.35	67.51	−2.26	7.6	22.5
Focal 5N411	0.775	0.4361	34.26	119.95	64.10	−0.80	7.0	21.1

*Values based on 4 drivers, series parallel connected.

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PHILIPS SPECIALS

1″	Dome Tweeter	$ 5.00
12″	Woofer	$12.00
10″	Woofer	$10.00
4″	Full-Range	$ 8.00
8″	Woofer	$ 7.00
5″	5060W	$ 7.00

PRECISION SPECIALS

TD	205R	$21.00
TD	165R	$17.00

These two models are in and others are due!

TWEETER BLOWOUT!!!

LR10 $5.00 SR10 $8.00

Original K010DT $12.00

TD 255F $20.00

POLYDAX SPECIALS

TX	2025TSN TPX Woofer	$27.00
HD	20B25J 8″ Woofer	$ 8.00
HD	20B25H 8″ Woofer	$17.00
HD	13B25J 5″ Woofer	$10.00
HD	24B45 10″ Woofer	$27.00
HIF	13J 5″ Woofer	$ 8.00
HIF	17H 6.5″ Woofer	$15.00
HD	100HR 4 Ohm Tweeter	$ 8.00
HD	12x9 4 Ohm Tweeter	$ 7.00
HD	9x8 8 Ohm Ferrofluid	$ 7.00

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SYNTHETIC SOUND FORETELLS GOOD LISTENING

BY LEIF RYDEN

It has been said that an art is a science with more than six variables; and among the sciences, that of musical acoustics seems to be unusually complicated and obscure. Not only must we make calculations and measurements in four dimensions—three spatial dimensions plus time—but for this work to be useful, it must be correlated with the listener's perception.

The interrelation between musicians' instruments and the concert hall is very influential on their ability to make great music, and different kinds of music demand different kinds of acoustic conditions. Thus, I believe the acoustic design of concert halls is an art.

Some of the greatest of the concert halls are quite old; the Symphony Hall in Boston, for example, was built in 1900; the Grosser Musikvereinssaal in Vienna, in 1870; and the Concertgebouw in Amsterdam, in 1888. Yet despite presumed advances in our understanding of acoustics, halls that have been built during the twentieth century often have been mediocre, if not outright disasters. Not unexpectedly this has given rise to rumors of forgotten secrets; in fact, current research has shown that the acoustical excellence of older buildings is very much due to the constructional restraints and architectural taste of the nineteenth century.

The old halls were rather narrow, with a rectangular plan-form and a high ceiling, a shape usually known as "shoebox." They were also richly decorated with pillars, pilasters and statues, and had coffered and ornamented ceilings and a good deal of plush fabric and upholstery. All this contributed to good acoustics.

In the twentieth century, halls became wider to accommodate larger audiences and their style of architecture changed. The rich decoration disappeared; walls and ceilings were designed as more or less flat surfaces. Acoustics were seldom the equal of some older halls, yet at first the reasons for this were far from clear.

At the beginning of the twentieth century, science was introduced into room acoustics by Wallace Clement Sabine, with his concept of "reverberation time." This is the time required for the sound level to decrease by 60dB after cessation of the sound that created it. The reverberation time depends on the ratio between room volume and the amount of acoustic absorption in the room. In typical concert halls this ab-

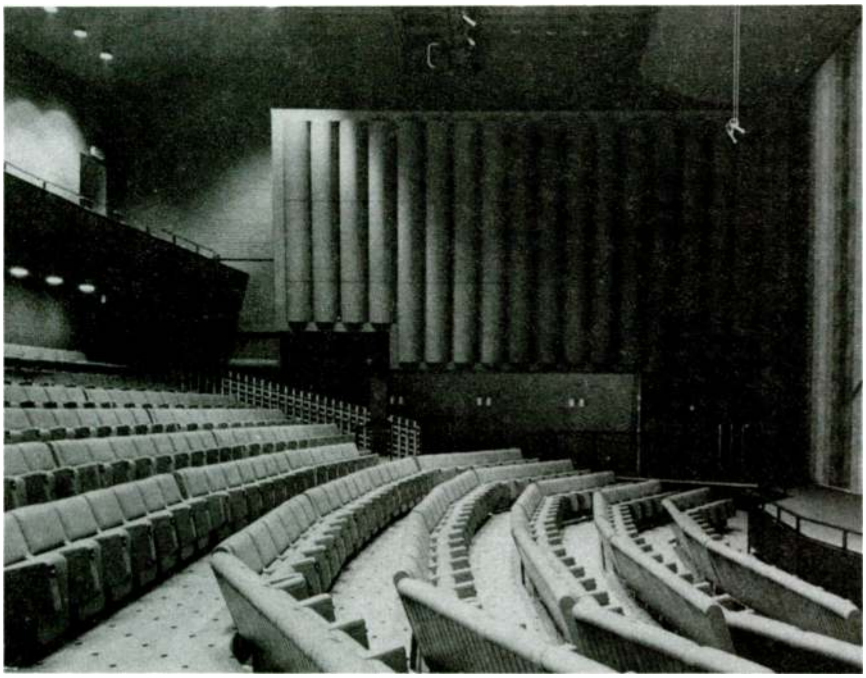


PHOTO 1: The concert hall in Motala, Sweden contains a multi-purpose auditorium with variable acoustics. The cylinders on the side walls are covered by a sound-absorbing material on half their circumferences. With the cylinders so positioned that the absorbent faces the auditorium, the acoustics are suitable for theater performances where voice is the dominant sound. Turning them to present their reflecting sides, to attain a longer reverberation time and more lateral reflections, and enclosing the stage with a concert shell to direct sound into the auditorium, the hall becomes suitable for musical performances.

ABOUT THE AUTHOR

Leif Ryden graduated from Chalmers University of Technology (Sweden) with a Masters Degree in architecture, specializing in architectural acoustics. He has been working with research in room acoustics and in industrial noise control, and marketing acoustical products. He is currently a consultant with Ingemansson Acoustics in Stockholm, working in architectural acoustics and sound system design. Mr. Ryden is mainly interested in loudspeaker design, but his current intention is to get his hi-fi system finalized, i.e., "to stop tinkering and start listening." He is a member of the Audio Engineering Society.

sorption is mainly due to the presence of the audience, and as the audience area is approximately equal to the floor area, the reverberation time is in turn roughly proportional to the ceiling height.

As investigation continued into room acoustics, researchers became aware of "early reflections"—the direct reflections from walls and ceilings. Sound propagates, within certain restrictions, in the same way as light and it is possible to draw "sound rays" to determine how it is reflected. In the wide halls that were built in the twentieth century, ceilings were designed to reflect sound into the audience. This gave clarity and brilliance to the sound, but little richness of tone and little of the sensation of being immersed in a sound field.

MODERN ACOUSTIC CRITERIA. During the past 30 years new measuring equipment and evaluation methods have enabled acousticians to understand much more clearly the criteria that lead to good acoustics. To compare different concert halls, to discover what factors are important, is difficult, as one has to travel great distances between different halls and then, probably, listen to different kinds of music performed by different orchestras.

Another approach is to keep the listeners seated in one position and "bring the halls to the audience." This can be done rather well by the "dummy head" recording technique: microphones are

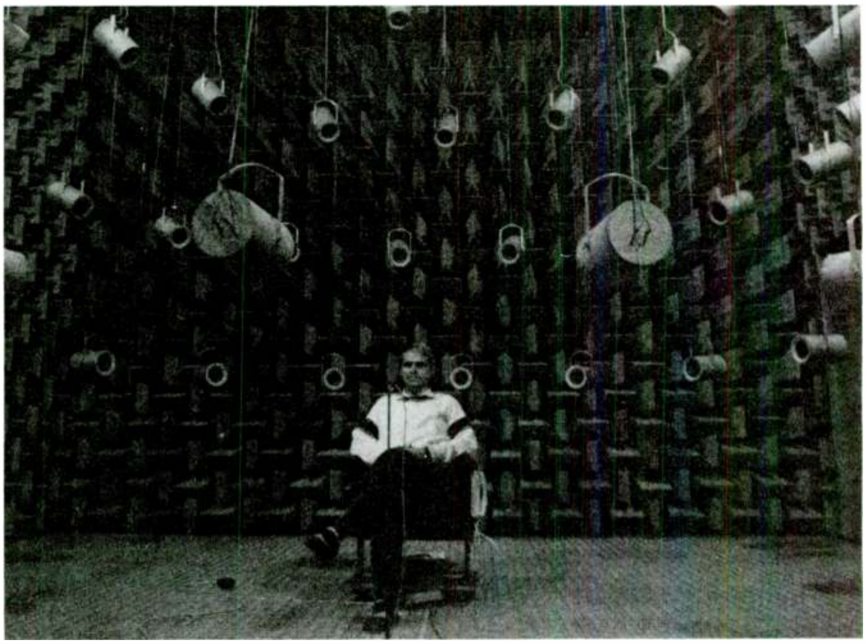


PHOTO 2: A synthetic sound field in the anechoic chamber at Chalmers University of Technology in Gothenburg, Sweden. Fifty loudspeakers are placed in a hemisphere, facing to the center. The loudspeakers can give direct sound, reflections or reverberation from any direction, and thus changing acoustic conditions can be presented to a listener seated in the center of the field.

This article appeared originally in *Veritas*, European edition, ©1986, 112 Station Rd., Sidcup, Kent, England.

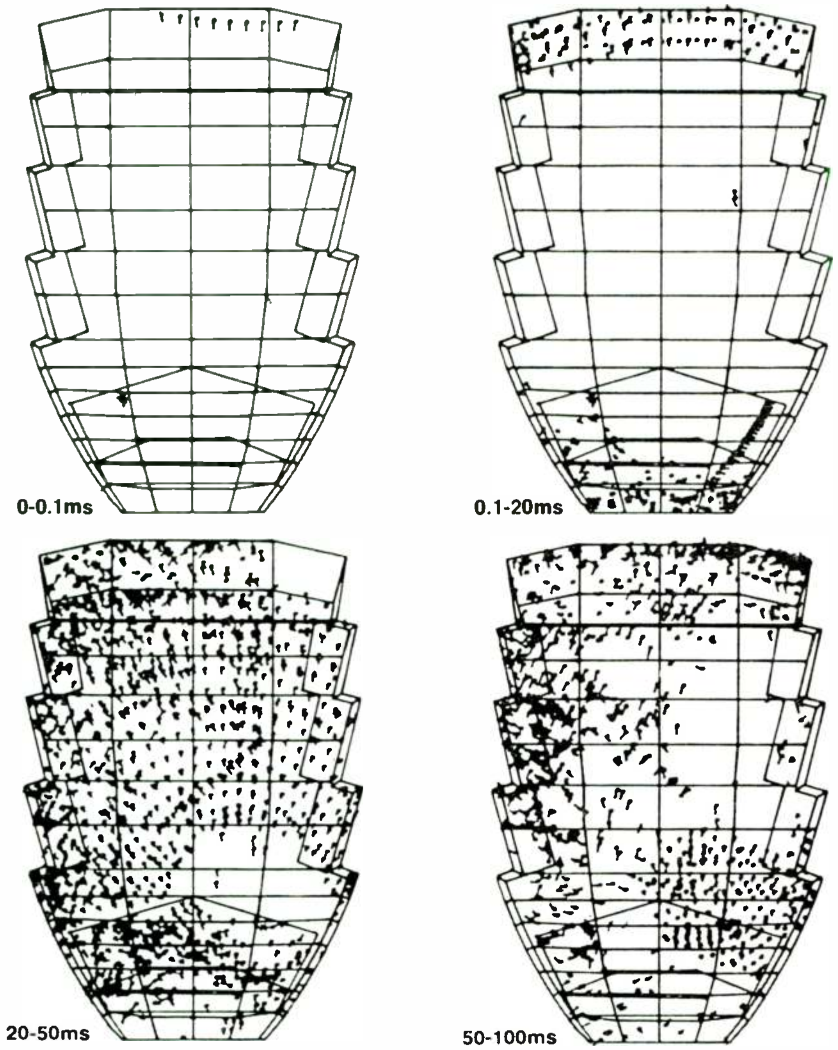


FIGURE 1: A ray-tracing plot of the Gothenburg Concert Hall before its modification. The four views depict sound-ray hits for different time intervals. Each hit is shown by a dot whose tail shows the direction of incidence. The length of the tail is its projection on a horizontal surface—a vertical hit shows only as a dot.

The sound source is at the front and to the right-hand side of the stage. The time is referenced to the arrival of the direct sound, so that zero time in each case is dependent on the distance from the sound source.

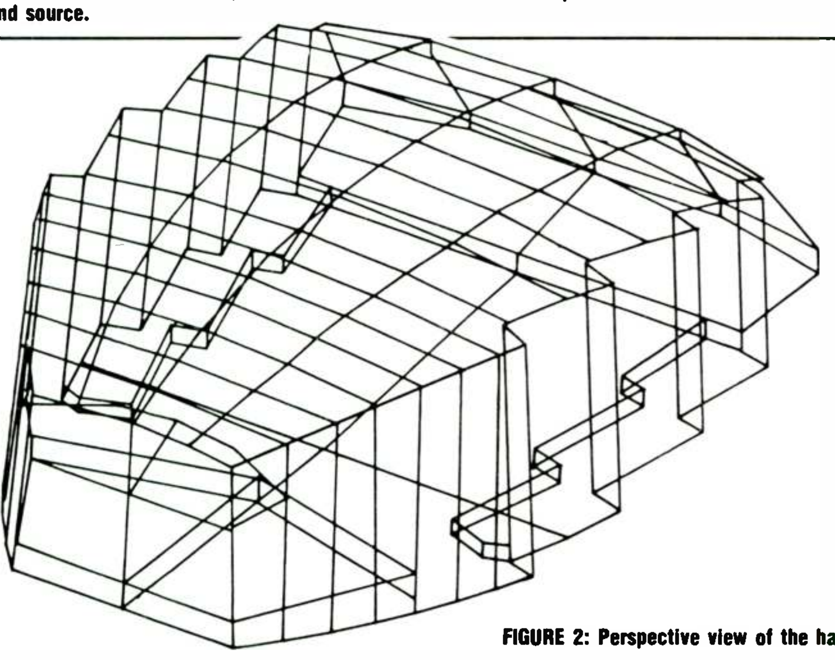


FIGURE 2: Perspective view of the hall.

placed in the ears of a dummy head and the sound is recorded through them. When this recording is played back through headphones, the original acoustics are reproduced very accurately, and this method gives results close to those obtained by listening in the actual auditorium.

Dummy-head recording takes place in closely controlled conditions. A recording of music that has been made under anechoic conditions is replayed through loudspeakers on the stage of the hall being examined. The dummy head is placed in those listening positions that are of interest, and a series of recordings made. Using the same original record, replayed and recorded in many halls, the dummy-head recordings are then used for making listening tests.

In this way it has been possible to make listening tests under reproducible conditions, and some of the more important factors are the following.

- *Early decay time* is the time, multiplied by six, for the sound level to fall by 10dB. This corresponds more closely with your sensation of reverberance than does reverberation time.
- *Clarity* describes the ratio between the energy during the first 80msec of an impulse, and the energy which arrives after that time.
- *Lateral efficiency* is the ratio between energy that comes from the sides of the hall and the total energy, again during the first 80msec. The higher the proportion of these lateral reflections the better; in other words, the more stereo the better.

Due to their narrow, rectangular design, the best nineteenth-century halls give a high proportion of lateral reflections, their abundance of decorative elements gives a good diffusion of the sound and their high ceilings give a relatively long reverberation time. All this adds up to excellent musical acoustics.

REAL AND COMPUTER MODELS.

Certainly, enough is known today about acoustical factors to design a good concert hall. However, the relation between these acoustical factors and the physical design and building of the hall is quite complex. It is seldom possible to simply calculate these acoustical properties; instead designers use a variety of modelling techniques.

As early as the 1930s, scale models were built of concert halls during the design stage. Sound was then radiated in the model and recorded. In such models, the frequencies of the sound used must be multiplied by the reciprocal of the scale, so that in a 1:10 scale model 1kHz becomes 10kHz, for example, and all other acoustical properties must be scaled in the same way. In the early days, this meant that either the model must be very large and expensive or, with a smaller model, technical difficulties were encountered due to the high frequencies necessary.

In recent years computer models have been constructed by designers. Two different models have been developed for use at Ingemansson Akustik, Stockholm, the ray-tracing method and the mirror-image method. In both, the coordinates of the surfaces of the hall are stored in the computer memory, and the sound propagation then simulated in the computer.

In the ray-tracing method, sound rays are radiated from a sound source into the hall. Each sound ray propagates through the auditorium and is reflected at the surfaces, according to certain rules, until it impinges on the receiving surface—the audience area—where its "hit" is registered. For each ray, the computer stores the position of the hit, the direction of incidence and the total ray length. Since the velocity of sound is constant, the time delay of each hit can be calculated from the ray length.

The hits are then plotted on a plan of the hall for different time intervals. The energy incident on a certain surface can also be plotted as a function of time, or angle of incidence. In this way, we can determine the spatial and temporal dispersion of the energy and the angular dispersion of the hits. The method is relatively simple, and is useful for a quick overview of the problem.

In *Fig. 1*, the early reflections into the auditorium are close to the end walls only. During the next interval, the reflections are dispersed throughout the listening area, except the right rear corner and a significant "hole" in front of the sound source. The hole is filled with reflections from the ceiling during the next interval, which could cause a major echo there.

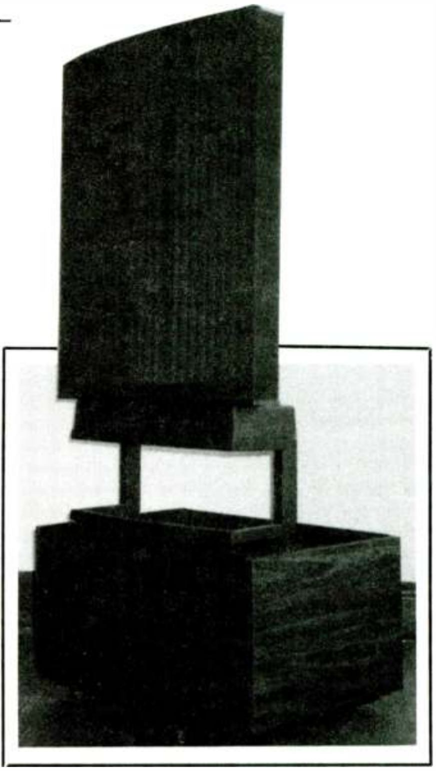
The mirror-image method is more complicated but gives more detail. The mirror images of the sound source in the hall surfaces are first calculated in terms of first-order images. Second-order images are then calculated as reflections of the first-order images, and so on, up to sixth-order. The direction and distance for each image is then calculated for each receiver position. From this data we can calculate the factors necessary for good acoustics, and by repeating the exercise for many receiver positions, determine the acoustic conditions of the entire hall.

The mirror-image program as used by Ingemansson Akustik may also include control of a synthetic sound field. This consists of 50 loudspeakers arranged on a hemisphere, pointing to its center, in an anechoic chamber. The listener sits in its center and hears the sound as it should be heard in the hall being modelled. In this way it is possible to listen to music "in the auditorium" while the actual building is still on the drawing board. We can store changes to the hall in the computer memory, and make instantaneous comparisons between different shapes and proportions of auditorium.

Good acoustics is not only a matter of technical knowledge, but also of taste and discrimination; in this particular field, art and science are mutually dependent.

Craftsman's Corner

Dual-Chamber Reflex



I have constructed a variation of the dual-chamber vented enclosure to use with my Quad 63 ESL. My subwoofer serves two purposes: to raise the center of the electrostatics to ear height (for improved dispersion and to avoid floor reflections), as well as to extend the bottom end of a very good speaker—both worthwhile improvements.

While I designed the enclosure's dimensions (approximately 24 by 24 by 14 inches) to match the Quad, it could easily be used "as is" with other systems, or modified to suit your particular decor or system requirements. It also makes a great endtable.

I made a pair, one for each Quad, and am very happy with the results.

The drivers are Dynaudio 24W100s in a 2.7-ft.^{3} enclosure. I used an automotive undercoat and sand mixture to damp the ¾-inch particle board cabinet. The front, rear and bottom panels are 1½-inch thick, which I beveled to provide a more attractive enclosure than a squared-off box. I attached the bottom kick base permanently, and then used four, point-loading spikes.

These woofers successfully match the Quads' transient speed and low coloration. I realize some will take this as heresy (to even propose a subwoofer) but this particular design does not detract from the 63s' performance, and enhances dynamics, bass extension and freedom from low-frequency coloration. I do not recommend the use of other drivers or less time-consuming cabinet construction without careful consideration of the consequences—although I am sure several other drivers would work well. I researched and listened to a number of drivers before settling on the Dynaudios.

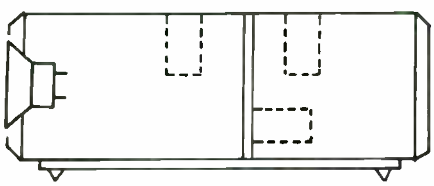


FIGURE 1: Side view.

I have been intrigued by the double enclosure, ever since Dick Marsh brought it to our attention (*SB* 3/80) and David Weems showed us how to accommodate different drivers and tuning criteria (*SB* 4/85). Both stress it is a fairly forgiving design.

The top was really the only space available to vent them, and as the wavelengths are fairly long, it didn't upset the design too much. Grille cloth covers the front, top, and back of the enclosure; use a very loose weave to avoid buzzing in the port exit area.

Construction is fairly straightforward (except for curing time of the undercoating) and I used approximately 100 wood screws per cabinet (*Fig. 1*).

Results

While not subterranean in extension, it does noticeably complement the Quads. It doesn't sound like a subwoofer, but like Quad 63s with a good, solid, detailed and high-speed bottom end (which is exactly what we want).

The relatively high impedance of the Quads in the low end means that no crossover elements need be inserted in its signal path for a passive network. And while the dual-chamber design does exhibit several high impedance peaks, it seems to work well in this instance. Although I am sure biamping would be the preferred method, it does allow a minimal component, passive network (without excessively low impedance) for those who use one good amplifier to run their system (*Fig. 2*).

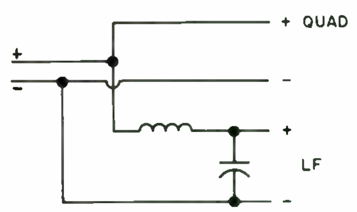
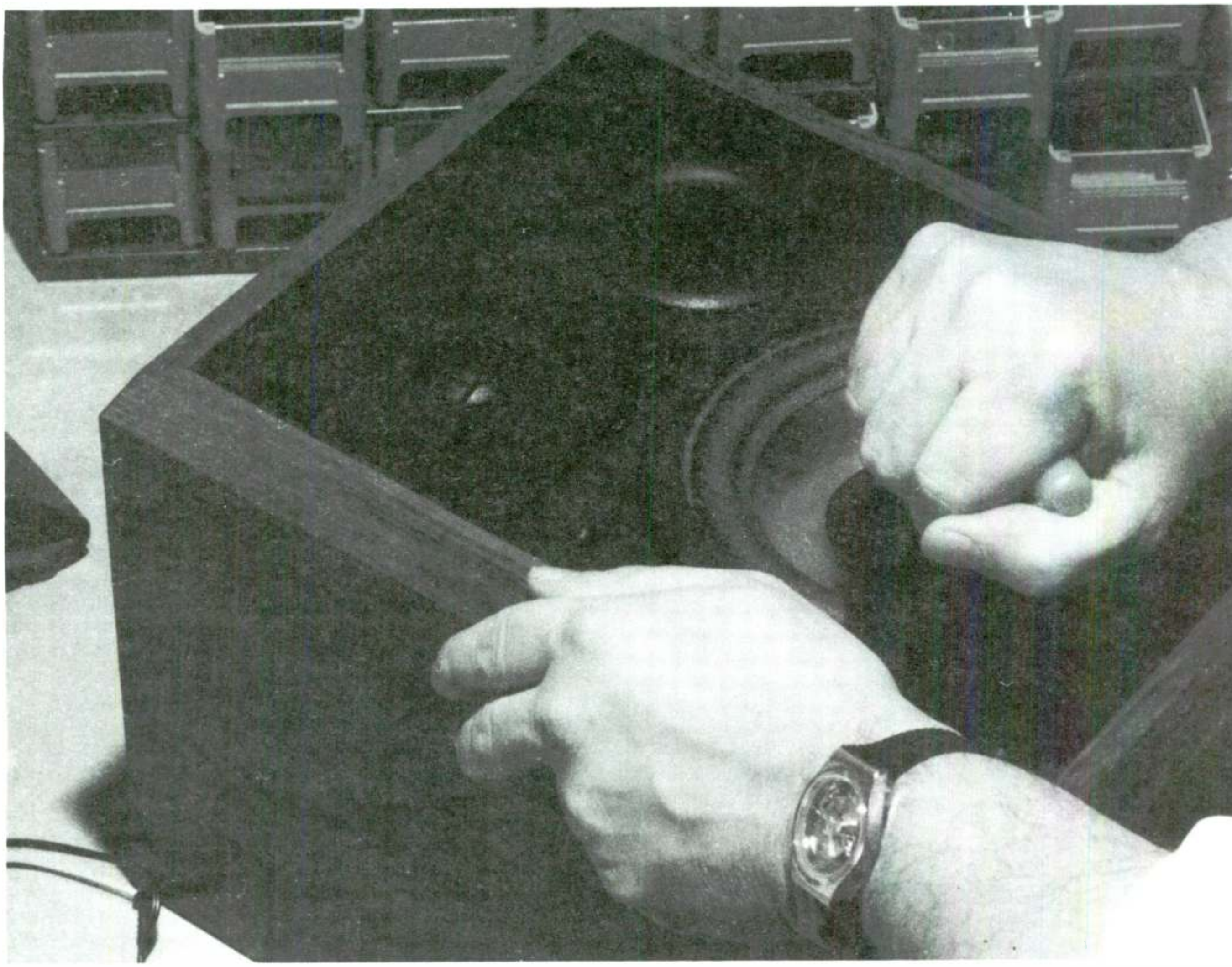


FIGURE 2: Passive crossover for the subwoofer.

The 12dB/octave passive network has proven to be a good compromise between rolloff rate, audibility, and component expense. My system consists of a POOGEd Philips 650; a POOGEd, Sulzer-powered Dayton-Wright SPA Preamp; a 100W/channel, Class A amplifier; and an Oracle, well-tempered arm and Shinon Shaphic cartridge. Royce/Neglex interconnects complete my system.

Dave Pitt
Weston, Ontario
M9R 3H8 Canada



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Fast Reply #ED186

Software Report
Numbers and Networks

by Bob White

CACD: Computer-Aided Crossover Design, Version 1.0.8; Scientific Design Software, PO Box 3248, Chatsworth, CA 91313. Price: $349; machine: IBM XT, AT, PS-2, and true compatibles.

In the engineering world I have been exposed to all manner of design tools; mainly tables, charts and graphs, to make design easier and faster. Over the last few years there has been a big push in our industry to use computer-aided design and drafting (CADD) and as a computer buff I have embraced them enthusiastically. Now that this is commonplace in the corporate world they are now trickling down to the small manufacturer/hobbyist level.

An excellent example of CADD for the speaker industry is Computer-Aided Crossover Design (CACD), which is the work of two people, Ted Telesky and Robert Caudle, of Scientific Design Software. The basis for the program is the original work done by Peter Schuck in his Master's thesis "Numerical Optimization of Loudspeaker Crossover Networks."

To use this program and others like it, we assume that loudspeakers are minimum phase devices. Standard crossovers using ladder topology are minimum phase, so the addition of a minimum phase driver and a minimum phase crossover circuit should produce a minimum phase response ("minimum phase" for our purposes means that there is a smooth, constant rate of change in phase across the operating range of the driver). Some research suggests that most of the response error produced from a textbook crossover is due to the interaction of the bandpass nature of all real drivers and the additional filter. Many of us assume the response errors are all created by textbook constant resistance component values. Not so. The flexibility designed into this software makes it a snap to explore this and many other aspects of driver/crossover interaction.

In the past we have all designed our crossovers with the required number of components and proper placement, built them and hoped for the best. For example, a fourth-order, low-pass Linkwitz-Riley crossover has two series inductors and two shunt capacitors. The optimized network that produces the desired fourth-order transfer function in this case, contains one series inductor and a series resistor/capacitor combination in shunt across the driver. Some of you would recognize this as a second-order topology. The addition of a modified second-order filter topology to a roughly second-order driver rolloff produces the desired result. Keep in mind that this is not always the case, but the crossover works if the driver's rolloff frequency is close to the desired cutoff and a close approximation to a natural second-order rolloff rate.

The software package includes a ring binder that contains three manuals and seven 360Kbyte floppy diskettes. A speaker design tutorial, "Loudspeaker System Design Handbook," is included and should be required reading for any serious program users. It is well-written and should be made available as a separate product. The two remaining manuals are for the individual parts of the software—FILE MANAGER and CACD.

CACD hardware requirements are as follows: IBM PC/XT/AT, or PS computer or equivalent with 512Kbytes of memory, MS-DOS 2.0 or later, one floppy and one hard drive with at least 1.2Mbytes, CGA, EGA, VGA video card and monitor. Optional equipment includes an Epson standard printer with Centronics interface; and an 8087 math coprocessor will speed things up 10 times, which is a significant improvement when you are optimizing many components in a network.

CACD is split into two large programs with separate functions. File Manager func-

PHOTO A: Original JBL L100 speakers used for this review. (The revised version uses a Focal T120K tweeter.)

tions as a large driver database. You enter Thiele/Small parameters, driver response and impedance data, using screens with easy to learn keystrokes for editing. Upon entering File Manager the screen clears and displays the following choices at the top of the screen:

1. FILE (Alt-F)—about File Manager; shell; quit.

2. BRAND (Alt-B)—list brands; add or delete a brand; update brand file; view all brand information.

3. DRIVER (Alt-D)—list all drivers; load a driver; add, update or delete a driver; view parameters; view response curve (SPL or 0dB); view impedance curve; view impedance model.

4. SEARCH (Alt-S)—search by parameter; search by system design; search by response curve.

5. MODEL (Alt-M)—generate free-air model.

6. UTILTIES (Alt-U)—create driver database.

7. OTHER (Alt-O)—define screen colors; define quickeys; driver's path name; define cursor; define notes; update information.

Most of the choices are self explanatory, but I will highlight some of the more useful features.

The brand part of the File Manager has space for 250 brand names which can store 250 drivers, so the total driver storage is 62,500 drivers. Each driver is a record composed of the Thiele/Small parameters, a collection of data points of actual SPL and relative output (0dB), raw impedance curve data, and some components from the MODEL utility.

Search allows you to find that one driver in the database that suits your needs, by generating reports of the available drivers that meet the search criterion. The criterion can be one or more Thiele/Small parameters or a response curve within ± 2dB, 200Hz-2kHz. In case you need space for more drivers or you want to set up your own libraries, you can make floppy disks for database storage.

You will spend much of your time in the File Manager looking at parameters and response curves for that one driver suitable for your project. Although few drivers in the database include their response and impedance information, as most of you know, manufacturers' response curves seldom match your driver sample. Thus, results will be more accurate, since you must supply each driver's own unique response and impedance curves.

The second part is CACD itself. It starts out just like File Manager with a banner of choices across the top of the screen, as follows:

1. FILE (Alt-F)—run file manager; about CACD; shell; quit.

2. CIRCUIT (Alt-C)—load circuit file; add, update or delete a file; edit; new circuit; starting values; optimize circuit; print file.

3. DRIVER (Alt-D)—load driver data; view parameters; view response curve (SPL); view impedance curve; view impedance model; transfer driver to circuit.

4. TARGET (Alt-T)—calculate acoustic target; edit acoustic target; correction voltage data; optimizer's target data.

5. GRAPH (Alt-G)—predicted acoustic response, input impedance and correction voltage; working response; working impedance; correction voltage; target response; overlay graphs.

6. MODEL (Alt-M)—view free-air; calculate closed and vented box.

7. UTILTIES (Alt-U)—edit working response and impedance; impedance compensation; make a circuit file disk.

8. OTHER (Alt-O)—define colors; define quickeys; define cursor; driver's path name; circuit's path name; update information.

A typical session might go something like this:

• Load a driver from the database and view the impedance and the response curves. Transfer driver to circuit.

• Load the circuit file "STD.DRIVER.CIR."

• Load the circuit file "STD.LP.2.CIR."

• Edit the circuit to see the driver model and the crossover, and obtain the starting values.

• Go to TARGET and calculate the target transfer function. Then, enter your frequencies and optimize the circuit.

• View the predicted acoustic response under GRAPH.

```
SCIENTIFIC DESIGN SOFTWARE
Driver Parameters From Measurement Data

Date: 07-16-1988
Data for driver: jbl 123a

Entered Data as Follows:

          Entered driver DC resistance (Re)      4.50 ohms
     Entered driver resonance frequency (Fs)    22.00 hertz
     Entered driver maximum impedance at Fs     71.00 ohms
               Entered driver F1 frequency      17.90 hertz at   17.90 ohms
               Entered driver F2 frequency      28.80 hertz at   17.90 ohms
            Calculated Square root of F1*F2     22.70 hertz
                   Calculated error factor       3.20 percent
Compliance calculated by ADDED MASS method
                       Entered added mass       20.00 grams
     Entered driver new resonance frequency     19.00 hertz
           Entered driver piston diameter      255.00 mm
            Entered driver magnet gap depth      6.00 mm
            Entered driver voice coil length    12.00 mm

Calculated Thiele/Small Parameters:

         Free Air Resonance (Fs)=SQR(F1*F2)     22.70 hertz
                                        Qts      0.5243
                                        Qes      0.5598
                                        Qms      8.27
       Equivalent acoustic compliance (Vas)    384.81 liters
                          Piston area (Sd)       0.0511 square meters
                        DC resistance (Re)       4.50 ohms
                   Volume displacement (Vd)    153.21 ccm
                  Linear displacement (Xmax)     3.00 mm
                       Power handling (Pe)   TO BE ENTERED
                       Coil Inductance (Le)   TO BE ENTERED
              Reference Efficiency (Ref Eff)    0.77 percent
           Efficiency Bandwidth Product (EBP)   40.55 hertz

Other Calculated Data:

          Moving Mass of Diaphragm only (Mmd)   40.25 grams
  Moving Mass of Diaphragm & Air Load (Mms)     46.79 grams
          Mass of Air load on diaphragm (Ma)     6.55 grams
                           Compliance (Cms)      0.00105 m/N
                            BL product (BL)      7.32 N/A
                     Sensitivity (SPL 1w/1m)    90.88 dB

                              END OF REPORT
```

FIGURE 1: Thiele/Small parameters report for JBL 123A woofer, generated by the Parameter Measurement Module.

Did the optimizer meet your target? If not, then re-edit the circuit topology. Success? Build the network.

I describe a highly simplified version of designing a crossover using this software. You must make many judgement calls concerning the suitability of the driver/crossover combination and research the compatibility of the drivers. After using it awhile, you will develop an intuition about circuit topology and its effects on the driver, for example, knowing where and when to add a resistor in the shunt arm of a crossover.

I found a likely crossover project; a pair of 15-year old JBL model L100s (*Photo A*) with good drivers, popular in their day, which present-day design techniques should improve markedly. The primitive crossover consists of two capacitors ($8\mu F$ and $3\mu F$) and two 8Ω L-pads.

First, we measure the impedance curves and the frequency response of the drivers in their enclosures. That done, I used a separate program from SDS called "Parameter Measurement Module" to calculate the T/S parameters from the impedance curve (*Fig. 1*). Next, enter the parameters of the drivers from CACD's File Manager. The raw frequency response (*Fig. 2*) and impedance curve (*Fig. 3*) are entered for each of the drivers from a screen (*Figs. 4* and *5*).

To optimize the impedance, "Generate Free-Air Model" approximates the circuit model of the driver in free-air using four frequencies above the minimum impedance and some T/S parameters. These components are stored as part of the driver data

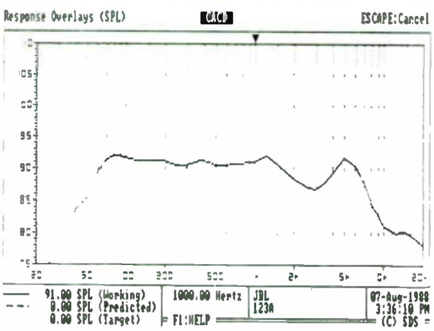


FIGURE 2a: Frequency response of JBL 123A woofer in enclosure.

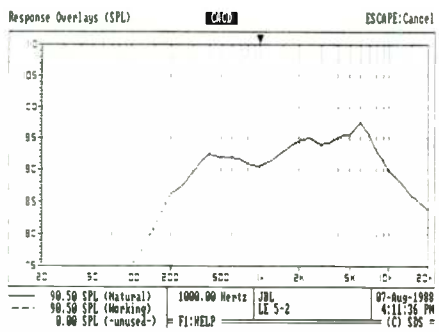


FIGURE 2b: Frequency response of JBL LE5-2 midrange in enclosure.

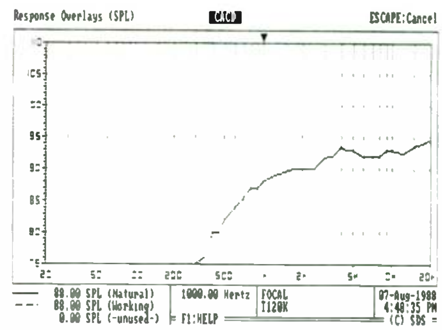


FIGURE 2c: Frequency response of Focal T120K tweeter on baffle.

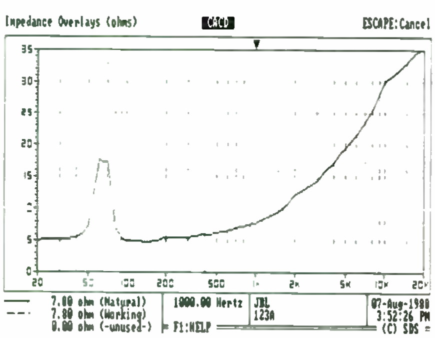


FIGURE 3a: Impedance curve of JBL 123A in 40-liter enclosure tuned to 18Hz. This is the "working impedance."

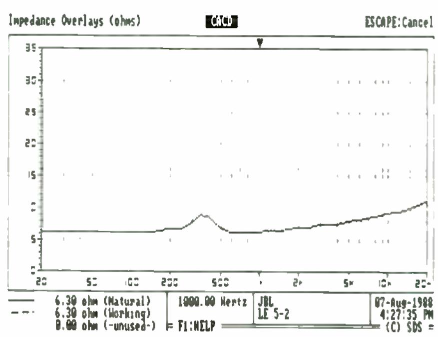


FIGURE 3b: Impedance curve of JBL LE5-2 in .5-liter enclosure.

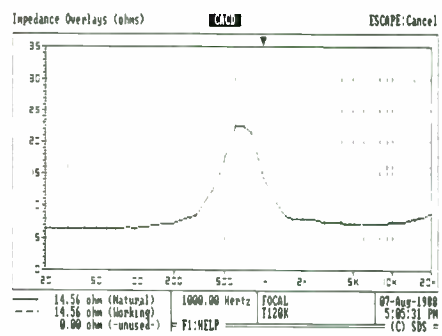
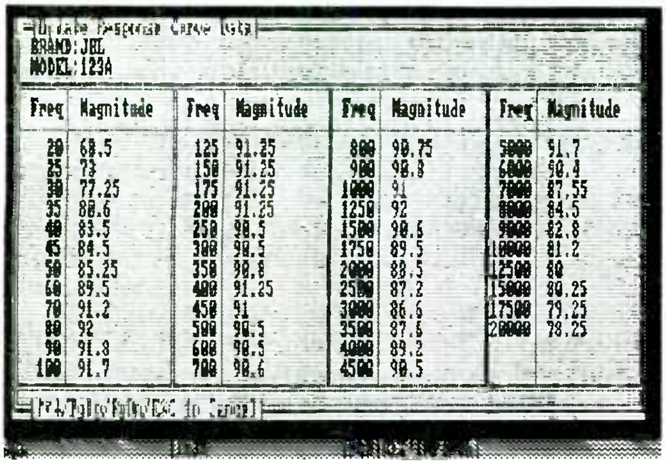


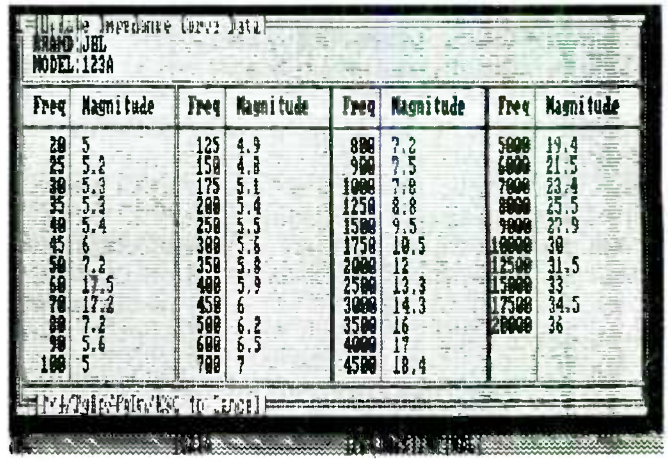
FIGURE 3c: Impedance curve of Focal T120K.



Update Response Curve Data
BRAND: JBL
MODEL: 123A

Freq	Magnitude	Freq	Magnitude	Freq	Magnitude	Freq	Magnitude
20	68.5	125	91.25	800	90.75	5000	91.7
25	73	150	91.25	900	90.8	6000	90.4
30	77.25	175	91.25	1000	91	7000	87.55
35	80.6	200	91.25	1250	92	8000	84.5
40	83.5	250	90.5	1500	90.6	9000	82.8
45	84.5	300	90.5	1750	89.5	10000	81.2
50	85.25	350	90.8	2000	88.5	12500	80
60	89.5	400	91.25	2500	87.2	15000	80.25
70	91.2	450	91	3000	86.6	17500	79.25
80	92	500	90.5	3500	87.6	20000	78.25
90	91.8	600	90.5	4000	89.2		
100	91.7	700	90.6	4500	90.5		

FIGURE 4: Frequency response entry screen in CACD.



Update Impedance Curve Data
BRAND: JBL
MODEL: 123A

Freq	Magnitude	Freq	Magnitude	Freq	Magnitude	Freq	Magnitude
20	5	125	4.9	800	7.2	5000	19.4
25	5.2	150	4.8	900	7.5	6000	21.5
30	5.3	175	5.1	1000	7.8	7000	23.4
35	5.3	200	5.4	1250	8.8	8000	25.5
40	5.4	250	5.5	1500	9.5	9000	27.9
45	6	300	5.6	1750	10.5	10000	30
50	7.2	350	5.8	2000	12	12500	31.5
60	17.5	400	5.9	2500	13.3	15000	33
70	17.2	450	6	3000	14.3	17500	34.5
80	7.2	500	6.2	3500	16	20000	36
90	5.6	600	6.5	4000	17		
100	5	700	7	4500	18.4		

FIGURE 5: Impedance data entry screen in CACD.

in the FILE MANAGER. So we have entered the frequency response, the impedance curve and generated a free-air model of the driver. We are through with the FILE MANAGER for this part of the project.

The rest of the project will be executed in CACD. The first order of business is to refine our approximation to the drivers' impedance data. The free-air model generated from FILE MANAGER with its four frequency points is not accurate enough. We can select the Focal T120K tweeter, for example, for our impedance model optimization, a unique feature of this software. The driver is "loaded" and "transfered to circuit," next a driver circuit model is "loaded" and the whole driver model is displayed in "edit circuit." The electrical equivalent of a mechanical resonance is displayed and an approximation of the voice coil inductance/mass effects (*Fig. 6*).

Since we are interested in optimizing the voice coil inductance components (Lvc1, Lvc2, Rvc2) we type "y" in the "optimize ?" column of "edit circuit." Next, we go to "Optimizer Target Data" and choose 11 frequencies from Rv (the minimum impedance) to 20kHz. The program already has all the "working" impedance data in memory for the 46 frequencies the database stores. So you simply enter the frequency, and the program fills in the impedance value. The program senses that you are optimizing the impedance and automatically sets the output node to 1.

If a crossover component is in the circuit, the program prompts you for the output node. Curious users will want to look at the "Predicted Input Impedance" to see how closely the "Free-Air Model" fits the actual "working impedance." For our example, the resonance frequency and height match, but the rest of the curve is off. Next, "Optimize Circuit" takes 59 seconds at

CIRCUIT FILE

Circuit : T120K.OPT.MDL
Brand : FOCAL
Model : T120K
Filter Topology:
Acoustic Target: 2 nd Hp @ 4000

BRANCH	PART	FROM	TO	(+)	(-)	OPT	VALUE
1	RV	1	2	0	0	N	7.200 ohms
2	LVC1	2	3	0	0	N	[illegible].020 mH
3	LVC2	3	4	0	0	N	[illegible].010 mH
4	RVC2	3	4	0	0	N	1[illegible].098 ohms
5	LR	4	0	0	0	N	2.062 mH
6	CR	4	0	0	0	N	2[illegible].760 uF
7	RR	4	0	0	0	N	16.778 ohms

FIGURE 6: Circuit model of Focal T120K in "Edit Circuit."

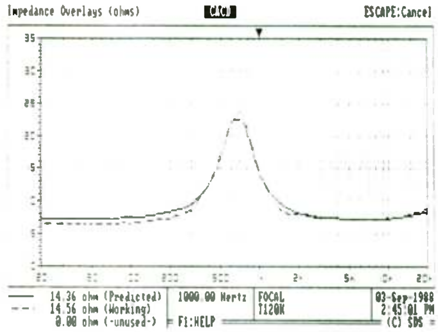


FIGURE 7: Impedance curve raw data overlaid with optimized model generated by CACD. Raw data, dashed lines; CACD model, solid lines.

4.77MHz. The new component values are displayed in "edit circuit" and their results can be checked with the graph, "Predicted Input Impedance."

Optimization of the motional impedance components (Lr, Cr, Rr) uses the same procedure, except the choice of frequencies in "Optimizer's Target Data," where we choose frequency points below Rv that include the resonance. We store the results of the optimizations in a "circuit file" which is added to the circuit file library for future use and display the results in *Fig. 7*. Each of the drivers impedance models are optimized in turn and stored in the circuit file library.

Another feature is optimizing the inductance and motional impedance compensation components that take care of the voice coil impedance rise and the driver resonance. For the woofer in our project, we use "Impedance Compensation" from UTILTIES, to calculate the voice coil inductance compensation network and transfer it to the circuit model as (Rc and Cc). With the model already optimized to fit the "working impedance," we add Rc and Cc to find the components to make the driver look like a constant resistance (see *Figs. 8* and *9*, which are before and after compensation). *[The author generated these curves from PSPICE to check the accuracy of the model and the compensation—Ed.]*

We now enter the crossover design phase of this project and start by choosing the woofer. Load "123A" from "Driver" and transfer to "circuit." Next, load circuit with "123A.Mdl" and "Std.Lp.2," which installs the driver circuit model and the crossover circuit topology in the circuit editor. Use "Starting Values" in CIRCUIT to try values in the crossover. Go to TARGET and calculate the target function, which in this case is a second-order low-pass at 600Hz. Select the "Optimizer's Target Data" frequencies (16 frequencies, 100Hz-1kHz).

The program will automatically enter the "correction voltage" necessary to produce the required target response. "Predicted Acoustic Response" (*Fig. 10*) shows the response of the impedance equalized model and the "starting value" crossover. As you can see, if you equalize the impedance of the driver you get a pretty good result with textbook crossover components. We will optimize this network and view it under "Predicted Acoustic Response" (*Fig. 11*). To ensure the crossover components did not produce a network with too low an input impedance, we look at "Predicted Input Impedance" (*Fig. 12*). Our minimum impedance is safe; about 4Ω.

We load the midrange driver LE5-2, the model and the second-order circuit cross-

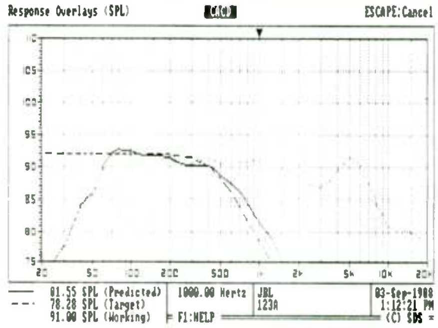


FIGURE 10: Frequency response of JBL 123A with equalized impedance and textbook crossover components values for 5Ω.

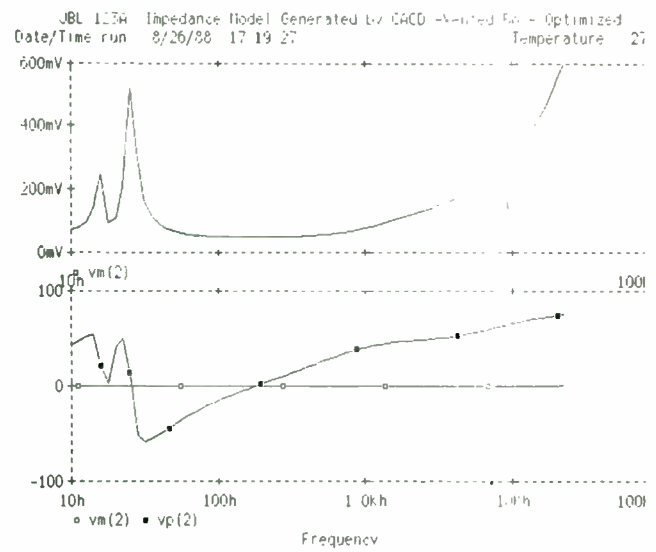


FIGURE 8: Impedance curve of JBL 123A generated from the circuit model of CACD. Upper curve is impedance and lower curve is phase. Generated by PSPICE.

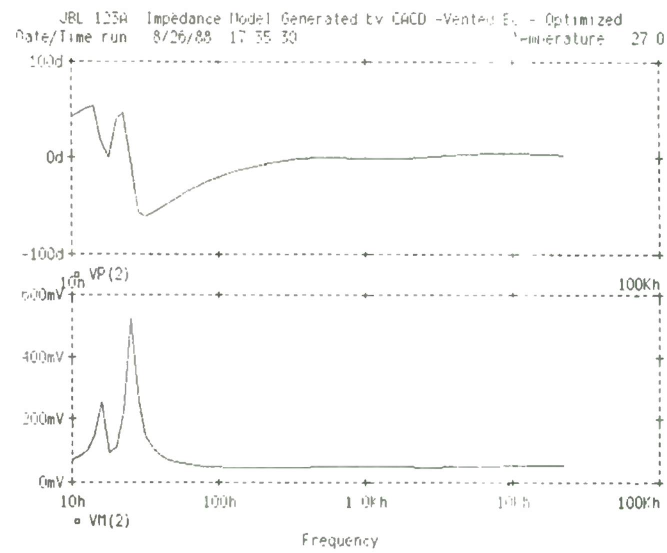


FIGURE 9: Impedance curve of JBL 123A generated from the circuit model of CACD, this time with the optimized components Rc and Cc in the circuit. Notice the flat impedance. Upper curve is phase and lower curve is impedance. Generated by PSPICE.

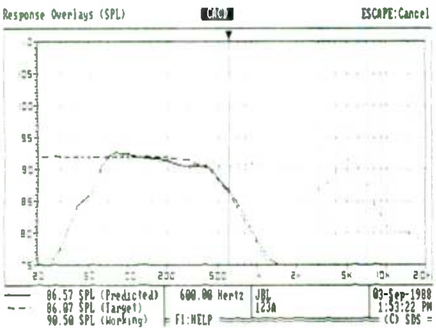


FIGURE 11: Final results of crossover optimization by CACD for JBL 123A. Dashed line is the target transfer function, dotted line is the raw frequency response of the driver and the solid line is the result.

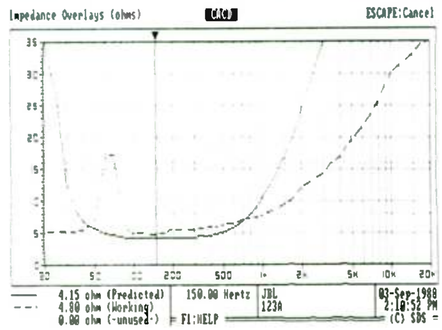


FIGURE 12: Input impedance of JBL 123A/crossover combination. Dashed line is "working impedance" and solid line is the input impedance.

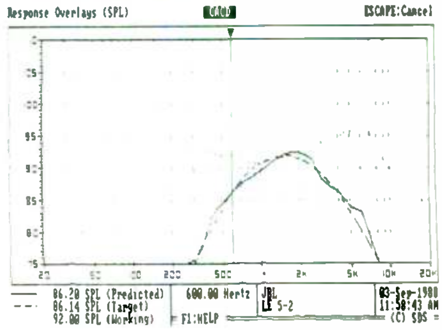


FIGURE 13: Final results of crossover optimization by CACD for JBL LE5-2. Dashed line is the target transfer function, dotted line is the raw frequency response of the driver and the solid line is the result.

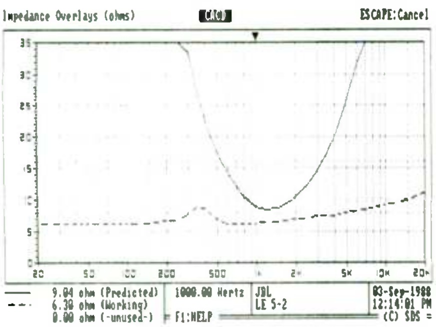


FIGURE 14: Input impedance of JBL LE5-2/crossover combination. Dashed line is "working impedance" and solid line is the input impedance.

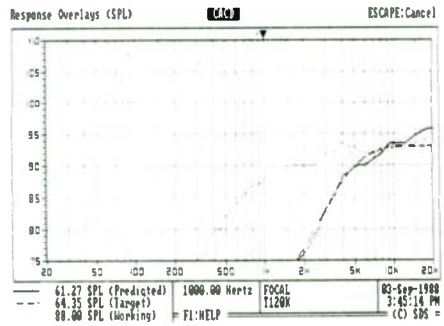


FIGURE 15: Final results of crossover optimization by CACD for Focal T120K. Dashed line is the target transfer function and the solid line is the result.

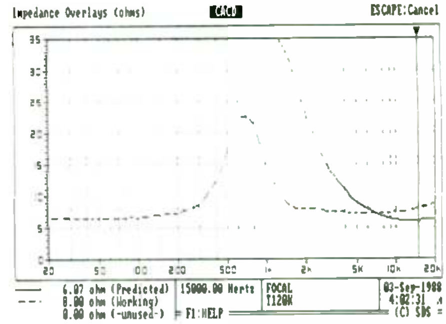


FIGURE 16: Input impedance of Focal T120K/crossover combination. Dashed line is "working impedance" and solid line is the input impedance.

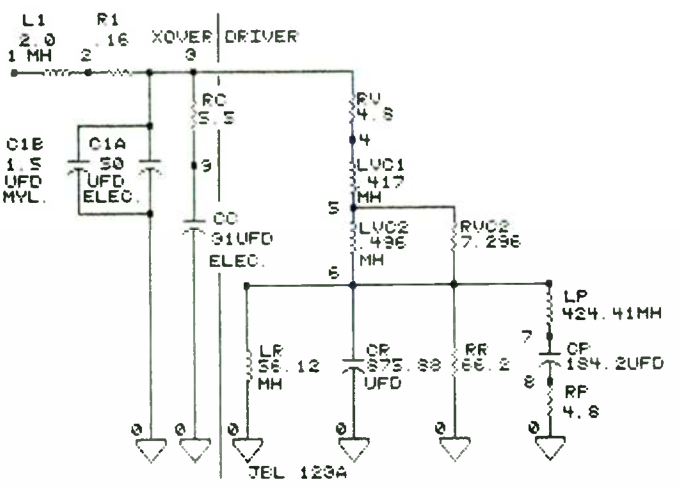


FIGURE 17a: Complete driver/crossover circuit model of JBL 123A and crossover.

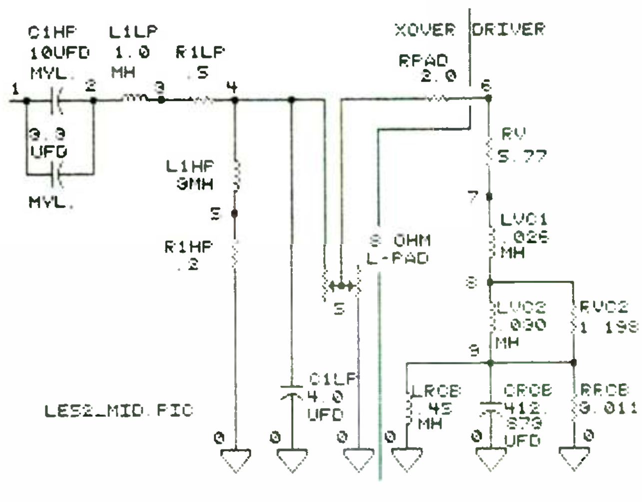


FIGURE 17b: Complete driver/crossover circuit model of JBL LE5-2 and crossover.

over to prepare us for the midrange section of this project. The procedure is the same as for the woofer, and produces *Fig. 13*, which shows the target as a dashed line and the optimization, a solid line. *Figure 14* is a plot of the input impedance of the midrange driver/crossover combination. The target function is a symmetrical second-order bandpass network with −6dB points of 600Hz and 4kHz.

Finally, the tweeter is treated in the same manner. Load the driver, the driver model and the circuit model (STD.HP.2); get starting values; select optimizer target data and calculate target transfer function; optimize, then view the results. *Figure 15* is the response of the Focal T120K with its optimized network and *Fig. 16* is the input impedance.

Most of the time it will not be this simple. The drivers I used are well-behaved and have a wide bandwidth. If the proposed circuit will not generate an acceptable network you must change the topology in the proper way. This is not a big problem. As you use the program, you gain great insight about the placement and starting values of the necessary "fixes."

The optimized driver/crossover circuit model for the woofer, midrange, and tweeter is shown in *Fig. 17*. The actual crossover schematic is shown in *Fig. 18*, with the Parts List. *Photo B* shows the finished and the factory network.

As I was working on this project, I wondered how it might have turned out if I used standard textbook 8Ω crossover components. I "loaded" the optimized driver model and added the crossover components for both the JBL 123A woofer and the JBL LE5-2 midrange, and took a peek at the "Predicted Acoustic Response." *Figures 19* and *20* are the result, with no impedance equalization and standard 8Ω crossover component values. Another curi-

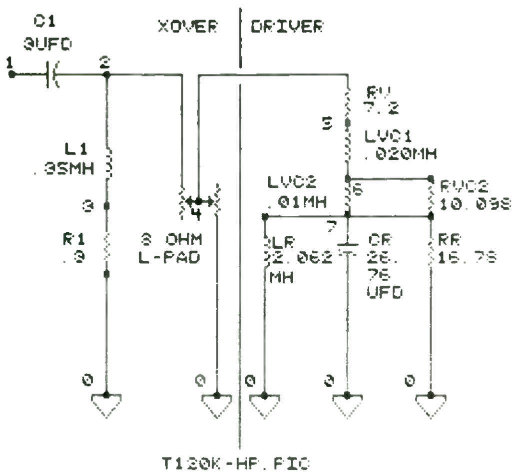


FIGURE 17c: Complete driver/crossover circuit model of Focal T120K and crossover.

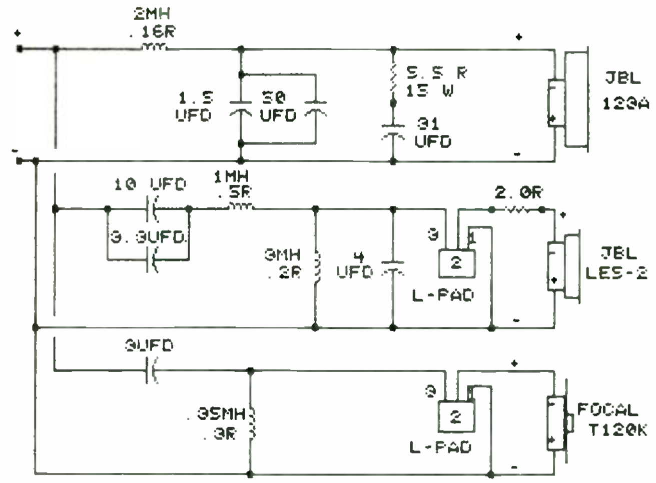


FIGURE 18: Schematic of network installed in revised JBL L100 (see Parts List).

osity, in the accompanying literature JBL claimed a midrange crossover of 1,500Hz. I "loaded" the LE5-2 model and added a 8μF capacitor to the circuit. *Figure 21* shows the raw frequency response, the 1,500Hz 6dB per octave transfer function, and the predicted response. As you can see, the curve is not smooth and would result in a less than accurate speaker.

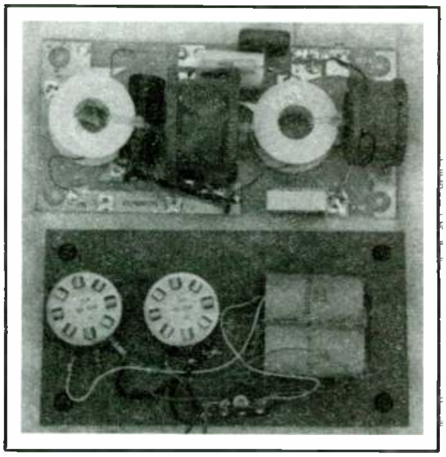


PHOTO B: Crossovers: upper crossover is new CACD generated one; lower one is the original. Notice the inductors turned 90° to one another on the author's new crossover. This reduces crosstalk.

After all this effort, did we improve the sound? The answer is an emphatic "yes." Human "audio memory" is very short, so I listened to an unmodified L100 on one channel and a modified one on the other channel, using my balance control to listen to each separately. The unmodified L100 has a very diffuse, grainy midrange with no stability or depth. The modified version had a rock-solid presence that did not move about with frequency. Vocals are well-defined, drums are tight, and depth is increased. I replaced the stock tweeter with a high performance model purely because of my own taste. It is not mandatory for good results but the tweeter crossover must be re-designed if you wish to keep the existing JBL LE25s. If you own pair of these speakers and are considering this modification, check the midrange suspension on the drivers. Mine were quite stiff due to time and such a high crossover point. I suggest you loosen them by applying a sine wave below their measured resonance for an hour or more, before you measure them. I measured a Q_{ts} of 1.38 before, and a Q_{ts} of 0.85 after.

PARTS LIST

Tweeter

C1	3.0μF Mylar capacitor
L1	.35mH air-core inductor

Midrange

C1HP	10.0μF Mylar capacitor
	3.3μ Mylar capacitor
L1LP	1.0mH air-core inductor
L1HP	3.0mH Sledgehammer inductor
C1LP	4.0μF Mylar capacitor
Rpad	2.0Ω 15W resistor

Woofer

L1	2.0mH Sledgehammer inductor
C1	50.0μF electrolytic capacitor
	1.5μF Mylar capacitor
Rc	5.5Ω 15W resistor
Cc	31.0μF electrolytic capacitor

Number required on all designations: 2

All components available from Madisound Speaker Components, 8608 University Green, Box 4283, Madison, WI 53711, (608) 831-3433.

Although CACD is high-priced for hobbyists, it is an absolute must for all professional loudspeaker designers. It presents a complete and well-written program in the most flexible way possible. Like all power-

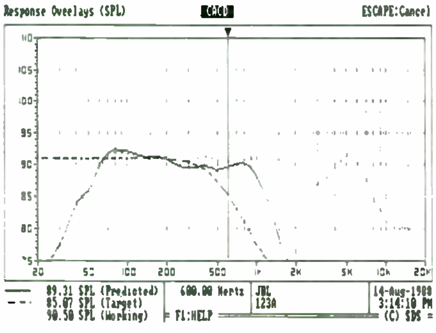


FIGURE 19: Frequency response of a JBL 123A connected to a textbook valued 8Ω crossover.

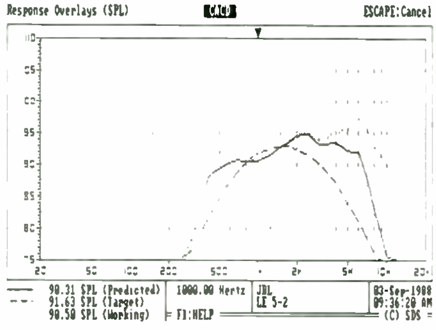


FIGURE 20: Frequency response of a JBL LE5-2 connected to a textbook valued 600/4,000Hz crossover.

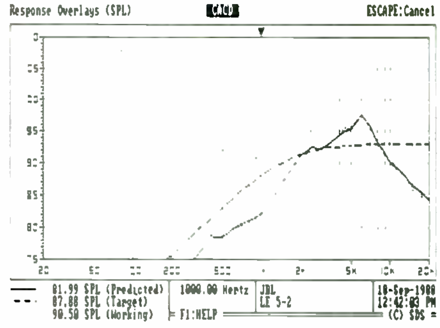


FIGURE 21: Frequency response of JBL LE5-2 when connected to factory 8μF midrange crossover. Dashed line is the target 1,500Hz first-order high-pass crossover.

ful software, you must take some time to learn the procedures.

I have a minor gripe that the starting component values for the optimization are Butterworth, not all-pass, which is the default target transfer function. Another gripe is the whole frequency band, 20Hz-20kHz, is covered by only 46 default frequencies with a maximum of 20 to be used in any optimization. To be fair, I must mention that you can use frequencies other than the default ones, but you have to enter the desired "correction voltage" data yourself rather than have it calculated automatically. A useful feature of this software I did not review is that it will also optimize active crossovers. All in all, I think this is one of the best of this type of software for crossover optimization.

Ted Telesky replies:

I would like to clarify a few points. First Bob Caudle was working on optimization before Peter Schuck's *JAES* article. In fact, we use a different optimization algorithm than Peter.

We do not necessarily assume a minimum phase device for loudspeaker design, though the program presently does not calculate phase. We looked at employing a Hilbert Transform to generate a predicted acoustic phase curve for the driver, based on the driver response curve, but found too many obstacles in implementing this. Instead, since most CACD customers have the ability to measure driver phase at the same time they measure response, a future CACD release will allow incorporation of a phase curve in the driver database, and allow the calculation of a predicted acoustic phase curve for the driver/network/enclosure simulation. Also, once this has been added, we will be releasing a "System Simulator" program which will allow loading multiple circuit files, and entering driver baffle locations, to provide total response and phase data of the "system."

The *System Design Handbook* is available separately for $34.95, and any purchaser who later buys CACD may apply this as a credit. We are now offering a CACD version which will do all the analysis, without optimization, for $199.95. You can add the optimizer later for a $199.95 upgrade.

CACD only supports an Epson compatible printer. We suggest a printer utility such as Pizzaz to allow graph printouts on any printer, as well as allowing the graphs to be scaled for overlays on actual plotted curves.

With respect to storage, any disk directory can hold up to 62,500 driver files, and 250 circuit files, and the number of directories is unlimited, allowing up to a complete hard disk of data to be stored.

The comment on the number of frequencies over the predicted bandwidth is well taken. We also had various opinions and tried a 220-point database (1/24 octave points), but elected to go with 46 for the following reasons. First, very few driver response problems which are narrower than ⅓-octave can be fixed with crossover design, so more resolution might only cause more frustration. Second, since the data points are hand-entered, many might pass out at the keyboard if 220 points per curve had to be entered for every driver curve. However, a future version of CACD will directly read Audio Precision, IQS and Techron TEF data files so, in theory, this may not be a concern anymore.

We also hope to support direct schematic editors such as Tango or OrCAD, so users can draw their networks instead of entering a net list.

Thank you for a very accurate user report and I hope this clarification is useful.

Bob Caudle replies:

I became interested in computer-aided circuit optimization in 1980, based on the work of David Seymour, with an optimization program he was using at Texas Instruments in Dallas; and in 1982 by an AES paper by Adams and Roe of B&W Speakers.

In 1983, after reading about Peter Schuck's program, I knew that optimization on a small computer was practical, and in 1984 I had a working program on the Apple II, LSCD (Loudspeaker Crossover Design). With this program I designed some crossovers for the speaker design group at Tandy. In 1985, I had a version of LSCD running on the IBM PC.

Peter Schuck's work is important because he developed a very efficient routine for optimizing passive crossovers to run on a small computer.

The optimization routine I finally used is known as the BFGS procedure. I tried several, including the Davidon optimization routine Peter described in his Master's thesis. Another popular routine is the DFP (Davidon Fletcher Powell), used in Dean Jensen's Comtran program. The BFGS routine is both computational and memory efficient.

David Seymour showed the complete electrical equivalent of the driver's impedance at a 1980 AES seminar in Dallas. The impedance model in CACD is a simplified version of his complete model.

The reason for modelling the impedance of a

Continued on page 65

REVIEW SOFTWARE

SCHEMATIC MASTER V. 1A
Orivation (David or Pat O'Riva)
2726 Hostetter Rd.
San Jose, CA 95132
(408) 259-2223
Used to generate all schematics. See *Figs. 17* and *18*. (Available from Madisound Bulletin Board (608) 836-9473).

PSPICE
MicroSim Corporation
23175 La Cadena Dr.
Laguna Hills, CA 92653
(714) 770-3022
Used to perform analysis on the models of the drivers generated by CACD. See *Fig. 8* and *Fig. 9*. (Evaluation Version) (Available from Madisound Bulletin Board (608) 836-9473)

PCWRITE V. 2.6
Quicksoft
219 First N #224
Seattle, WA 98109
(206) 282-0452
Used to produce this write-up

PASSIVE CROSSOVERS
Old Colony Sound Lab
PO Box 243
Peterborough, NH 03458
Price: $25
(603) 924-6371
Used to check the accuracy of the components generated under resistive termination.

CASD
Scientific Design Software
PO Box 3248
Chatsworth, CA 91313
Price: $200
(818) 718-1201
Used to check the power handling capabilities of the midrange and woofer (reviewed in *SB* 4/87).

PARAMETER MEASUREMENT MODULE
Scientific Design Software
PO Box 3248
Chatsworth, CA 91313
Price: $100
(818) 718-1201
Generated driver Thiele/Small parameters for midrange and woofer. *Figure 1* is the woofer sample.

POLE-ZERO ANALYZER
Scientific Design Software
PO Box 3248
Chatsworth, CA 91313
Price: $100
(818) 718-1201
Used to investigate transfer functions other than Linkwitz-Riley (all-pass).

Madisound Audio Projects Bulletin Board 300, 1200, 2400, 9600 baud 8-bit, no parity, 1 stop bit: (608) 836-9473)

REVIEW EQUIPMENT

Apple II Plus Computer
Eventide Model AIB232 ⅓-Octave Spectrum Analyzer
AKG C451E microphone with CK22 capsule
Sescom microphone amplifier
B&K 3020—Sweep-Function Generator
Heathkit IM-38—AC Voltmeter
Leader LCR-740—LCR Bridge
WestSide Electronics Pink Noise Generator
Hafler DH-200 Power Amplifier
Hafler DH-100 Preamplifier
Sony CDP-750 Compact Disk Player
IBM compatible personal computer 4.77/8MHz XT, with 8087

Tools, Tips & Techniques

SPEAKER STANDS

I have been listening to and reading ads expounding the merits of speaker stands or feet for the last several years. Decoupling from the floor makes sense if you remember the times when you could hear the bass radiating through the walls in an apartment. The prices for these items range from reasonable to a couple of house payments. As an audio constructor, I desired to build mine and came up with one for the wholesome sum of $5.47. For this project you will need a table saw, plus a drill press.

At a cabinet shop I found a kitchen cabinet door, without handles and with the sides cut back to 45 degrees. I selected a birch front, about 2.5 times larger than the two pieces that I would need. First, measure the depth required for your speakers and cut the wood from each end. My speakers are 15.25 inches wide by 14 inches deep.

To cut the angle on the one side, put the angle against the blade and tilt the blade until the wood lies flat against it. Now move the guide bar over to the flat side, align for a straight cut, and tighten the lock down. Turn the wood around, place the angle side against the guide bar, and now you can cut an exact copy of the angle. Mark the width and cut from the back to front. You now have two pieces with angles on three sides and a flat side for the back.

The "feet" are ¾-inch carriage bolts, 3-inch length. From each of the eight corners, measure in 1.75 inches and drill the four ¼-inch holes for the bolts. You should countersink the holes ⅛-inch deep on the top side, for the square shank of the bolts.

I ground the ends of the bolts to a moderately sharp point to penetrate the carpet for balance and isolation from the floor. One last step is to put a small rubber stick-on foot in each corner of the speaker, this will give you a gap between the speaker and stand. The stands look almost like part of the speaker; the 45° angles tend to reduce the frontal area so they appear as the bottom of the speaker. With the 3-inch bolts, the stand sits just above the top of the carpet.

The stands have increased my speaker system's mid to lower bass level, but I think perhaps the most noticable improvement is the additional clarity in the mid to lower midrange. Try building a pair, I think you will like the difference.

Robert Lewis
Portland, OR 97233

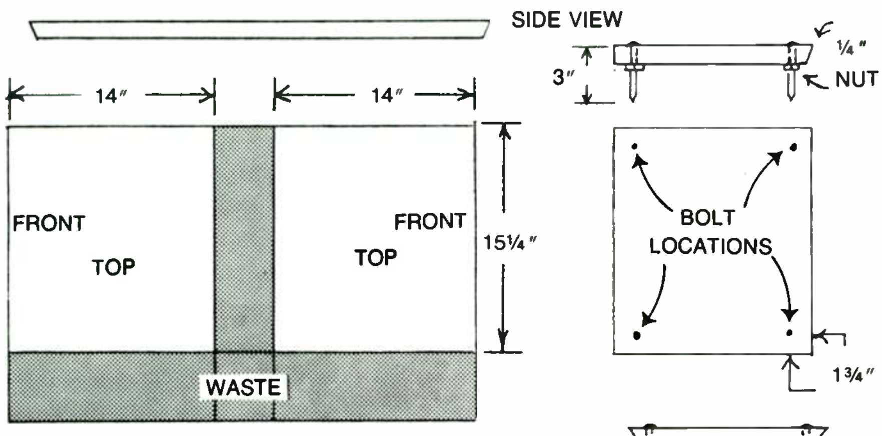


FIGURE 1: Speaker stand construction.

THE PERFECT HOLE

PVC or ABS pipe is available in many diameters and is commonly used for vents in reflex-box construction. Anyone who has tried to cut circular holes in particle board knows how difficult it is to get a clean, uniform hole to insert the vent tight and flush. After some experimenting, I found a dandy way to make a port perfectly airtight.

After cutting the hole as neatly as possible, lay the baffle face down on several layers of newspapers. Insert the vent tube, flush to the front. All the inevitable little gaps you see must be filled. I use industrial type two-component epoxy that comes prepackaged in dual tube modules. These drop into a squeeze gun, and a long nozzle with a mixing spiral attaches at the business end. The epoxy comes out thoroughly mixed in the correct proportion and can be applied from the nozzle just like caulking.

This stuff runs freely and will readily seep into the gaps. The newspaper dams the flow underneath. The cut edge of the particle board soaks up a little, and you need to "needle" the joint gaps to coax out any bubbles. Then, apply more until the gaps are filled completely. When the epoxy is dry, you just peel off the paper and sand the front of the baffle, tube, and epoxy until all is flush and smooth (I use a medium grit at first).

Epoxy does not bond PVC or ABS. In fact, you can even force the tube out of the hole. But the epoxy does not appear to shrink when setting, and this slip-joint is very tight. Apply adhesive caulk liberally around the joint on the "inside" of the baffle. This is sufficient for permanently affixing short lengths of Schedule 40. Weightier tubes could be supported with angle brackets or some other, more mechanically stable arrangement.

With a couple of coats of sanding sealer and an enamel paint job, this joint can be finished so cleanly it is virtually invisible.

Paul W. Graham
Independence, MO 64050

WAVE REFLECTORS

To reduce standing waves, most builders resort to some form of damping material on the inner cabinet faces. Unfortunately, this can have unpredictable results on V_B and is only partly effective.

The other alternative is to build complex cabinet shapes, which again are only partly effective, tend to be larger and are often more expensive for the majority of projects.

Try this for a cheap, quick solution. Obtain some large bore (200-300mm) cardboard or plastic tubing, slit it lengthwise and glue the resulting "reflectors" on the inner panel faces.

Start with one large reflector behind the woofer and then position one for the top or bottom and another for one side. I think my diagram shows the idea clearly. If you have the facility, bevel cut them by 5 or 10 degrees.

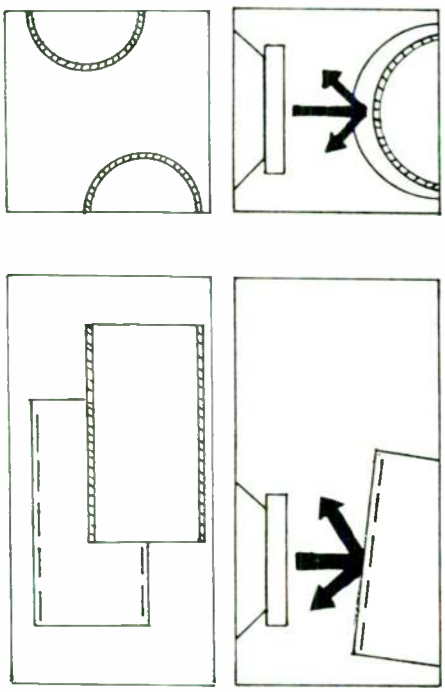


FIGURE 1: Typical reflector placement (top and side views).

Don't forget, unless your speaker is adequately braced, trying to stop standing waves is pointless. Therefore, don't sacrifice brace positioning to accommodate the reflectors, just notch them where required.

This idea will reduce V_B by a completely predictable (small) amount. A typical installation will subtract only about 0.5 liters from V_B and is therefore ideal for improving existing speakers.

Ron Poelman
Townsville 4810
Australia

FOAM RUBBER SOUND ABSORBERS

An inexpensive alternative to some of the absorptive foam rubber products on the market came to mind when I saw a hospital product called an "egg crate" mattress. These mattresses come in yellow foam rubber and are excellent absorbers of sound above 500Hz. At $8 per 38 by 96 by 2" sheet, they are inexpensive enough to experiment with.

You could easily set up a "live end, dead end" room with this foam, though the textured yellow might look a little strange. I have used this material to line TLs and ported systems with excellent results, and appearances inside a box are not important.

Source: Any hospital supply store, or: Mill End Stores, 168 E. Huron Ave., Bad Axe, MI 48413.

Richard Painter
Ubly, MI 48475

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

Technology Watch

Subwoofers, Part II

By Peter Muxlow

As a followup to last issue's column on subwoofers, I thought it would be interesting to examine European Patent #0267650. In it, Bob Carver describes an ingenious method of optimizing moving coil drivers for dipole operation at low frequencies. In a dipole, the baffle-induced rolloff begins when the average baffle dimensions equal half the wavelength (see *Fig. 1*—point A of curve 1).

This fall off is at a rate of 6dB/octave and is caused by the front-to-back cancellation of the driver output. To equalize this baffle rolloff, Carver uses drivers with a high Q, which gives a rising response, as shown in *Fig. 1* (curve 2). The resultant curve is the equalized system response.

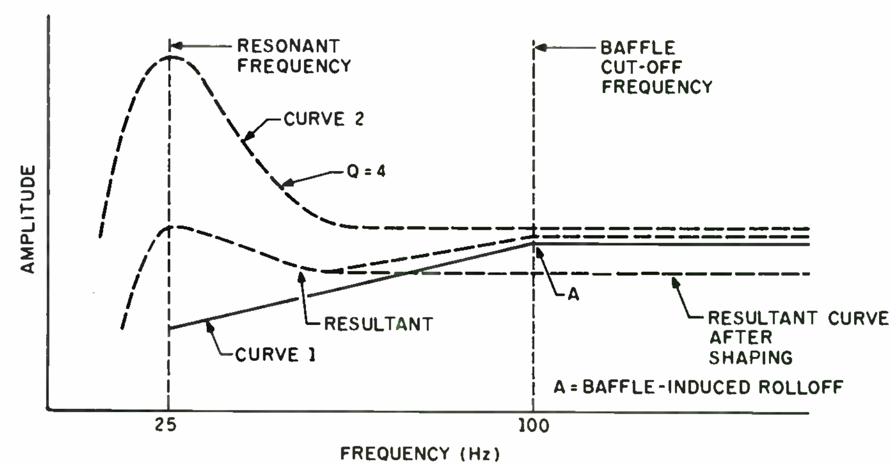


FIGURE 1: Frequency response curves.

Additional methods of fine tuning to get the required flat frequency response are mentioned in the patent; for example, stagger-tuning the resonant frequency of the drivers, or shaping the effect of the Q through the crossover network. As a bonus, the high-Q drivers used are cheaper, because they use smaller magnets, since the driver Q is inversely related to the magnet size.

Now let's take a look at the efficiency. For a closed box:

$$n\ (0.5 \times \text{space}) = (16 \times 10^{-12}) \times (F_0^3 V_{as}/Q_e)$$

where,

F_0 = driver free air resonance;
V_{as} = air volume equivalent to the driver's suspension compilance;
Q_e = the driver's electrical Q value.

For a Carver dipole:

$$n = F_0^3 A/Q_e K$$

where,

A = area of diaphragm;
K = driver's suspension stiffness or compliance.

The important point here is the box's volume, as in the sealed-box equation, is replaced by the ratio of the drivers' cone area to the drivers' suspension compliance. Both these parameters are a function of the driver design only. The efficiency is not related to the size of the dipole baffle, unlike the closed box where the efficiency is related to the volume of the box.

Carver's next trick is to redefine the efficiency equation another way. If:

$$F_0 = \tfrac{1}{2}\pi - \sqrt{K/M}$$

where,

M = the total moving mass of the diaphragm and those components that move with it.

If you substitute this in the dipole efficiency equation, then:

$$n = F_0 A/Q_e M$$

This equation shows the relationship between the low-frequency cutoff and the efficiency, which now becomes linear instead of cubic. To maximize the efficiency, the drivers' moving mass should be as small as possible, consistent with the practical maximum radiating area. Carver suggests a cone area-to-mass ratio of at least 2:1.

Typical results of drivers quoted in Carver's patent are: resonant frequency, 23.4Hz; Q, 4.12; total moving mass, 34.4g; area of diaphragm, 75 in.^{2}.

In Carver's prototype he uses four 12-inch woofers with a cone excursion of 1.5 inches. The frequency response is flat to 26Hz.

MADISOUND SPEAKER COMPONENTS
8608 UNIVERSITY GREEN
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PHONE (608) 831-3433

PERFECT LAY WINDING Air Core Inductors

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

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

Induct	Resistance		Size	Price	.91mh	.20Ω	.75	3.0	$8.90	2.75mh	.39Ω	.88	3.5	$16.85
L-MH	DC Ω	Ht	Diam.	Each	1.0	.21	.75	3.0	9.35	3.0	.42	.88	3.5	17.60
.22	.08	.56	2.25	$4.10	1.1	.23	.75	3.0	9.90	3.3	.45	1.0	4.0	18.50
.33	.10	.63	2.5	5.05	1.2	.26	.75	3.0	10.80	3.7	49	1.0	4.0	19.25
.47	.13	.63	2.5	6.25	1.3	.27	.75	3.0	11.55	4.0	.50	1.0	4.0	20.15
.56	.15	.63	2.5	7.05	1.5	.28	.75	3.0	12.10	4.5	.56	1.0	4.0	21.55
.62	.16	.63	2.5	7.50	1.8	30	.88	3.5	13.65	5.0	.59	1.0	4.0	24.20
.68	.17	.75	3.0	7.80	2.0	.31	.88	3.5	14.30	5.5	.63	1.0	4.0	25.75
.75	.18	.75	3.0	8.15	2.25	.33	.88	3.5	15.20					
.82	.19	.75	3.0	8.60	2.5	.36	.88	3.5	16.05					

Values between sizes listed are also available. Add 10% to cost of value larger than your requirement.
Madisound stocks **audio standard** inductors as well as the popular **SIDEWINDER** and **SLEDGEHAMMER** inductors.

CHATEAUROUX POLYPROPYLENE CAPACITORS

We are pleased to announce that we now have in stock *CHATEAUROUX* METALLIZED POLYPROPYLENE CAPACITORS of exceptional quality and excellent price. This type of dielectric has been characterized by Walter Jung and Richard Marsh as "outstanding" when compared with all other dielectrics in the areas of:

DISIPATION FACTOR PERCENTAGE—DIELECTRIC ABSORPTION PERCENTAGE—STABILITY

- Low dissipation factor
- Low dielectric absorption factor
- High insulation resistance
- High frequency and temperature stability
- Good self healing characteristics of polypropylene dialectric
- High Current Capacity
- Excellent Overvoltage and Pulse handling capability
- Low self inductance
- Low equivalent series resistance
- Excellent stability
- Leads: Tinned pure copper multistrand insulated sleeves, radial, 4 inches.
- Superior high frequency characteristics
- High Ionization level
- Dissipation factor @ 20o C: Less than .01 %
- Dielectric absorption factor at 20o C: Less than .01%
- Insulation resistance @ 20o C:More than 100 K megohms/mfd
- Temperature range: -25o C to+85o C
- Dielectric: Polypropylene Film.
- Working Voltage: 250 VDC or higher.
- Test Voltage: 2.15 times rated voltage
- Capacitance tolerance: +/- 10%: (grading to 1% available)

Dimensions in inches:	12mfd:1.3D,2.6L	50 mfd: 1.38 D, 3.4 L	120 mfd: 1.9 D, 4.4 L
2 mfd: .83 D, 1.9 L	15 mfd: 1.4 D, 2.6 L	70 mfd: 1.62 D, 3.4 L	150 mfd: 2.1 D, 4.4 L
4 mfd: .98 D, 1.9 L	25 mfd: 1.6 D, 2.6 L	80 mfd: 1.75 D, 3.4 L	180 mfd: 2.2 D, 4.4 L
8 mfd: .98 D, 2.6 L	35 mfd: 1.4 D, 2.6 L	100 mfd:1.70 D, 4.4 L	200 mfd: 2.4 D, 4.4 L

Audiophile prices		5.0mfd	$2.75	25.0mfd	$7.55	100.0*	$23.75
1.0 mfd	$1.85	5.6*	3.05	30.0	8.90	120.0*	29.00
1.5	2.05	6.0*	3.05	35.0	10.00	150.0*	34.50
2.0	2.05	7.0	3.25	40.0*	11.10	200.0*	43.50
2.5*	2.15	8.0*	3.50	45.0*	12.20		
3.0*	2.15	10.0*	4.20	50.0*	13.25		
3.3*	2.50	12.0	4.30	60.0*	15.45		
4.0*	2.50	15.0	5.30	70.0*	17.60		
4.7	2.75	20.0	6.80	80.0*	19.75		

Pairs matched to within 1% of each other: add 10%. 10 pieces or more of the same value: deduct 10%.

*axial leads

Audiophile Price List

Fast Reply #ED182

SB Mailbox

POWER TO LM12

I enjoyed Mr. Newcomb's article, "Sub-Bass Power Boosting" (*SB* 4/88) very much. My specific interest is in using the LM12 in the bridged mode with my current tuner/preamplifier. I am not familiar with the operation and design of op amp circuits, and I have several questions. First, have Mr. Newcomb's circuit boards been produced in any quantity and are they for sale?

What is the recommended power supply for both the bridged and non-bridged version of the amplifier—that is, the required voltage, current and amount of regulation?

What is the recommended input voltage swing to this amplifier? I specifically wonder whether I need another stage of amplification between my preamp and this amplifier.

My intended application is to use these amplifiers to drive the Swan IV speakers described in the same issue. I would have one 200W bridged amplifier for each cabinet for a total of four amplifiers.

Thank you for an interesting article and your time.

Raymond Bahr
Holliston, MA 01746

Art Newcomb replies:

I am glad you find my article useful. I am not prepared to offer the circuit board in question for sale, but I suppose Old Colony may do so. I have sent an updated layout to *SB* which includes a correction to this board (published in issue 6/88). I am currently using the differential configuration in my main amplifier with much success (three 200W amplifiers, stereo plus a subwoofer). Be sure the power supply bypass capacitors (Cps) use the separate return to your single point ground to eliminate noisy loops. These capacitors reduce high frequency conversation between amplifiers.

In all configurations of the LM12, I use ±30V unregulated supplies; the LM12's high rejection permits this, although lower noise can be realized with regulation. Power supply current capacity will depend on speaker impedance—to swing ±25V with a single LM12 at 8Ω you will need just over 3A per unit (plus the idling 60–70mA each); for the 50V capability of the differential unit you'll need +6A. Regulation ensures no power supply sag will reduce loss of power with heavy demand (my unregulated units have only a peak RMS power capability due to sag).

The voltage gain of the single unit in the article was 1 + 47k/2.2k = 22.4 (1 + Rf/Ri) or about 27dB. This will give full output with about 1.2V peak at the input. The differential unit has twice the gain with the same values. You may even want to reduce the gain in this case, depending on existing gains in your setup.

Good luck!

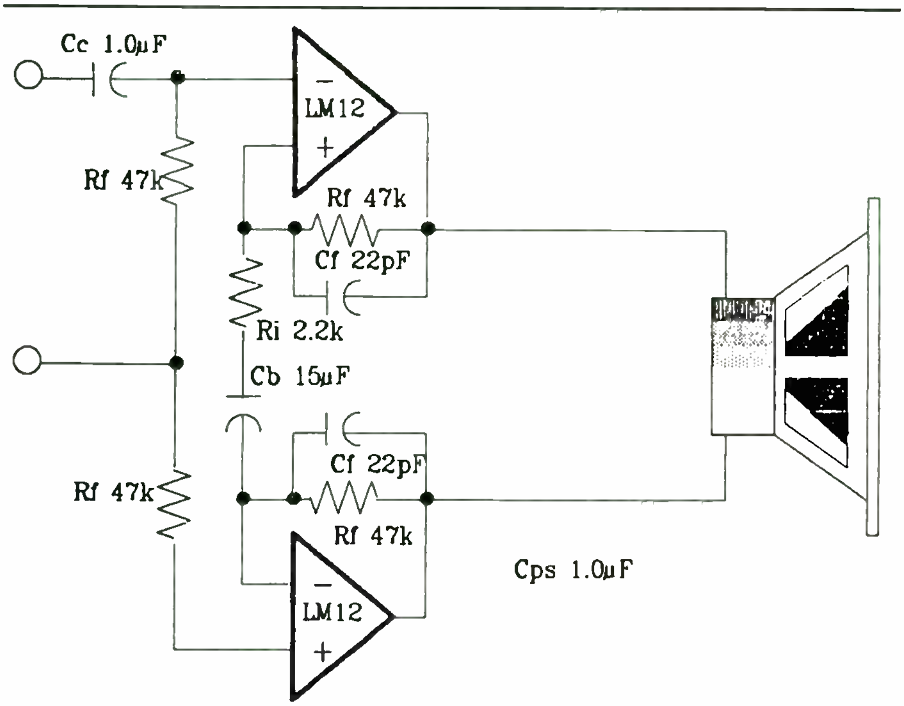

FIGURE 1: Differential configuration for an LM12 amp.

[If readers express sufficient interest, Old Colony will offer boards for author Newcomb's project. Please use Fast Reply #992 to indicate interest in the boards.—Ed.]

ESL RESPONSE

In the correspondence between Wagner and Sanders (*SB* 3/88, pp. 60–65), Beveridge's patent #3,688,335 is cited as an ESL operating in a non-*constant charge* mode. Examination of the patent shows the electrodes are of a high resistivity material—carbon loaded epoxy; volume resistivity 10^8 to $10^{11}\Omega$ centimeters. The diaphragm is Mylar, with a coating of vacuum deposited aluminum—a low resistivity layer.

Even though this is the reverse to the normal ESL configuration, its operation is still in a constant charge mode.

Also discussed was the size of electrode perforations. The size of this hole area has other important characteristics not mentioned. The *space-to-hole ratio* of the electrode determines the acoustic impedance the diaphragm sees. The size of the hole and the thickness of the electrode contain a "lump of air" which has its own inertia. This adds mass to the mass of the diaphragm which results in a lower high-frequency cutoff. Also, the electrode space-to-hole ratio has the added function of applying a necessary resistive load to the diaphragm. Because the Mylar diaphragm has a mass which is not high compared to air, the diaphragm's resonances need to be resistively damped. The equation for the acoustic impedance of a perforated plate can be found in *Acoustics*, by Beranek, pp. 138–9.

Peter Muxlow
Wellington, New Zealand

RESPONSE STEP ROUTINE

In reading *SB* I note that you have to synthesize your crossover designing routine. For instance, most articles never mention the "response step." So, you have to experiment in eliminating it and crossing over suitably.

I'm working on my first commercial speakers, and so I want to use few crossover parts and accomplish my response goal of compensating the response step at 30° off axis, have great side lobes, and good phase response.

Perhaps some of my experiences will help other readers.

I am aiming to make the woofer and tweeter roll off at the same rate for an octave either side of FC, since this seems to give an even response from many angles. This can be done by using different (asymmetrical) electrical slopes on the woofer and tweeter; the phase must match as in LMP.

First, I take the woofer and compute the response step, as in the LMP articles, according to the box width. The step may be only 4dB at 30°, caused by the mids beaming. Then I check to see whether one coil will compensate the step and give a usable rolloff. I start with a coil for a −3dB halfway up the step, at the nominal woofer impedance. I do not use a Zobel so the coil's effect is around 2.5dB/octave due to rising woofer impedance. I try smaller and larger coils to find one which compensates the step and rolls off the woofer, probably at its natural rolloff point.

If the FC is too high, or the slope too gradual, a two-pole crossover may exist which will do it. Try a coil and cap for a Linkwitz-Riley at .9 the step's top frequency, and play around. You may get a usable response with an FC around 1.8 the step's top.

Add a Zobel and you can take the step, and cross over *at* its top. With the −3dB point of a Linkwitz-Riley at the midpoint of the step or just above, the FC will occur at the top of the step. (L-R gives a 6dB cut in the first octave). Measuring the woofer's rolloff (with dB meter and sine waves) you can get its dB/octave in the first octave past FC.

Now you need a tweeter whose rolloff below FC in the first octave, plus an appropriate crossover, equals the woofer's in its first octave of rolloff. You can vary a two-pole crossover between 6 and 12dB in the first octave to accomplish this. For 12dB use 6dB *single-pole* formulas. For 6dB, use 2 × mH and 0.5 × μF.

You must check it out on LMP or add the phase degrees by using the article's charts and formulas (woofer + low-pass + tweeter + high-pass + woofer/tweeter offset). They should total near 360°.

Ed Light
Los Angeles, CA 90064

V_{AS} CLARIFIED

In *SB* 6/88, Scott Ellis writes ("Push-Pull Design," Mailbox, p. 73), "As far as the 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."

Readers should consult his article, "The Curvilinear Vertical Array" (*SB* 2/85), to appreciate that he refers to parallel bass loading; not to series bass loading, as is the case with *compounded* drivers, for which the V_{as} is halved. I think this clarification is in order so no one is misled by the statement out of context.

Paul W. Graham
Independence, MO 64050

ROCK REBUTTAL

Thank you for David Davenport's objective review of the L.A.W. "Nonresonant Loudspeaker" (*SB* 4/87, p. 35). In L.A. White's reply to that review, he made some statements I must object to.

First of all, Mr. White seems to equate full bass response with undesirable resonance. ("No way these speakers can be made to have a strong resonance in the bass such as Mr. Davenport seems to prefer.") This, of course, is not the case. The goal of any low-frequency design is *accurate* bass response that is uncolored by improperly controlled resonances. Rock and electric music, which Mr. White referred to, simply have a much larger bass content. If the L.A.W. speakers can't reproduce this, they should be considered inaccurate.

This complete lack of bass is, as Mr. Davenport said, not at all surprising. After all, we're looking at a woofer that's practically in free air. Estimating dimensions from the photo, I think any frequencies below about 240Hz will neatly cancel. I thought the function of a speaker cabinet was to eliminate this cancellation?

An open speaker like this will all but eliminate the low and mid bass. If these speakers are "nonresonant" it is only because they sacrifice almost 3 octaves of bass.

Mr. White makes the proposal of checking for resonances by comparing impedance curves with the woofer in free air and in the cabinet. This test is certainly useful, but it is a far cry from an exhaustive analysis of the system's low frequency behavior. Here again, I see Mr. White equating bass response to resonance. He seems to use the word "resonance" as a blanket term which encompasses all speaker evils and shortcomings. Resonant characteristics are, of course, only one of several major factors contributing to a loudspeaker's performance.

Secondly, Mr. White makes several slams at rock and roll music which I, as a rock/fusion enthusiast, cannot let pass without comment.

1. Modern rock, especially with the trend toward synthesizer-based music, presents no less of a challenge to a loudspeaker than classical and orchestral music. In some cases, it may present more. The whole issue boils down to a question of production quality, which is admittedly under par for some rock recordings.

2. "Plugging the electric instruments directly into the recorders" is a common practice, but an equally common practice is to mike into recorders from amps in the studio. A less common practice I have heard of is to do both and mix them later. These decisions depend on the "sound" that is most desired.

3. Musical instrument/P.A. speakers have different requirements than hi-fi units. They must be able to: a) respond strongly to a limited bandwidth; b) handle a lot of "dirty" (distorted) power; and c) withstand physical, electrical, and environmental abuse; and yes, they have to pump plenty of decibels. The distortion you hear coming out of guitar and bass amps isn't caused by the amp's speaker. It is purposely introduced by the amp's electronics before it ever sees the speaker. *This* distortion gives rock its characteristic sound. Some artists have spent a lot of time trying to get that "perfect" distortion effect for a song. When they finally get it, you will not accurately reproduce it by playing it through cheap speakers.

I get the impression that L.A. White's prejudice against certain forms of music are stronger than David Davenport's "prejudices" against unorthodox speaker designs.

Keep it up *SB*! You have a good, unique magazine. Same goes for *TAA*.

Dale S. Oglesby
Rose Hill, KS 67133

SWAN AMP?

Congratulations to authors D'Appolito and Bock on their excellent article (*SB* 4, 5/88). The Swan IV speakers "sound" so good on paper that I am going to build a pair for my system.

I have several 4 by 8-ft. by ¾-inch sheets of oak plywood. While I realize the material is not as dense as particle board, would I realize decent is not similar results—assuming the interior dimensions are the same? I can make a "Chinese copy" of the veneer/particle board enclosures, but, as usual, I'm looking for a shortcut.

What minimum power amp rating would the authors recommend for the bi-amp process for each cabinet? I currently have an all Carver system including the M1.5t amp; which of their amps would best be suited (my listening room is about 17 by 25 feet)?

I have the facilities to reproduce printed circuit boards. Is there a full-size printed circuit layout for the Pedal Coupler?

Also, the article (*SB* 5/88) mentioned the bass drivers are not currently available. Any recommendations on replacements for any of the three drivers?

Any assistance you can give me would be greatly appreciated.

Robert W. Schmidt
Eagle River, AK 99577

Contributing Editor D'Appolito replies:

I would use the plywood. Your biggest problem will be with the bass modules. The large panels will ring badly. Build these boxes to the correct external dimensions; then tap the panels to find the areas of strong vibration. Add snug fitting cross braces at these points and continue the process until you are happy with the result. Place the braces from side to side, or from front to back, to place the opposing panels in tension and tie them together. This approach is far more effective than trying to stiffen individual panels. You may also wish to add damping compound or bituminous felt to problem areas.

I am not familiar with the Carver amps, but in rooms of 3,000–5,000 ft.^{3} we like to use 200W per channel for both the bass modules and the satellites.

Remember, the bass modules are 4Ω, while the satellites are 8Ω.

A printed circuit layout is not available for the Pedal Coupler. I redesigned the circuit to accommodate the current drivers (Eclipse 10W38 10-inch woofer available from Madisound or Meniscus) I now recommend for the bass modules. The revised crossover is available from Swan's Speaker Systems, made for them by Ace Audio.

SWAN IV CROSSOVER

I very much appreciated the "Swan IV" article. As I read it, the unfiltered acoustic response of the treble driver leads the midrange driver pair by 90°. The authors point out if these drivers are filtered by networks that produce a conventional third-order acoustic response (−3dB, ±135° at the crossover frequency) the 90° tweeter phase lead will reduce in-phase acoustic output whose summation is up by 3dB.

The authors' solution to this problem is to spread the crossover frequencies, so they are down by 5.7dB at the frequency where their characteristics cross. Because the individual crossover frequencies have been spread, the individual phase angles will no longer be ±135° and so, the 90° treble driver phase lead can no longer produce in-phase output, yet the authors say in-phase acoustic output results. Did they mean to say "near acoustic output?"

As a possible alternative, have the authors considered filtering the drivers through networks that produce −6dB, ±135° at a common frequency? If practical, this, in combination with the 90° tweeter lead, would produce exact in-phase acoustic output and a perfectly flat summation.

Is it correct that the low-pass filter is a third-order, constant resistance network whose load resistance is 12.91Ω, and whose crossover frequency is 2,054.6Hz? Is it correct that the high-pass filter is a second-order, all-pass network whose load resistance is 5.55Ω, and whose crossover frequency is 2,560.4Hz?

David Meraner
Scotia, NY 12302

Contributing Editor Joe D'Appolito replies:

The filter you suggest sums to one at the crossover frequency, but is otherwise not perfectly flat. It has about 0.3dB ripple in the octaves either side of crossover. No known third-order in-phase network sums to flat magnitude response. The closest I've been able to come to it via computer search is one that sums flat to within a ripple of .22dB. This is the response we aimed for in the Swan IV satellite. We obtained this response by making use of the large frequency range over which the mid-bass drivers and tweeter are approximately 90° apart, in-phase.

The design was complicated because the high-pass portion of the Treble Coupler also had to compensate for the response characteristics of the D-28. The important property of a third-order network is not that it has $\pm 135°$ phase shift at the crossover frequency, but that it produces an interdriver phase difference of 90° at all frequencies. This property, combined with the opposing 90° phase difference between drivers, leads to an in-phase condition throughout the entire crossover region. Third-order networks alone do not produce in-phase driver combinations.

From *Fig. 7* of the Swan IV article (*SB* 4/88), you will see the acoustic response curves of the mid-bass and tweeter drivers are essentially those of an in-phase third-order crossover network. However, because driver responses are nonideal and driver input impedances are complex, the electrical crossover networks which produce the final desired acoustic responses are not obvious. Final selection of the crossover components was accomplished using an iterative process of computer-based optimization and direct measurement. The mid-bass drivers are relatively flat to 5kHz, so the starting point for the low-pass portion of the Treble Coupler was a standard constant-resistance third-order Butterworth network at 2kHz with a load impedance of 12.5Ω. Only the Zobel components, R1 and C2, were allowed to vary during the optimization process to control the damping around the crossover. The high-pass filter is also third-order, but it has no relationship to any of the standard filters. All components in the high-pass section were allowed to vary. R2 causes the high-pass filter transition to second-order at low frequencies.

The input impedance and acoustic response of all drivers were carefully measured, and fed to a modified circuit analysis computer program. We conducted an iterative search until we obtained a response with only 0.3dB ripple across the range from 1–4kHz. I built and tested a crossover network with the computer-derived values. We did only minor tweaking of these values to obtain a measured response with the same ripple. The resulting third-order high-pass filter has a characteristic frequency of about 2,800Hz, but clearly is not any standard alignment.

The driver responses must sum properly over the entire crossover region and not just at the crossover frequency alone. Making the summation perfect at the crossover frequency alone usually leads to larger ripple across the band; this is the nature of mathematical optimization. When dealing with real-world drivers, the standard constant-resistance crossovers rarely provide flat frequency response. Computer optimization allows us to quickly zero in on a good real-world design. The computer doesn't know a Butterworth from a Linkwitz. It just grinds away until it finds the Rs, Ls and Cs which optimize the design criterion. The resulting networks often produce what would be considered overlapping, or in our case, underlapping crossover frequencies by conventional criteria, but it is hard to criticize the results. In addition to almost flat amplitude response, satellite interdiver phase differences are no more than $\pm 4°$ over the same 1–4kHz range. As a result, narrow-band pulse response is essentially perfect. I know of no real-world system that has ruler-flat frequency response with all drivers exactly in-phase. I believe, however, that we have come as close as practically possible to this desired goal in the Swan IV.

A VIFA DRIVER ARRAY

As an avid reader of *Speaker Builder*, beginning in 1986, with back issues to 1983, and as a home builder of speaker systems since 1980, I have made some progress in my design realizations. The speaker systems I constructed have improved both sonically and in appearance, thanks in great part to *SB* and its contributors.

The most successful realization of my efforts, according to family and friends, is a three-way sealed-box design with a 10-inch Vifa poly woofer, a 4-inch Vifa poly midrange, and a 1-inch Vifa dome tweeter. This system employs an all odd-order passive crossover network, 650Hz and 3kHz. This realization is a result of listening to the Boston Acoustics A150, along with several other good commercial designs, while helping an acquaintance select a pair of replacement speakers.

My most recent design and construction project is a three-way, four driver tower including a 10-inch Vifa woofer in a fourth-order alignment, two 4-inch Vifa midranges and a Vifa H25TG horn tweeter. The midrange tweeter array is arranged in D'Appolito's symmetrical fashion ("A High Power Satellite Speaker," *SB* 4/84). My crossover network is passive with the crossover points at 400Hz and 3kHz, all odd-order. The slopes on the low side of the tweeter as well as the low side of the midrange are 18dB/octave. The results of this design and construction were good, however, the outcome was less than expected.

I am planning a new construction. It is a five-driver, three-way tower design including two 10-inch Madisound woofers in a second-order alignment, two 4-inch Vifa midranges and the Vifa H25TG horn tweeter. The mid-high array will once again be a symmetrical arrangement, with an all odd-order crossover network having crossover points at 650Hz and 3kHz, including the same steep midrange and tweeter slopes I employed in earlier designs. In addition, I plan to roll off the lower woofer at 125Hz.

On page 12 of the Swan article (*SB* 4/88), authors D'Appolito and Bock indicate (as not presented in the 4/84 feature) that if the Dynaudio D-28 (similar to the Vifa HT25G) is connected in a conventional 18dB/octave arrangement with the mid/woofers, the combined response will be up 3dB at the crossover point; a result of the D-28 horn type-tweeter having the same phase as the mid/woofer driver pair. The authors suggest that by spreading the high/mid crossover by a factor of 1.4, the combined array response will be nearly flat at the crossover point.

For my project, can I use Vifa's data to spread the crossover point of 3kHz by a factor of 1.4?

Would the voice coil of the HT25G horn tweeter, in close alignment with the voice coils of the midranges, have any effect on the phase relationship of the driver array, thus affecting the array's combined response?

Mr. D'Appolito (*SB* 4/84) also indicates the importance of an interdriver spacing of two-thirds of the wavelength at the crossover point between the tweeter and the mid/woofer pair. At my selected crossover point of 3kHz, two-thirds of the wavelength is about three inches—difficult to attain with my driver's flange OD of about four inches. Is this interdriver spacing critical?

Also, in response to *SB* readers' requests for more cookbook designs: learning the principles and practical theories of speaker design, and their application and construction, results in the growth of knowledge and experience. Great chefs did not attain their skill via Betty Crocker!

John D. Williams
Rochester Hills, MI 48309

Contributing Editor Joe D'Appolito replies:

I obtained all the data used in the design of my satellites by direct electrical and acoustical measurements on the drivers in their intended geometry. Manufacturers' curves simply do not provide enough data for a thorough design. Most importantly, phase behavior and driver acoustic center location are not provided. Once mounted on a baffle, however, we can measure frequency response, interdriver phase differences and acoustic center offsets to design an appropriate crossover. Although the Vifa combinations you plan to use look quite promising, I can give you very little specific advice without this data. My reply to Dave Meraner's letter (this issue) explains in more detail how the Treble Coupler design was developed.

Turning to some of your specific questions: despite the longstanding folklore, voice coil location is a poor indicator of driver acoustic center. For large woofers, the mean acoustic center is typically 15 to 20cm behind the plane of the voice coil. For mid-range drivers this figure drops to 2–3cm. For a tweeter, the physical and acoustic center offsets are generally less than 1cm.

There is nothing magical about the driver spacing. My geometry works best when the interdriver spacing is less than one wavelength at the crossover frequency. I do not recommend spacings greater than two thirds of a wavelength. Wider spacings will lead to a narrower central vertical lobe and may produce a vertical "sweet spot." The round flanges of the Vifa drivers may be squared off to obtain closer spacing.

ANATOMY OF AN SK

As a follow up to a letter I had written to Dr. Bruce Edgar about the Speakerlab SK (*SB* 4/86), I wrote another letter to Dr. Edgar, confused about the throat area of the SK. Prior to 1977 when I built my SKs, Speakerlab had apparently been using a 6 by 13-inch "slot" between the woofer and the horn, but had changed to a 3 by 13-inch at that time.

When I questioned this, Speakerlab's Pat Snyder stated that the slot did not set the throat area, it was set in the "duct" leading from the slot. The narrower slot simply provided some mass loading on the woofer (as if trying to compensate for a woofer with the wrong specs?). In a subsequent publication (*The Loudspeaker Design Guide*), Speakerlab places the throat area of the SK at 50 in.[2].

As shown in *Fig. 1*, the SK divides the sound wave going through the slot into two equally dimensioned, exponentially expanding paths (you can't see the expansion in front of the woofer in *Fig. 1* because it is in another plane—*Fig. 2*). The smallest cross-sectional area of the throat

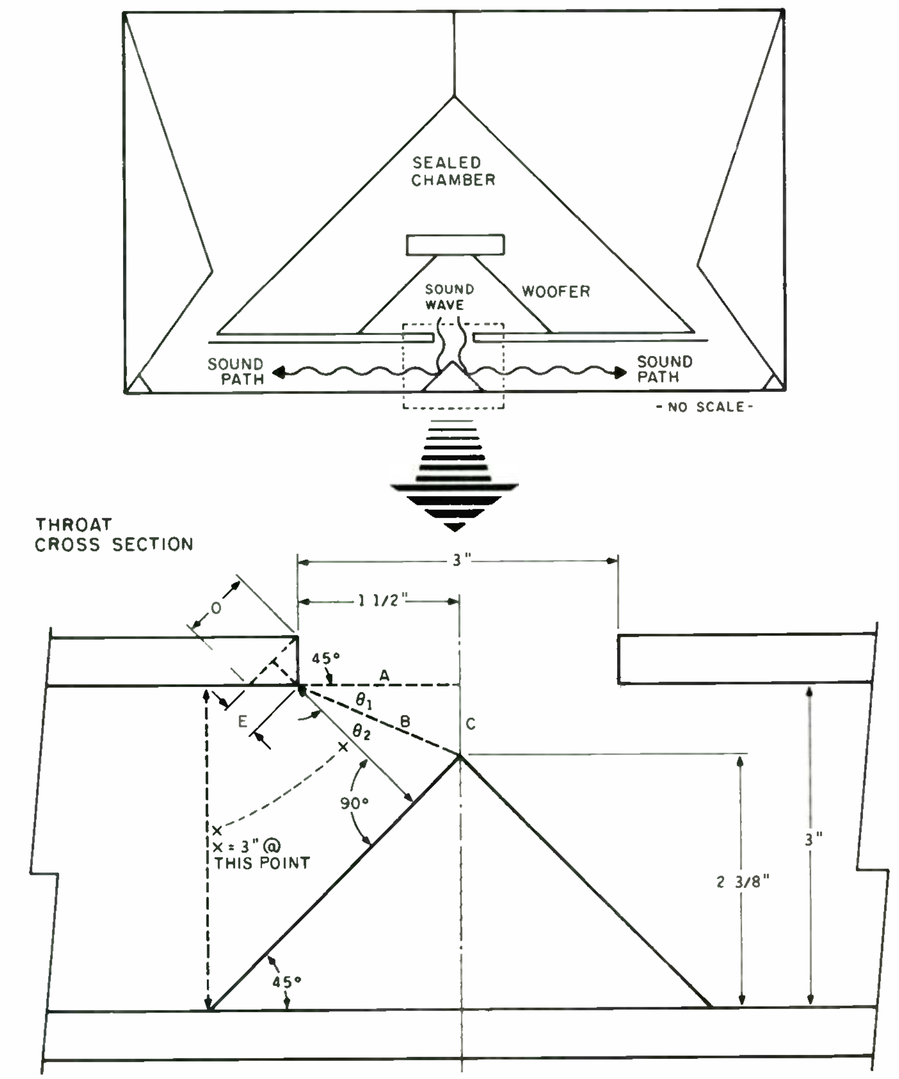

FIGURE 1: Throat cross section. The enclosure's top faces left, as shown here.

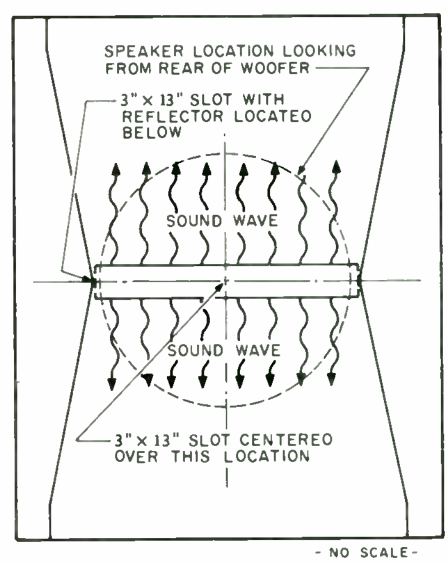

FIGURE 2: View from the back of the woofer.

occurs through line X, which can be mathematically calculated to be 1½ inches. Thus, the throat area at this point is approximately 2 by 1½ by 13 (39 in.^{2}). The slot is 13 inches wide at its centerline, but the sound path starts expanding immediately (*Fig. 2*).

After this "point" is passed, you can see the area expands rapidly until dimension X is 3 inches. If you cut the edge of the 3 by 13-inch slot at a 45° angle (line D in *Fig. 1*), the width (X) at that point would increase by "E," or 0.353 inches. In this case, the throat area would be 2 by 1.853 by 13 (48 in.^{2}).

However, I don't believe the throat area is set at this "point" location at all. I believe it is set where the main duct starts (where X = 3 inches) and is simply 2 by 3 by 13 (2x for two sound paths), or 78 in.^{2}.

I plan on trying the Focal Audiom 15PA (47.7Hz, 0.34, 6.1 ft.^{3})—optimum throat area 79.1 in.^{2}, mass cutoff 280Hz, in my SKs in addition to enlarging the 3 by 13-inch slot to 6 by 13. My thanks to Dr. Edgar for his help.

Dave Wharran
Clearville, PA 15535

ON VENEERING

On the advice of a friend (an electrical engineer, stereo fanatic and owner of every issue of *Speaker Builder*), I built the transmission line subwoofer based on Craig Cushing's article (*SB* 1/85). I am very impressed with the performance of the unit and how much it adds to my system. Since I completed the subwoofer, I became a subscriber to your magazine and now plan to build the Swan IV satellites as outlined by authors D'Appolito and Bock in issue 5/88. This design was again recommended by my friend as being one of particularly excellent design. I must take his word for it, since most of the technical information is beyond me. He was right about the subwoofer though, so I'm excited about this new project.

I read the Swan IV article several times to familiarize myself with the design and actual construction of the satellites. The article is well-written and the speakers look fairly simple to complete. When it comes to woodworking and veneer in particular, I have a great deal of practical experience. I do have a couple of comments that might aid individuals who have experience with the technical end, but could use a hand with the actual construction aspect.

First, I am fortunate to live around the corner from one of the best veneer houses in the country: The Woodshed, 1807 Elmwood Ave., Buffalo, NY 14216, (716) 876-4252.

Their mail-order catalog, available for a dollar or two, has an excellent selection of veneers, finishes, adhesives and general veneering and woodworking information.

Second, when veneering anything—a table top, bookcase or speaker cabinet, *both* sides of the plywood or particle board must be veneered. You can use a cheap veneer on the back, but if you omit this step, the board may warp as the veneer moves about from humidity and temperature changes. On a small cabinet like the satellites, it might not be a problem, but on the large woofer enclosures outlined, you are begging for trouble. I usually purchase preveneered ¾-inch MD-44 particle board. Locally, I can get oak in 4 by 8-ft. sheets for about $35, but you can use birch or whatever else is inexpensive. This also assures that your cabinet will be well-sealed as a finished product.

Without access to a veneer press, most adhesives can be very troublesome to work with and the results can be discouraging. I disagree with the authors on their choice of epoxy as the adhesive to use in this situation. It is expensive, a pain to work with, difficult to sand due to its hardness, irritating to the skin and will permanently discolor some species of wood.

If you go to all the trouble of clamping the panels, as the authors suggest, the adhesives of choice are polyvinyl resin glue (white glue) which will set up under pressure or heavy weight in 40 minutes or so and be ready to work with in 12 hours. The same is true for aliphatic emulsion glue (yellow glue, Titebond), except that it tacks and sets faster and can be difficult to work with on large panels.

Contact cement is another alternative, but requires some experience and is very unforgiving. Once the veneer is set in place you'll never move it.

Finally, I have the most experience with a heat-activated sheet glue (sold by The Woodshed) that comes in two-foot wide rolls with a paper backing on one side. You place the sheet glue over the particle board or plywood and using a clothes iron, set slightly below the cotton setting, firmly press the glue in place. Be methodical and press firmly to ensure you heat the glue enough. Practice on a scrap first. Peel off the paper and save it. Don't trim the excess glue yet. Place the veneer on the ironed-down glue, place the paper over the veneer and repeat the ironing process. Make sure you heat all the edges well and press firmly. After you remove the heat, use a rolling pin and go over the veneer several times while still hot to force out any air pockets. Let the piece cool for five minutes or so and carefully check the veneer at the edges to make sure you have good adhesion all around. Trim the piece with a solid carbide formica-type router bit and you will have a perfect piece.

This glue is very strong, you don't need special tools and you can finish a cabinet in one evening. The disadvantage is that small air bubbles can appear in your veneer where the glue did not take well. This is almost always a function of the type of veneer used or failure to clean the back of the veneer before ironing. Use VMP Naptha [*in a well-ventilated area*] to clean the veneer—it removes natural wood oils as well as dirt and evaporates quickly without damaging the veneer. Avoid veneers like teak that are naturally oily or have a high silicate level. They are tough enough to glue with the right equipment and can be very aggravating to use, especially if this is your first experience.

An excellent veneering book is *Veneering Simplified*, by Harry Jason Hobbs, published by Scribner's (and I believe available through The Woodshed). It does not cover the iron-down glue. I have never seen a textbook that addressed this method. I would be interested to hear of other people's experience with veneer and will offer any help or advice I can via letters. The gentlemen at The Woodshed should be able to help you with a veneer choice and the other items that you may need that the authors did not mention, such as veneer tape for joining two pieces of veneer for large panels.

Tom Basinski
Buffalo, NY 14216

James Bock replies:

Thank you, Mr. Basinski, for your informative letter. I am afraid that you may have misread the Swan article, with the impression that we used veneer. We did not, but did use veneer-faced ¼-inch plywood. We used plywood for several reasons: it is far easier to bond plywood with epoxy than to veneer, unless the builder is experienced in veneering; ¼-inch plywood and ¾-inch particle board are readily available; and this laminated structure is inherently self-damping.

It is quite true that both sides of a panel should be veneered. Indeed, both faces of a board should be finished with paint or varnish in good furniture making practice. The reasons are identical; to avoid an unbalanced, warp prone panel. The Swan IV enclosures present a very different situation. The enclosure structure itself resists any tendency to warp and the plywood/particle board laminate is quite a different structure from a veneered board. In the many years that Joe and I have built enclosures, we never had one problem with warping. I live on the ocean where humidity is always high. Perhaps you might have a problem if you do not use high strength epoxy adhesives, but use instead a water-based white or yellow glue.

I cannot recommend ¾-inch veneer faced MD-44 (a fiber board, not a particle board) for the bass enclosures. It is not thick enough to assure good joints, particularly where the corners are chamfered, and it will not be stiff enough. For our production version (which differs in construction from that of the article) we used 1-inch veneer-faced MD-44 and epoxy. You could adapt the structure to use 1-inch MD-44 if you can find it, and veneer the panels or veneer the assembled enclosure. The skills required are greater than those required to build as published.

Mr. Basinski's advice on using real veneers is right on the mark. For those of you who wish to tackle veneering, you will have to find 1-inch MD-44; ¾-inch particle board is not stiff enough for the bass enclosures, nor is it adequately self-damping. Furthermore, without the strong plywood laminated to it, unlike MD-44 fiberboard, particle board is relatively weak and breaks on impact. 1-inch plywood (¾-inch plywood is too thin) probably would be too resonant, and today's plywoods are so full of adhesive robbing voids that the joints as designed may not be adequate.

Since MD-44 weighs about 125 pounds for a 4 by 8 sheet, be warned that you must pre-cut to approximate size to handle it with ordinary home table saws. The published cutting plan may not leave enough extra room for such double cutting.

Everyone is free to build as they please, but I do not recommend white or yellow glues for this project. They require high clamping force to yield barely adequate pressure for thin glue lines, they bring water into the joint resulting in expansion and contraction, are not strong enough in themselves for the joints of this design, and lack gap filling qualities. (Not everyone will have 100% perfect fits.)

While I use yellow glues for some jobs, I have tried it for the plywood to particle board lamination and was not satisfied with the quality of the bond. As Mr. Basinski warns, the yellow glue may begin to skin over before you are ready to assemble. Tapping the finished laminate suggested areas not well bonded and the panels were warped slightly by the moisture. Laminating plywood to both sides might have balanced this out. For the structural joints of these enclosures, both white and yellow glues tend to migrate from the joint into the interstices of the particle board, leaving a "starved" joint. The cleats or glue blocks then become a major structural element of the joint. Worst of all, some joints in furniture, carefully made and properly clamped using white glue, have failed after about 20 years.

I cannot recommend contact cements for the plywood/particle board lamination. The good ones are extremely explosive and therefore are hard to find. The non-explosive ones are not as good. They work okay for veneer or melamine, but would not allow for an adequately strong bond for stiffer plywood, nor would they allow for accurate positioning. You get but one chance at alignment.

Hot melt sheet adhesives are useful for veneering, but not appropriate for plywood lamination; you cannot achieve the temperature at the bond (¼-inch away from the heat source) with anything available at home.

Any kind of glue which gets onto the finished surface will show. Epoxy smears are almost impossible to remove. For the article, I assumed that smears were likely to be encountered by home craftsmen. Rather than fight the problem, I chose to join it by coating the entire cabinet with clear epoxy. Smears are rendered moot, the surface is toughened significantly, the grain pores are filled, and the warm color is desirable.

Yes epoxies are more expensive, but in the quantities needed for this project, the extra cost is trivial. Yes, long exposure to epoxies can cause dermatitis, so glove up. They are strong, forgiving, moisture free, tolerant of low pressures, gap filling, provide relatively long working time when spread out, are long lived (more than 30 years in my experience), and no more difficult than any other glue properly used. While not an issue here, the curing can be speeded up by placing the assembly in a hot environment or using heat lamps, hot air blowers, and so on.

M-S Speaker

continued from page 15

ting far from the wall, the delay time must stretch into milliseconds.

To calculate rough delay times for most situations, I use as a rule of thumb one foot per millisecond; thus estimating sound travel at 1,000ft./sec. Sound actually travels a little faster: in dry air at 32°F (0°C), 1,087ft./sec (331m/s); and at 68°F (20°C), 1,130ft./sec (343m/s).[19] This is about 73.7–76.3μsec/in.

Driver Considerations. You could vertically align the front drivers by moving the midrange, high-frequency and rear drivers to the top of a woofer cabinet in a triangle configuration, or three sides of a square. This brings them to a near coincident position without adding delay.

Wall reflections from the rear drivers may be more desirable than sound that wraps around from the sides by diffraction. Cone midrange drivers rather than dome types may work better for this. Also, cabinet designs to isolate the direct sound from the rear drivers may be necessary.

I thought about using electrostatic panels in a "T" configuration (an ES-T?) The top or bottom of the "T" could be pointed toward the listener. Adding delay to the front driver would make the drivers effectively coincident. Could changing the delay of whichever panel is in front affect how the phantom left and right channels are beamed into the listening room?[20] Should the back be covered by sound absorbing material?[21]

I no longer have access to a wood shop to try these ideas so I'm turning my many questions over to you. I've started with my design and the theory behind it, and you can experiment to your heart's delight. Have fun.

NOTE. Please contact the author concerning licensing arrangements for commercial manufacture.

19. Halliday, David, and Robert Resnick, *Fundamentals of Physics*, John Wiley & Sons Inc., New York, 1981, p. 319.

20. Williamson, Reg, "The Quad 63," *SB* 1/82, pp. 10–18.

21. Cox, Thomas, "Spot Sound Absorption," *SB* 3/88, pp. 40–43.

ACKNOWLEDGMENTS

Thanks to Ray Taylor for the photography, my mom for makin' me do good English, and the University of Missouri at Kansas City Conservatory of Music Recording Department for the space to try out my crazy ideas.

Odyssey

continued from page 20

tried subwoofer amplifier gain adjustments from "off" to about 8dB higher than the final setting. The effects were immediately evident while listening to various types of music, that is, no support for the satellites to too heavy in the low frequencies. I chose the final setting by ear and confirmed it with the sound pressure meter.

SOURCES

Digi-Key Corp.
701 Brooks Ave. South
PO Box 677
Thief River Falls, MN 56701-0677

Old Colony Sound Lab
PO Box 243
Peterborough, NH 03458
computer program
Active Filter Cookbook (Sams)

Madisound Speakers
8608 University Green
PO Box 4283
Madison, WI 53711

REFERENCES

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

2. Bullock, R., and B. White, "Box-Response," *SB* 1/84, p. 13.

Third Dimension

continued from page 34

it is the best, but three years' practice, teaches me it has great advantages. Just try it, and if you wish, take the flag to carry it further. Enjoy yourself as much as I do.

SOURCES

La Maison du Haut Parleur
138 Au Parmentier
75011 Paris, France

Haut Parleurs Systemes
35 Rue Guy Moquet
75017 Paris, France

REFERENCES

1. Small, R.H., an anthology of articles on loudspeakers from *JAES*, Vols. 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.

Although the HIF12EB driver, specified for the bass section of this project, is no longer actively produced by Audax/France, Evan Struhl of Polydax has arranged for a limited number of sets of eight drivers (four per channel) to be available from US Polydax dealers. Use Fast Reply #185 for a list of dealers, specifications for the driver and pricing.—Ed.

Software Report

continued from page 49

driver, as we have, is so impedance compensation can be added or removed without measuring the impedance under each condition, as required with Peter's program. Also, by modelling the impedance with an electrical network, phase must not be measured. I found that even the largest companies did not have phase meters in those days. You can also optimize the impedance compensation network.

CACD does not require a minimum number of frequencies for the optimizer as does Peter's program. You can use just one frequency, if desired. A maximum of 20 seemed to be enough, even for a band-pass or midrange design, and it fit the screen layout.

The Loudspeaker System Simulator (LSS) will, in the upcoming version, assume the drivers are placed as close as possible to each other, in a vertical row, more or less. This is the case in many designs, anyway. Also, this assumes a measurement distance that makes the angle of the microphone to any driver very small and is on-axis with the tweeter. Again, this is the case with many designs, and the 1m measuring distance is standard.

LSS will allow loading predicted responses (magnitude and phase) from CACD generated data, to predict the composite system response. You will also be able to generate ideal acoustic target responses with the internal transfer function analyzer and sum them for a composite system response. In both cases, relative front-to-back distances between drivers can be entered to observe the effects of time offsets. By using the internal generator, you can explore combinations of Butterworth, Linkwitz-Riley and Bessel responses.

In later releases, more complex analysis and simulations will be possible. I believe LSS will prove to be a powerful prototyping tool.

Thanks for noting CACD will optimize active crossovers. A common misconception is that if active crossovers are used, you eliminate all the problems of passive networks. No universal crossover exists, either passive or active. Active filters do work into resistive loads, that is, input to an amplifier; however, they must provide the same correction of the driver's acoustic response that the passive network provides.

Classified Advertising

TRADE

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

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

CARDED WOOL: 100 percent clean and carded wool from the long-wooled sheep for the speaker builder, $11 per lb. plus shipping and handling, $3.75. Samples, $2 plus SASE. Send to ***BABYLON FARM,*** 95 Day St., Granby, CT 06035. T5/89

ACOUSTA-STUF—New long fiber synthetic stuffing material, multi-directionally crimped for sonic purity comparable to wool. $7.00 per pound. ***MAHOGANY SOUND***, 2430 Schillingers Rd., #488, Mobile, AL 36695. T3/89

AUDIRE: MODIFIED STRATHEARN Ribbon Drivers $200 each. Rebuild your drivers $100 each. (714) 968-1236. T2/89

AUDIO DESIGN SOFTWARE for PC and Mac. SpeakerCAD graphs predicted performance, calculates Thiele/Small parameters, vents. Filters calculates 1..6th order, Bessel or Butterworth electronic crossovers. Crossovers handles 1..4th order "All-Pass" or "Constant-Power" passive designs. $40 complete. ***ATRIUM ELECTRONICS***, 2302 5th NE, Salem, OR 97303, (503) 363-5143. T6/89

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

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

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

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

1. All "For Sale" ads must be personal sales and not for profit. Ads for resale at a profit must be submitted as "Trade" ads at 55¢ per word, prepaid.

2. Ads are limited to one per issue, 50 words or less. A word is any collection of letters or numbers with a space on both sides of it. Words over 50 must be prepaid at 20¢ per word. Longer ads not prepaid will be discarded and not published.

3. Ads cannot be returned. We will make every effort to substitute a revised ad for an earlier submission if both are clearly labeled with the same name and address at the top of the sheets.

4. Copy for ads or corrections to ads will not be taken by telephone. We cannot answer telephone queries about whether or not an ad is in a particular issue.

5. Ad copy should be clearly printed in block capitals or typed. Illegible ads will be discarded.

6. If you include only your name and a telephone number in your ad, your full name, street address, city, state and zip must accompany the copy.

7. Personal ads are a free service to current subscribers. It will help us a lot if you include your subscriber number (upper left corner of your mailing label) with your ad.

8. If you want an acknowledgment of your ad, including in which issue it will appear, please include a stamped, self-addressed postcard with your ad copy.

FOR SALE

Counterpoint SA-12, new in carton, warranty, $779; Hitachi D-220 cassette deck in box, $30; Graco T-160 FM Tuner, tubes, $20; Dynaco stereo power amp, tubes with A-410s, $60. With manuals: Magnepan MG-Ibs, like new in carton, $600; NEC SW-300 E subwoofer, built-in amp, carton, $225; JVC RX-150BK receiver in box, $65; Philips GA-312 turntable, $45; Philips GA-212 turntable, $35; Knight AM-FM stereo tuner, $25. R. Beem, Rt. 2, Box 492, Commerce, GA 30529, (404) 367-5654.

NAD 7175 receiver, $500; Oracle Delphi II, $649; Nakamichi CDC4A CD changer (sealed), $799; DBX 224 noise reduction, $100; or best offers. Also willing to trade new stereo equipment for new computer equipment. Tim (201) 232-0483 days, (201) 521-3774 evenings.

New Kevlar drivers with cast frames: Eton 7-inch 7-380/32, $150 pair; Focal 8-inch 8K515, $110 matched pair; Focal 1-inch tweeter T-90K, $45 pair; Scan-Speak 5¼-inch cone-dome 13M/8640, $70 pair. These are all outstanding drivers, but my project changed! Lynn Olson, 7740 Memory Lane, Portland, OR 97225, (503) 292-1921.

Koss K/4DS digital delay system with speakers, mint, $145; Most *Absolute Sound* between 9 & 46, $8 each, postpaid; *Audio Amateur* '81, '82, '85, '86—'88, $15/year postpaid. Audax MHD24B45 10-inch cast frame Bextrene, $38 each; HD20B25H2C12 8-inch Bextrene, $18 each; SEAS 21FWBX-DD 8-inch, $18. Steve (407) 870-9937.

SAE X10A amplifier, 100 WPC Class A, 3 months old, 5 year warranty. Cost $900 will sell for $495. Call anytime (213) 939-3482.

Back issues: *Speaker Builder*, *Stereophile*, *Stereo Review*, *The Absolute Sound*, *Audio*, *Goldmine*. Hunter, Box 40603, Jacksonville, FL 32203-0603.

Twelve passive 4kHz crossovers, 12dB/octave made by Custom Sound Reproduction Co., $40 each, all 12 for $360; Technics SH8000 acoustic test set, $160; rechargable battery for Audio Control Industrial SA3050A analyzer, $185; case for analyzer, very heavy duty, $125; Bose blue-cone replacement speakers, $35 each. Tom Young (203) 274-2202.

Classical stereo records, $1-$3 each; out of print Dolby-B quarter track open reel tapes; Yamaha M-35 2/4 channel power amp, $200; Dyna FM-5 FM tuner, built-in phono preamp, $70; complete *Stereophile*, $450. Stamp gets list of above plus more. All negotiable! Darroch, 1807 Elm Crest, Arlington, TX 76012.

Altec-Lansing model 15 monitor speaker systems (pair), oak wood finish, 27″H x 22″W x 17½″D, floor standing, like new, original owner with manuals. This was smaller version of model 19, and same as new model 14, $750/pair. Ron Brantley, 1604 Terrace Dr., Westminster, MD 21157, (301) 875-2571.

Thorens TD-166 MKII turntable, mint, $140; Mobile Fidelity Sound Labs Beatle Box, $500. Dan Dennison, RFD 10, Box 386A, Concord, NH 03301, (603) 798-5802.

Precision Fidelity C-8 tube preamp, good condition, very little use, $400; (4) Dynaudio 21W54/06 8-inch cast frame woofers, (list $120 each); Frame/magnet structure in good condition, needs reconing through Just Speakers for $65 each, $20 for all. Vance Hoyt, 23296 Brookhaven Dr., Moreno Valley, CA 92388, (714) 242-1155.

WANTED

New or like new pairs or singles of: Vifa P21WO-12, D25TG75, P17WH; Precision TD205R, TD165R; Dynaudio D-28; Morel MDT-28, 30, and 27. M. Thompson, 194 St. David Rd. W., St. Catherine's, Ontario, Canada L2T IR4.

Eico HF-35 tube power amplifier (in working condition); Pioneer SG-9800 equalizer. George Mueller, RR 3, Box 541, Bell Ave., Glen Gardner, NJ 08826, (201) 537-6245.

A photocopy of the service manual for a Delco model GM-2700-F AM/FM digital cassette car stereo with service reference number 06EECT1. Also wanting to purchase a solid state-type distortion analyzer. Greg Nawrocki, 21 Indiana St., Kitchener, Ontario, Canada, N2H 2A4, (519) 745-1579.

Miller 565 Crystal AM tuner; Karlson enclosure booklet/brochure, other Karlson data; Electrovoice "Temples of Tone" brochure, other early EV ads/data; old Klipsch catalogs; *Audio Engineering* and *High Fidelity* magazines volumes 1-5; Ampex 600 (price and condition); diaphragm for International Projector Corp. LU-1000 hf driver. D.R. Schaller, 6704 Schroeder Rd., Suite #6, Madison, WI 53711.

Copy of *Test Report's* "reviews" of Harman Kardon 870 power amp. Dan Dennison, RFD 10, Box 386A, Concord, NH 03301, (603) 798-5802.

GE variable reductance cartridge with a 3mm stylus for 78 rpms, or any 3mm decent magnetic that will scrape off sound well. Jarl V. Elo, 147T, Rindge Turnpike, Ashburnham, MA 01430.

Inovonics portable ⅓-octave analyzer; Ivie IE-30A; IBM PC with speaker design software, etc.; JBL 2397 diffraction horns. Tom Young, (203) 274-2202.

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.

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.

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

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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(20 cents/word)

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.

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).

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

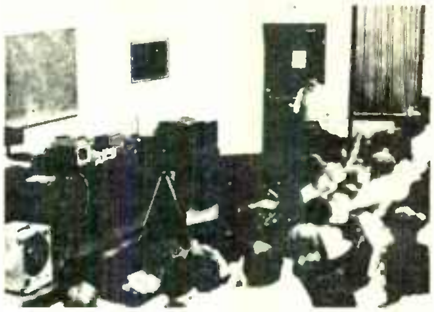
THE COLORADO AUDIO SOCIETY is a group of audio enthusiasts dedicated to the pursuit of music and audiophile arts in the Rocky Mountain region. We offer a comprehensive annual journal, five bimonthly newsletters, plus participation in meetings and lectures. For more information, send SASE to: CAS, 4506 Osceola St., Denver, CO 80212, or call Art Tedeschi, (303) 477-5223.

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, PO Box 611662, San Jose, CA 95161.

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

AUDIOPHILES IN THE Dayton/Springfield, Ohio area: I am interested in joining any existing music or audio club, or possibly forming a new one. I'm interested in electronics construction, modifications, recording and music. I'm also interested in "comparing notes" with others on modifying Magnavox CD players (POOGE-4 etc.). Contact Ken Beers, 462 Blose St., Tremont City, OH 45372, (513) 969-8402.

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.

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.

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

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

THE 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.

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.

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.

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

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

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.

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.

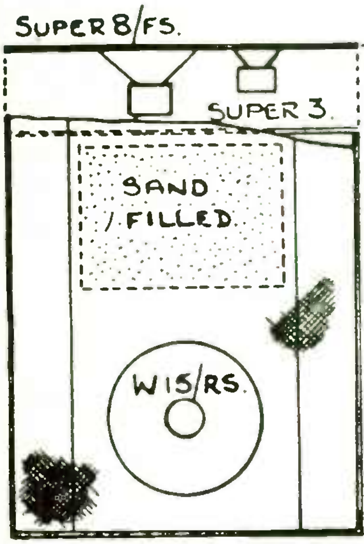
Advertising Index

Vintage Designs

Wharfedale Airedale

Designed by G. A. Briggs (circa 1961)

Courtesy of Gardiner McCauley, Columbia, MO 65201



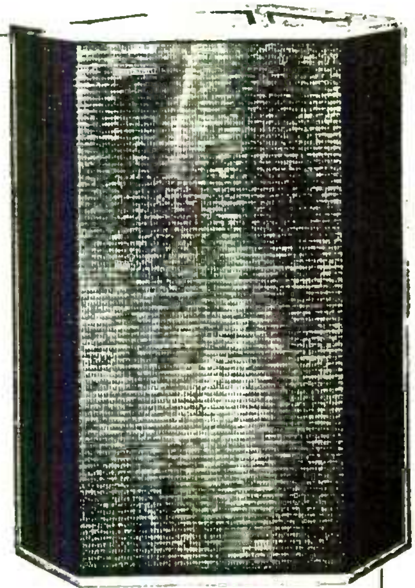
For many years the Wharfedale omni-directional 3-speaker Corner System has been recognised as a superbly natural reproducer.

The AIREDALE is the latest version of this famous speaker presented for the first time in a one-piece free-standing assembly suitable for corner or along the wall location, and ideally suited to work with the original corner model on stereo, especially where a second corner is not available, or is too far away.

The smooth, clean bass is characteristic of the high flux 15″ unit which is now fitted with roll surround.

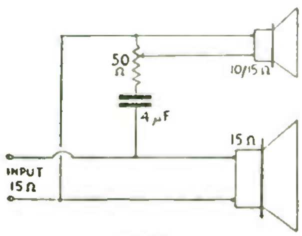
The 8″ mid-range and 3″ treble units face upwards for omni-directional treble, and are arranged in a manner which imparts a natural airiness to the reproduction.

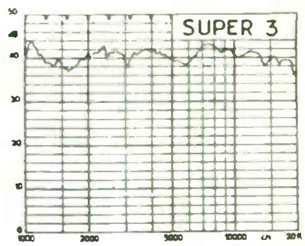
Cabinet resonance is avoided by sand filling the front panel and fitting ceramic tiles to rear side panels. Some idea of the solid construction is given by the fact that the total weight exceeds ½ cwt.



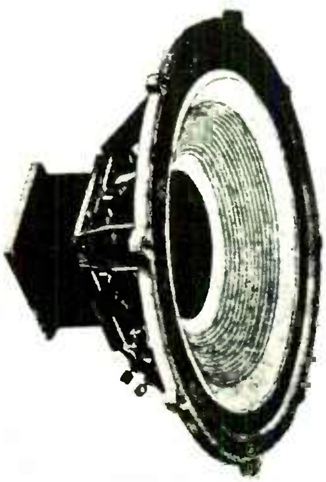
SUPER 3

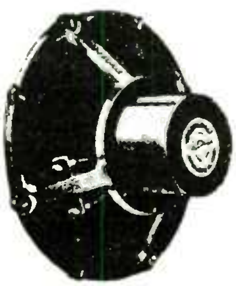
2/3 ohms or 10/15 ohms.
Flux density 14,500 gauss.
1″ dia. centre pole.
Aluminium voice coil.
Max. input 6 watts rms. or 12 watts peak above 1,000 c/s.
Frequency range 1,000–20,000 c/s.





Axial Response Curve.
Mic. distance 1′ 6″.
Input 4 v. at 1,000 c/s.



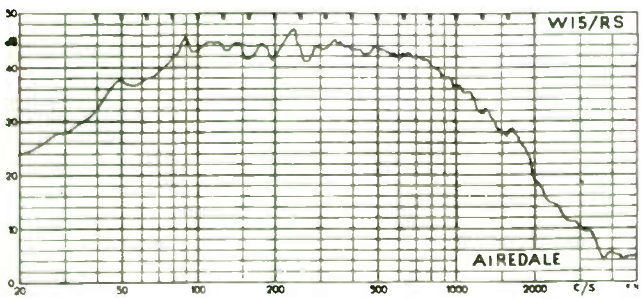


SUPER 8/FS

2/3 ohms or 12/15 ohms.
Flux density 14,500 gauss.
1″ dia. centre pole.

Max. input 5 watts rms. or 10 watts peak.
Frequency range 40–12,000 c/s.

12/15 ohms only.
Flux density 13,500 gauss.
2″ dia. centre pole.
Max. input 15 watts rms. or 30 watts peak.
Frequency range 25–2,000 c/s.



Response curve of new W15/RS in Airedale enclosure with crossover at 400 c/s.

Pox Humana

Flight Patterns & Phono Preamp

By Dick Pierce

Back when I was working in an audio retail store, about 1974, the search was on for a high-quality line of receivers that would meet our criteria of high performance, low distortion, reliability and good manufacturer support.

We selected the new Yamaha line (this was the CR/CA-400, 600, 800, 1000). In fact, we became the first dealer in the Northeast to carry this line, and did quite a bit of legwork to promote and support it.

The units were excellent tuners for the price. The power was typically underrated by nearly 40%, and they had vanishingly low distortion and intelligently designed, wide bandwidth amplifiers. The units looked conservative and pleasing. The phono preamps had good transient characteristics and were essentially overload proof, for all but the most bizarre conditions.

We sold quite a few and customers loved them. Sure, occasionally we dealt with complaints that the tuner cursor was sticking at times, caused by the Teflon track coming unglued at spots, and the everpresent noisy switch; but overall, customer satisfaction was high. At least for a while.

About 18 months after we started selling the units, a customer brought her CR-800 back, complaining (quite apologetically) that something might be amiss. She was unable to articulate exactly what the problem seemed to be, but she settled on an analogy, "It sounds like I live next to Logan Airport." Well, we agreed to look her unit over, and lent her a replacement. We set up her receiver with a pair of speakers, and listened. After an hour, nothing happened. We gave up and went home.

Later, the owner revealed it only happened while playing records. Okay, we listened to records. Again, after an hour or so, nothing happened. We were about to give up when the room started shaking. Everybody had this very frightened look on their faces. Soon, I heard what sounded like a Boeing 747 on final approach, a few yards away; the roar was deafening! I hurriedly shut off the speakers.

Into the lab we went, where we set it up again. Looking at the output of the power amp, and all the way back to the phono input, we saw nothing unusual. The output noise was equivalent to about $0.8\mu V$ total broadband noise on the input. For a while. After about two or three minutes, the noise started slowly and then increased rapidly. After another 30 seconds, the effective input noise level was equivalent to almost 0.5V, severely clipping most gain stages after the phono preamp!

Well, we thought, that's an easy one, just find which component is getting noisy with temperature and replace it. This took one or two blasts of spray cooler and we found the culprit: a single n-p-n signal transistor, a 2SC1345. Wizz bang—out it came, a new one went in, and presto—problem gone, customer happy, no charge, and Yamaha pays me $24 for five minutes work.

But (this would be a boring anecdote without a but), the unit came back about a month later—same problem, but different channel. Culprit? A different 2SC1345. Another five minutes, another $24—what a way to make a living.

Soon, several customers returned their units with the same problem. I notified Yamaha, and they said they were unfamiliar with the problem. Other problems then started to show up. Tuners were drifting (AFC driver happens to be a 2SC1345), stereo lights don't light up (lamp driver from MPX chip is a 2SC1345), output stage blows up, taking the tweeters with it (bias regulator is a 2SC1345), turntable starts running at erratic speed, sometimes as much as 100 rpm (you guessed it, the tacho sensor in the turntable speed control is a 2SC1345).

I replaced 2SC1345s left and right, making a bundle on in-warranty repairs. It turns out that *every* Yamaha unit we sold with a 2SC1345 somewhere in it, came back for repairs.

Then Yamaha pulled a fast one—they would no longer pay for repeat repairs on units with defective transistors, thus requiring repair centers to replace *all* 2SC1345s when the unit comes in for repair, which could take three to four hours for the more complex units. And how many repair people might butcher a board, replacing more than 20 transistors?

Interestingly enough, at the same time, Burwen Labs was having a similar problem in their noise reduction units, with uncontrolled noise after warmup. *All* Burwen units eventually had to be repaired.

The connection: both manufacturers were using semiconductor devices manufactured by Hitachi during a certain period of time. They discovered that an impurity in the curing resin for the plastic cases was corrosive on the more delicate parts of semiconductor fabrications.

I solved my Yamaha problem of not enough money/time to repair every unit by simply sending the motherboard or the entire receiver back and letting them handle it. Interestingly enough, they never officially admitted this problem existed.

As an aside, one CR-800 came back, also very noisy, but with different symptoms. Opening the case revealed a white residue covering much of the interior, along with what might have been minor fire damage in an area where a fire could not have started (the tone control/tape switchboard). I suspected this was not a warranty repair, and questioned the customer. After grueling cross-examination, I learned he was trying to free-base some cocaine on the shelf above, ended up spilling the whole mess into his receiver, and nearly burned his apartment down in the process.

SPEAKERS AND COMPONENTS

PIONEER · MOTOROLA · EMINENCE · MG ELECTRONICS · Polydax

WALNUT SPEAKER CABINET KIT

Super quality, genuine walnut veneer cabinet. Kit includes: routed and mitred top, sides, and bottom in unfinished 3/4" walnut veneer. Cut your own custom holes in the front and rear to match your drivers. 15" x 24" x 11". Volume: 1.9 cu. ft.

#260-350 $22.50 (1-3) **$19.95** (4-up)

SUBWOOFER CROSSOVER NETWORK

200 watts RMS. 12 dB per octave. 150 Hz at 8 ohm crossover point.

#260-220 $28.80 (1-5) **$24.40** (6-up)

TITANIUM COMPOSITE TWEETER

Polydax

Ferro fluid cooled dome. fs 1,200 Hz. SPL 90 dB. 50 watts RMS, 70 watts max. Polydax #DTW100T25F. Made in France.

#270-047 $27.50 (1-9) **$24.80** (10-up)

18" EMINENCE WOOFER

EMINENCE Made in U.S.A.

100 oz. magnet. 3" voice coil. 250 watts RMS, 350 watts max. 8 ohm. 30 Hz resonant frequency. 22-2700 Hz response. Efficiency 95 dB 1W 1M. Paper cone treated accordian surround. Net weight 29 lbs.

#290-200 $98.80 (1-3) **$89.50** (4-up)

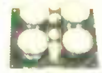
PIONEER HORN TWEETER

Mylar dome, 2.93 oz. barium ferrite magnet. 8 ohm. Response: 1,800-20,000 Hz. 35W RMS, 50W max. fs 2,000 Hz, SPL 106 dB. Pioneer #AHE60-51F.

PIONEER

#270-050 $6.50 (1-9) **$5.90** (10-up)

3-WAY 100W CROSSOVER



12 dB / octave rolloff. 800 Hz, 5000 Hz. 8 ohm. 100 watts RMS.

#260-210 $12.50 (1-9) **$9.95** (10-up)

15" 3-WAY, 125 WATT SYSTEM

Our "Top of the Line" system. The system features elements specifically selected to produce a balanced output throughout the full frequency bandwidth of the system. System includes: (1) #290-155 15" polypropylene woofer rated at 145 watts max, (2) #280-020 cup midranges, (1) #270-035 4" soft dome tweeter, (1) #260-215 200 watt 3 way crossover, (2) #260-265 100 watt mid, tweeter "L" pad attenuators, (1) #260-300 speaker terminal, and (1) #260-340 grille cloth.

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WOODGRAIN GRILLE CLOTH

Authentic woodgrain print design cloth. 36" x 60".

#260-340 **$5.95**

12" POLY WOOFER 100 WATTS RMS

PIONEER

Super duty 40 oz magnet. Polypropylene cone. 100 watts RMS, 145 watts max. 4 and 8 ohm compatible (6 ohm). 2" voice coil. fs 25 Hz. VAS 10.8 cu. ft. QTS .166. Response: 25-1,500 Hz. Net weight: 9 lbs.

#290-125 $36.80 (1-3) **$34.50** (4-up)

15" THRUSTER WOOFER

THRUSTER

Thruster by Eminence. Made in U.S.A. Forward poly roll foam surround, 56 oz. magnet. 2-1/2", 2 layer voice coil. 150 watts RMS, 210 watts max. 4 ohm. fs 23.5 Hz, QMS 9.86, QES .34, QTS .33, VAS 17.9 cu. ft. SPL 94.8 dB 1W/1M.

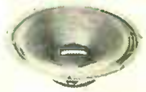
#290-180 $43.50 (1-3) **$39.80** (4-up)

12" 3-WAY, 100 WATT SYSTEM

System includes: (1) #290-125 polypropylene woofer rated at 145 watts max, (1) #280-045 heavy duty 5-1/4" midrange, (1) #270-035 4" soft dome tweeter, (1) #260-210 3-way 100 watt crossover network, (2) #260-255 50 watt tweeter and midrange "L" pads, (1) #260-300 speaker terminal, and (1) #260-340 grille cloth.

#12-100 **$73.95** Each

15" EMINENCE WOOFER



EMINENCE Made in U.S.A.

Ribbed paper cone with treated cloth accordian surround. 56 oz magnet. 2" 2 layer voice coil. 100 watts RMS, 140 watts max. 8 ohm impedance. fs 40Hz. QMS 3.7. QES .41. QTS .37. VAS 10 cu. ft. SPL 96 dB 1W 1M. Printed dust cap. Net weight: 15 lbs. Made in U.S.A.

#290-185 $42.50 (1-3) **$39.65** (4-up)

5-1/4" CUP MIDRANGE

PIONEER

50 watts RMS, 70 watts max. Response: 320-6,000 Hz. 8 ohm. Pioneer #B11EC80-02F.

#280-020 $11.50 (1-9) **$9.95** (10-up)

SPEAKER BUILDING BOOK

192 pages with 152 illustrations on speaker design and construction.

#500-020 **$12.95**

10" 3-WAY, 60 WATT SYSTEM

Woofer, tweeter, and midrange are Pioneer brand and are matched for compatibility. System includes: (1) #290-083 10" woofer, (1) #280-020 5-1/4" cup midrange, (1) #270-050 3-1/2" horn tweeter, (1) #260-200 3-way 60 watt crossover, (2) #260-255 50 watt "L" pads, (1) #260-300 speaker terminal, and (1) #260-325 grille cloth.

#10-60 **$48.95** Each

POLYDAX SOFT DOME

Polydax

8 ohm impedance, fs 800 Hz. Response: 800-20,000 Hz. 50 watts, 70 watts max. SPL 89 dB. Dimensions: 4.75" x 3.5".

#270-037 $16.50 (1-9) **$15.90** (10-up)

POLYDAX CUP MIDRANGE

Polydax

Paper cone cup midrange. fs 300 Hz, SPL 93.6 dB. 50 watts RMS, 70 watts max. Made in France.

#280-025 $32.50 (1-9) **$30.90** (10-up)

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morel

high fidelity range

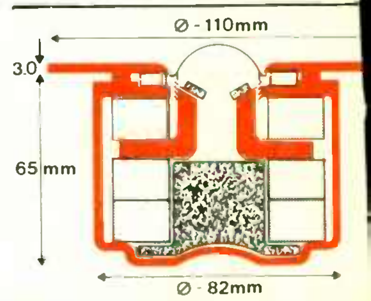
The MDT 33 is an extremely fast Tweeter, using a 28mm (1⅛") diameter voice coil and a chemically treated soft dome, and is ideally suited for two way systems with the possibility of a lower than normal crossover frequency, as well as for three and multiple way systems.

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

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

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

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

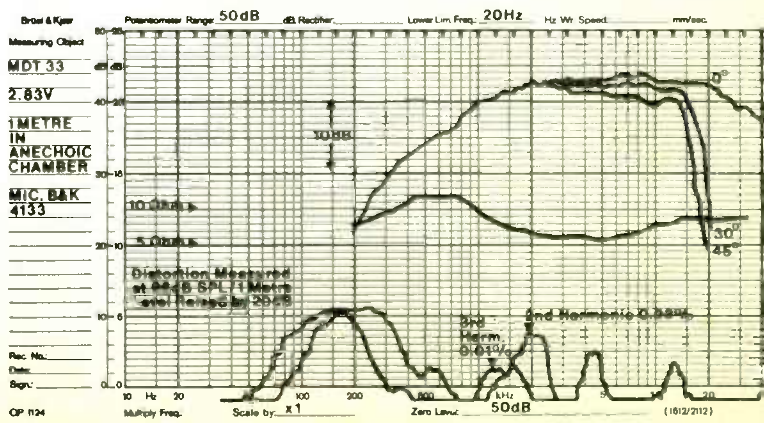


Specification

Overall Dimensions	Ø - 110mm × 68mm
Face Plate Thickness	3mm
Voice Coil Diameter	28mm (1⅛")
	Hexatech Aluminium
Voice Coil Former	Aluminium
Number of Layers	2
DC Resistance	5.2 ohms
Nominal Impedance	8 ohms
Voice Coil Inductance @ 1 Khz	0.09mh
Air Gap Width	0.75mm
Air Gap Height	2.5mm
Voice Coil Height	2.7mm
Flux Density	1.95T
Force Factor (BXL)	4.76 WB/M
Rmec	2.09ns/m
Qms	0.66
Qes	0.38
Q/T	0.24

Vas	0.016 l
Moving Mass including Air Load	0.44 gra
Effective Dome Area	8.5 c
Dome Material	Treated Fa
Frequency Response	1400-20000 ± 0.6
	(1000-40000 – 5
Resonant Frequency	500
Power Handling Din:	
X-Over 1400 Hz	10
X-Over 5000 Hz	50
Transient Power 10ms	150
Sensitivity	92.5d8 (1W/
Rise Time	10
Intermodulation Distortion	
for 96dB SPL	<0.
Harmonic Distortion	
for 96dB SPL	<0.
Nett Weight	1.2

Specifications given are as after 24 hours of running.



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